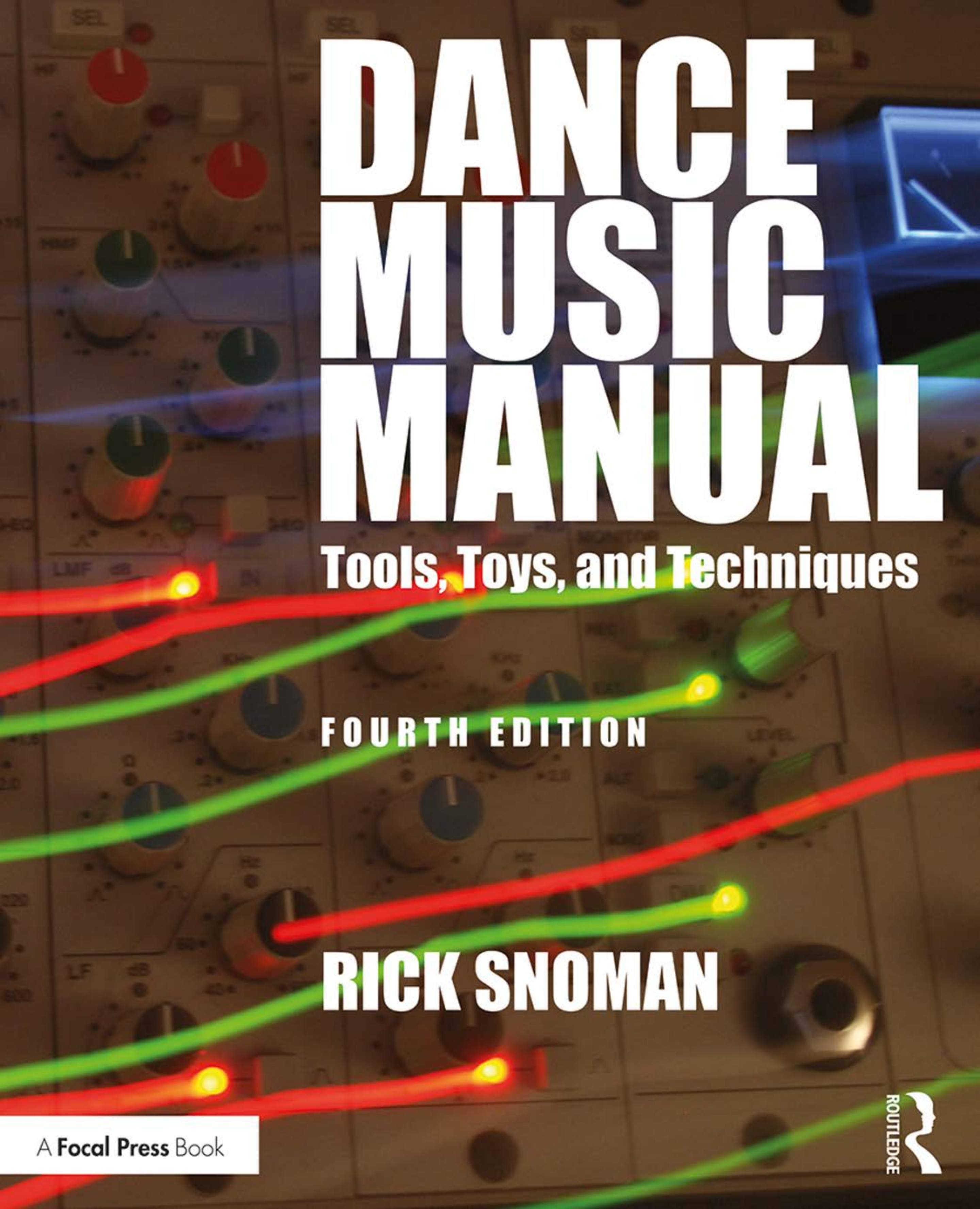

DANCE MUSIC MANUAL

Tools, Toys, and Techniques

FOURTH EDITION

RICK SNOMAN

A Focal Press Book

ROUTLEDGE

Dance Music Manual

Dance Music Manual, aimed at the novice and seasoned professional alike, takes the reader through the software and hardware needed to create original, captivating, and professional sounding music.

Key features of *Dance Music Manual* include:

- How to create compelling, professional-sounding original or remixed dance tracks.
- The differences between different genres and how to produce them.
- How to expose your tracks to their chosen audience and equip you with the skills to develop your career as a dance music producer and engineer.

Along with the book is a companion website, which provides examples of synthesis programming, compression, effects, MIDI files, and examples of the tracks discussed in this edition.

The new and improved fourth edition covers processes and techniques used by music producers, masters, mixers, and DJs. Each page is full of facts presented in a manner that is easy to absorb and implement.

Rick Snoman has over 30 years' experience in the production of electric dance music. He has released under various names including AeonSoul, Phiadra, NeuroKode, and Ascii, and has remixed professionally for numerous international artists. He has written articles and columns for leading music technology magazines, and is the founder of Dance Music Production, an industry leading educational website. He is the author of the first three editions of *Dance Music Manual*, which are required reading at a number of universities, and actively encourages the education/development of music producers. He currently runs his own recording studio and works as a ghost producer for artists and DJs.

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Dance Music Manual

Tools, Toys, and Techniques

Fourth Edition

Rick Snoman

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Visit the companion website: https://www.dancemusicproduction.com/

This book is dedicated to every aspiring electronic dance musician, and their passion for the music.

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Audio resources to accompany this book can be downloaded from http://www.dancemusicproduction.com.

Audio excerpts and examples used for the genre chapters of this book are released under various labels and available from most online record stores.

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Introduction

There would be no new school without the old school.

Vivian Scott, former Head of Urban A&R/Epic Records

Welcome to the fourth edition of the *Dance Music Manual*. Since the release of the first edition back in 2004, electronic dance music has evolved thorough the developments and improvements in both software and hardware. This latest edition covers many of these developments and techniques that are currently in use by today's producers and artists.

I was first introduced to dance music at the Hacienda in Manchester in 1988 and was a constant regular until its closure in 1995. During that time, I began to produce my own music using an Atari STE, limited MIDI instruments and 4-track tape recorder. As I progressed my skills over the years, I secured a job at a recording studio and have since released numerous white labels, worked and released music under numerous guises including Skreener, Neurokode, Phiadra, Aeon Soul, Ascii, and RedFive. I've remixed Kylie Minogue, Madonna, and Britney Spears, authored numerous articles and reviews, held a column in *Computer Music Magazine* (circa 2001) and currently run a recording studio, working as a ghost producer for DJs and artists. I also regularly hold seminars and lectures at universities and colleges on producing electronic dance music, and I also record tutorial videos for *www.dancemusicproduction.com*.

This book is a culmination of the knowledge I've attained over the past 30 years and I hope to offer you a shortcut of a good few years through my experience, discussions with other producers and artists, and personal observations of the techniques and practices that have developed.

However, I should stress that this book should not be considered a painting-by-numbers or a cookie-cutting course to your next number one hit. If that's what you're searching for, you won't find that information here. In fact, you won't find that information anywhere. Despite the growing trend for cliff notes or anything that will speed up an artist's rise to fame, there simply are *no* shortcuts or secrets to producing a great piece of music.

Instead, with this book, I aim to demystify the process and save you from having to sift through the minefield of misinformation from those looking to make a quick buck from impotent information, guesswork, and no practical experience of the music itself. More importantly, its purpose is not to suggest how it *should* be done but discuss how it *is done* today. It's down to you to show us how it's going to be done tomorrow.

Dance music is based around musicians and producers pushing technology in new directions. Merely reading a book, article, or tutorial will not turn you into an overnight sensation. Despite media representation of artists seemingly plucked from obscurity and thrown into the spotlight overnight, this isn't the case. It takes years of hard work and dedication to make it as an artist, DJ, or producer and there are no shortcuts.

While this book will provide you with the basic techniques and knowledge required to create some music, creativity cannot be encapsulated in books, pictures, video, or the latest product. It is your observations and analysis that will unlock your true potential and your experimentation that will produce the dance records of tomorrow.

Rick Snoman

http://www.dancemusicproduction.com
www.facebook.com/rick.snoman
www.facebook.com/dancemusicproduction

CHAPTER 1

The studio centerpiece

Computer games don't affect kids. I mean, if Pac-Man affected us as kids, we'd all be running around in darkened rooms, munching magic pills and listening to repetitive electronic music.

Marcus Brigstocke

The first requirement for anyone considering the creation of electronic dance music is whether to use a PC or an Apple Mac computer. Although there are a small number of dedicated hardware units that perform similar functions, the computer and its associated **D**igital **A**udio **W**orkstation (DAW) software form the centerpiece of almost every dance musician's studio. Whether to use a PC or a Mac depends on many factors, but finances are often the main reason. A PC running Microsoft Windows is often a far cheaper alternative than an Apple Mac running OSX. But for most studios and artists, the Mac is the more reliable and hardy platform of choice. A secondary consideration is the DAW you want to employ for your music creation. Most DAWs run in both Windows or OSX but one in particular – Apple Logic Pro X – is only available for the Apple Mac.

Most DAW's require a minimum of a quad-core Intel or AMD processor, 16GB of RAM, 100GB of free HDD space, and a screen resolution of 1280 x 768. These are the absolute minimum specifications; however, more memory, faster processors, larger HDDs, and plenty of display real estate are beneficial and will become a necessity as your skills grow.

In addition to the above, you'll also want a high number of USB ports since some audio production software will require a USB dongle. These are to prevent software piracy, consisting of a hardware USB 'key' to be connected to the computer's USB port before the software will function. One DAW in particular – Steinberg's Cubase – requires a USB dongle to be connected before you can use the workstation.

To use further software plug-ins within the DAW, you may be required to use a Syncrosoft dongle, an E-Licenser dongle, or an iLok Dongle. Or, depending on your software of choice, all three of them. Just these alone need three USB ports, and that's before you even consider an external USB HDD for additional storage, a suitable audio interface, and a USB piano keyboard. While you can employ a multi-USB hub for these purposes, some software protection devices and audio interfaces do not perform reliably through USB hubs. Also, many portable HDDs and some keyboards require a powered USB hub; thus you also need a free wall socket to power the hub.

Once the computer is chosen, the choice leans towards the appropriate DAW. With this software, you can record and edit audio, apply effects and processing, employ software representations of both classic and new synthesizers, and mix and master the resulting music. In fact, the DAW is such a powerful software application that some artists choose to create their music on a laptop alone.

Although all DAWs are similar regarding functionality and features, they do comprise different workflows and what works for some may not work for others. It would, therefore, be misleading to recommend any one specific workstation. Moreover, with the regular upgrades and updates that introduce new features (and usually bugs in equal measure), it would be unproductive to discuss the pros and cons of each. So, this chapter will not discuss specific options but instead cover the most commonly shared features to offer you a basic understanding.

Most DAWs divide their editing and workflow over several windows or pages. The most important of these is the arrange or session window. This is the page you're greeted with on opening the workstation and consists of an overview of the entire project. An example is shown in Figure 1.1.

The arrangement page consists of individual tracks – created by the user – that appear down the left-hand side of the page. When creating these tracks, you typically choose between audio, MIDI, or virtual instrument tracks.

FIGURE 1.0
The 3rd Gen iLok software protection dongle

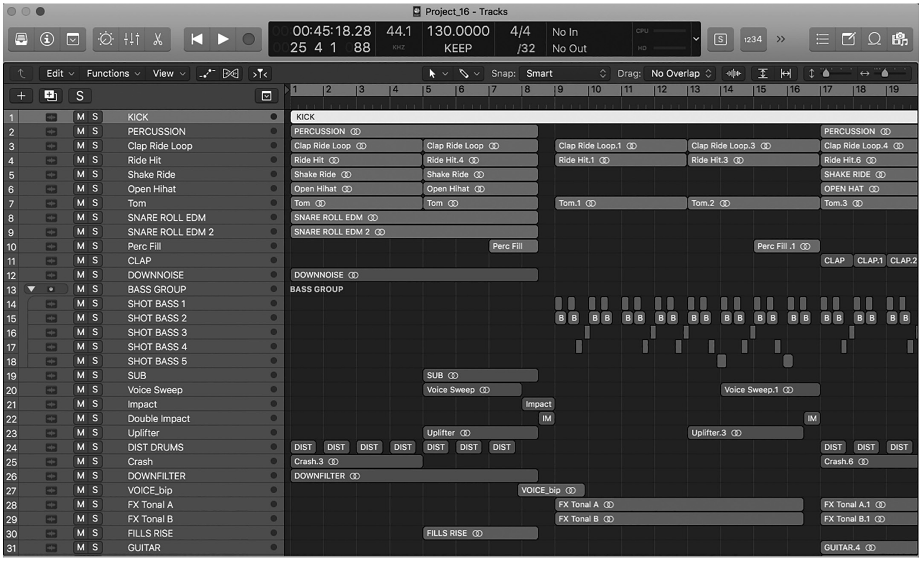

FIGURE 1.1
The arrangement page in Logic Pro X

AUDIO TRACKS

Audio tracks contain digital audio files. These could be to record audio onto, such as a vocal take, or they can be used to accept pre-recorded audio data held on the hard drive. These audio files can be dragged and dropped into the workstations arrange window where many workstations will display the audio as a region. Depending on the workstation, a region may be called an event, but these are interchangeable terms.

A region (or event) is a graphical representation of the audio file, displaying the audio waveform and its start and end points in the arrange window. The audio data is copied to a specific location on the hard drive but is only referenced by the workstation, and not imported into the project. This approach offers many benefits.

If you choose to repeat the same region 100 times in the arrange page, the workstation repeatedly plays the same audio file 100 times. This reduces the amount of hard disc space required as it doesn't have to store multiple copies of the same audio file. You can use editing tools to manipulate regions. These include cutting, slicing and rearranging the audio; modifying its pitch; reversing and creating fade in and fade-outs. However, since only the region is edited and not the audio data themselves, it ensures that many edits are non-destructive and can be undone numerous times if required.

While basic audio editing is performed in the arrange window, in-depth audio editing may only be available in a sample edit window. In many workstations, this window is opened by double-clicking on an audio region.

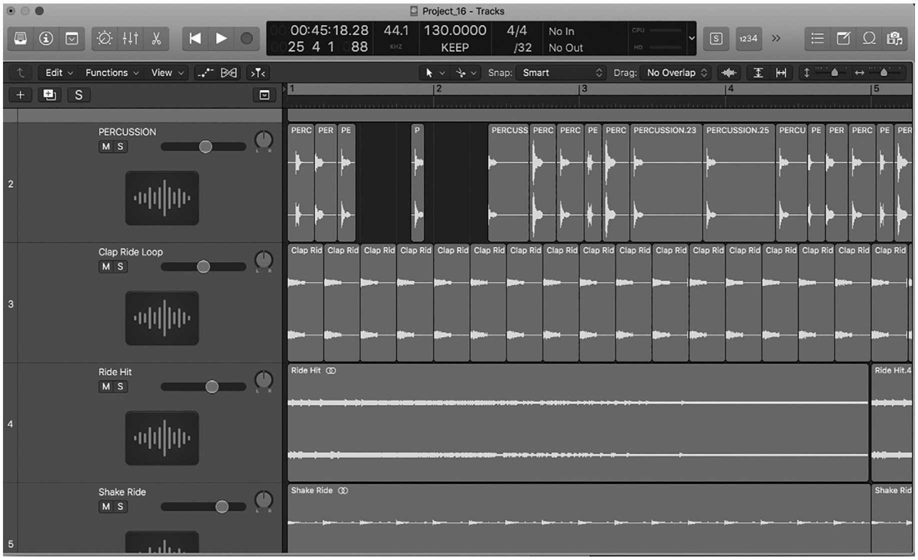

FIGURE 1.2
Audio 'regions' sliced and edited in Logic Pro X

The features in a sample editor window will vary, but they will feature further audio editing options such as normalizing, sample reversing, time stretching, dynamic manipulation, and slicing. Even though some of these functions can be performed on the arrange page, it is important to differentiate between editing in the arrange window and editing in the sample editor window.

As previously discussed, any editing performed in the arrange window only applies to a region and does not affect the audio data stored on the hard disk. Conversely, in many cases, edits carried out in the sample editor window will affect the audio data directly. This means that any edits performed in the sample editor window will be reflected in the arrange page. So, if you were to reverse a sample in the sample editor window, every copy of that region in the arrange page could also be modified.

Because of this, if you wish to make an edit in the sample editor window, you may need to first create a secondary audio data file for the region you're editing. This way, further copies of the same regions are not affected. This is commonly found in an audio menu as a duplicate region or convert region. By doing so, any changes made to this audio data will not affect the rest of the events or regions in the arrange page that point to the original file.

MIDI AND VIRTUAL INSTRUMENT TRACKS

Musical **I**nstrument **D**igital **I**nterface (MIDI) tracks and virtual instruments tracks are fundamentally the same as both can contain MIDI events. MIDI is a protocol that was established in 1983 for digital musical instruments to communicate with one another. The data transmitted in MIDI consists of simple commands such when to play a note, for how long the note is played for, and how hard the note is struck (aka velocity).

Within a DAW, a standard MIDI track is created when you want to send note data outside the DAW onto any attached hardware synthesizers. Alternatively, on a virtual instrument track, the MIDI remains internally routed to a software emulation of a synthesizer that is graphically represented within the confines of the workstation.

Regardless of whether the track is a virtual instrument or MIDI track, only very basic editing can be accomplished within the arrange window. Here, you are often limited to inserting, deleting, slicing, and moving MIDI regions around the arrange page, much in the same way as audio. Further in-depth MIDI editing is performed within a specific piano roll editor.

The piano roll editor is like an advanced musical conductor. You can enter notes and send the subsequent event data to any number of external hardware or internal virtual synthesizers. This is shown in Figure 1.3.

A piano keyboard appears down the left-hand side of the editor and denotes the pitch. To the right is the quantization grid on which the notes are 'drawn' (inserted) with a pencil tool. When playback begins, any notes in this field are played from left to right at a speed determined by the tempo of the piece. If many notes occur at the same position in time, they will play simultaneously.

Alongside drawing notes via the pencil tool, it is also possible to record MIDI data live. By connecting a MIDI capable piano keyboard to the DAW via a MIDI or USB connection, notes played on the piano keyboard will be recorded live

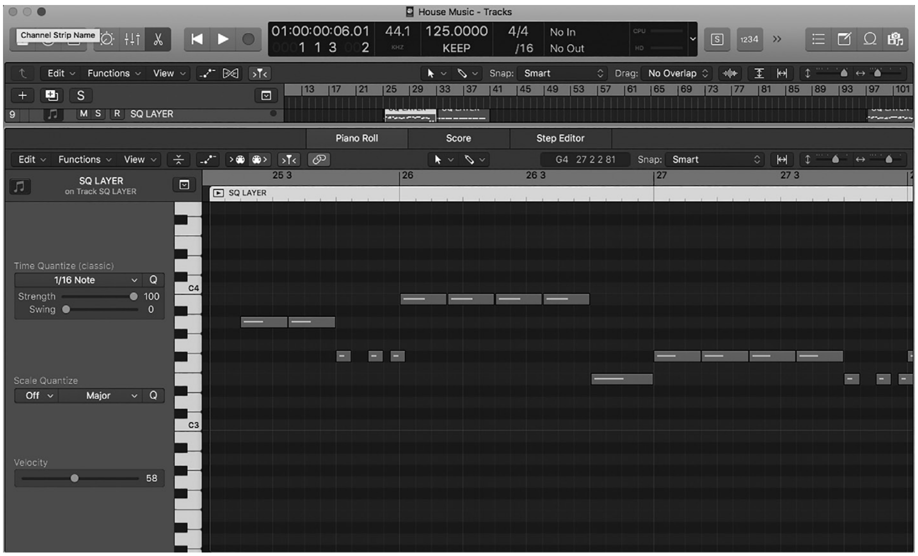

FIGURE 1.3
The piano roll editor in Logic Pro X

on the piano roll. It is important to understand that audio is not recorded from the keyboard.

Indeed, many MIDI keyboards do not produce sounds, only data instructions, and the DAW records these instructions such as note on/off and velocity.

Once recorded, the timing of these notes can be modified, and notes can be lengthened, shortened, or deleted to perfect the performance. Once editing is complete, this MIDI is transmitted to any connected synthesizer or sampler that will repeat the performance in whatever sound is currently selected.

Further editing windows may be available such as music notation or specific editing environments. These additional editors depend highly on the workstation, and since the majority of electronic musicians spend their time moving between the arrange page, piano roll, and sample editor, they are beyond the scope of this book.

ARRANGEMENTS

Any successful electronic music track will employ multiple audio and MIDI channels on the arrange page of the workstation. These are arranged from left to right (over time) to produce the finished music. Similar to the piano roll, these audio and MIDI regions may be layered under one another to play simultaneously or one after the other to create builds and drops. A typical arrangement is shown in Figure 1.4.

In the arrangement page, it is possible to perform multiple functions including copying and repeating regions across the length to produce an arrangement.

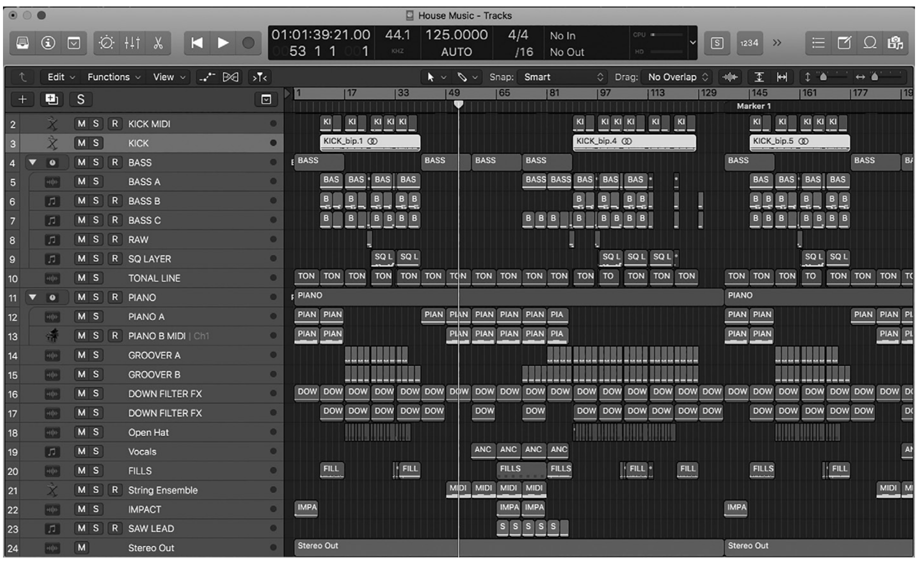

FIGURE 1.4
A typical arrangement in Logic Pro X

For example, if there is a one-bar drum loop region in the project, this region could be repeated so that it plays for any number of bars throughout. The same could then be performed with the bass and lead regions etc.

With playback initiated, a timeline moves from left to right to play all the individual channels and regions simultaneously. It is also through this arrange page that you can insert any number of processor or effects plug-ins onto any number of channels.

Plug-ins are the most significant development since the introduction of the digital audio workstation. Introduced in 1996 by Steinberg, **V**irtual **S**tudio **T**echnology (VST) permits the workstation to integrate virtual instruments and effects processors into their environment. These virtual plug-in instruments, effects, and processors can be modeled on rare or vintage analog technology and installed into a workstation at a fraction of the cost of the original hardware units.

This has opened the floodgates for thousands of third-party manufacturers to create emulations of classic effects, processors, and instruments that offer multiple benefits to producers. Whereas with hardware, if we wanted to apply compression onto five audio channels, it would require five hardware compressors, with VST technology a software emulation of the same compressor could be inserted on as many tracks as needed (computer permitting).

These virtual processors, effects, and instruments are routed internally within the workstation and negate the introduction of any noise you may experience from external cabling. Moreover, as all the parameters on a plug-in are controlled via a mouse, mix automation can be recorded. This is a form of recording any movements you may perform in the software. Much in the same way as MIDI, you can record parameter movements on a plug-in interface, edit them further and then play them back. Plug-ins are available in many formats including the original VST, VST 2 and VST 3, alongside Direct X and audio units. Audio units are Mac-only interfaces for use within their OSX operating systems. Similarly, Direct X is a Windows-only application. VST 2 and 3 are available for both platforms but are often workstation specific. Logic Pro X on the Macintosh, for example, will only accept AU plug-ins and, at the time of writing, offers no support for VST. Presonus Studio One and Ableton Live, however, use both AU and VST instruments.

THE MIXER

Once an arrangement is complete, the music requires mixing. This involves carefully balancing the volume of each channel in the arrangement so that all the channels are clearly heard. In many workstations, as soon as audio, MIDI or an instrument channel is created in the arrange window a new mix channel is introduced into the mixer.

This mixer lets you add effects, panning and change the overall volume of a channel. It displays all the associated features you would expect to find in a

FIGURE 1.5
Logic Pro X mixer

high-end mix console such as inserts, sends, aux channels, group channels, and so forth. The mixer and further DAW functions are discussed in more detail throughout this book.

HARDWARE SEQUENCERS

While most electronic dance musicians will rely on a DAW for their production, a smaller number of artists prefer hardware step sequencers. They often employ these for the majority of their creations and only turn to a DAW at the end of the creative process to arrange the music into a complete track.

The argument is that the mouse is not a musical instrument and is therefore counterproductive when creating music. Dragging audio and stamping in MIDI notes with a mouse click isn't particularly musical. But, perhaps more importantly, having so much visual feedback on a computer monitor can produce the McGurk effect.

This is a perceptual phenomenon whereby what you see will actively determine what you hear. As unlikely as this may appear, I can guarantee that you will – at one time or another – experience this effect. While adjusting an EQ (or any other plug-in processor) on screen, you will perceive an audible change only to find that the plug-in is bypassed …

A step sequencer involves a different approach than a DAW. They rarely have large screen estate and require you to input notes using a series of pads. As such, these need different production ethics, and this difference in approach means that different creative avenues can be explored. Indeed, the growing popularity

FIGURE 1.6
Native Instruments' machine

of the Elektron Machinedrum, Analog4, and Native Instruments' Maschine are a testament to how the DAW hasn't entirely replaced hardware sequencing. I'd, therefore, recommend that rather than just settling on a DAW, you also investigate hardware step sequencers to see if you find these offer a preferable working environment.

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CHAPTER 2

The audio interface

Well, it's not really hi-fi, and not really lo-fi. It's just kind of 'fi'.

Aimee Mann

An audio interface provides the all-important link between the DAW and the outside world. Most computers today are pre-installed with an internal soundcard so you could be forgiven for thinking that you can just connect your speaker monitors or headphones directly to the computer. Even though this is possible, the quality of the converters on factory-fitted soundcards are not suitable for professional audio production. To understand why we must consider the process that occurs when recording real-world analog audio into a digital device. This is the same process whether converting digital to analog (D to A conversion; DAC) or analog to digital (A to D conversion; ADC).

With any digital recording system, a sound must be converted from an analog signal to a digital format for editing and storage. This is accomplished by measuring the incoming analog signal at a number of specific intervals. These measurements are based on the two most important factors of any audio signal: time and magnitude. Figure 2.0 shows an ADC process in operation. On the time axis, the waveform is sampled a specific number of times every second. The total number of 'samples' measured every second is called the 'sample rate.'

At an absolute minimum, the sample rate must always be more than double the frequency (pitch) of the analog signal to avoid errors. This was theorized by Harry Nyquist of Bell Laboratories and is called the Nyquist Theorem. It states that to recreate analog accurately in digital form at least two different points of the analog waveform's cycle must be sampled. As an example, if the converters were to sample a 400 Hz sinusoidal waveform, they must take a minimum of two measurements. If not, the waveform could be mistaken as a different frequency altogether. This error process is shown in Figure 2.1.

This behavior produces a side effect termed *aliasing* and is a result of the converters unable to accurately represent the analog signal. Audibly, aliasing manifests

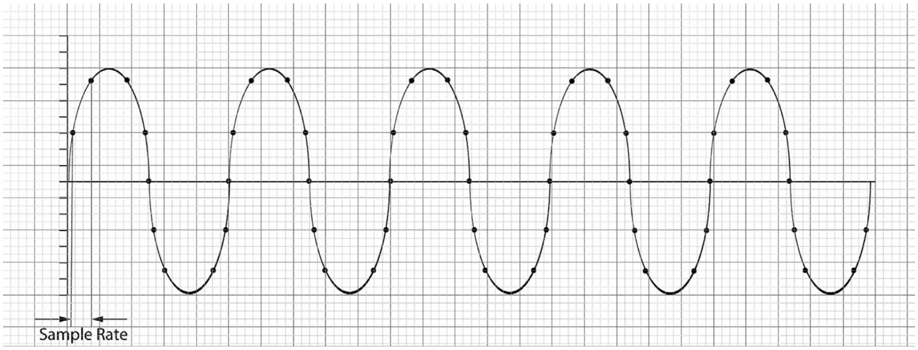

FIGURE 2.0
The sample rate

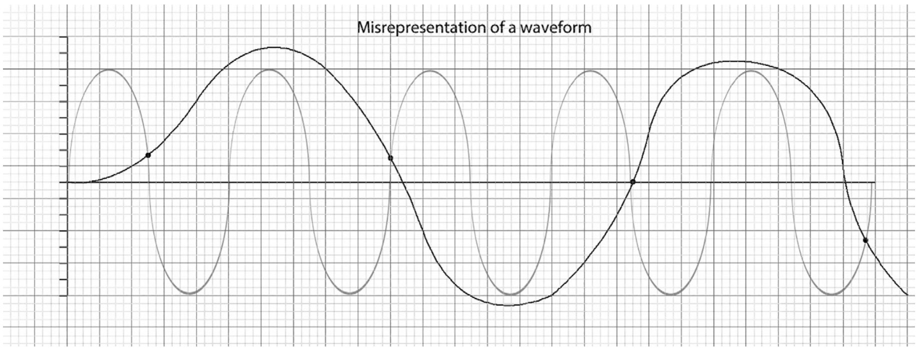

FIGURE 2.1
If two or more points of a waveform are not sampled, the audio will not be measured accurately, and the converters may determine that the frequency (pitch) of the waveform is lower than it is

itself as a series of spurious spikes in the audio and, in severe situations, can result in complete silence. As the highest range detectable by the human ear is a frequency of approximately 20 kHz, the common sample rate is just over double this at 44.1 kHz:

Human hearing limit = 20,000 Hz
20,000 Hz x 2 = 40,000 Hz + 4100 Hz (to make the rate more than twice the optimum frequency)

Although 44.1 kHz is the de facto standard for many audio interfaces and digital recording devices, some employ higher sampling frequencies of 48,000 Hz, 88,200 Hz or 96,000 Hz. While these reach far beyond the frequency response of human hearing, it can contribute to the reduction of unwanted side effects such as frequency cramping.

Frequency *cramping* occurs when a processor or effect is used on frequencies that are close to half the sampling rate. For example, if the current workstation project is 44.1 kHz and you choose to increase the gain of a wide range of frequencies at 18 kHz, the boost occurring on the high-frequency side of 18 kHz would

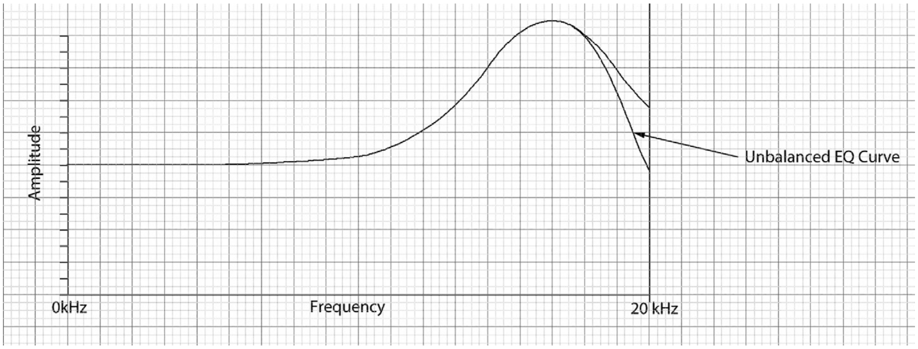

FIGURE 2.2
Frequency cramping

likely extend well beyond 22 kHz. This is more than half the project's sample rate, and can result in aliasing. The result is *frequency cramping* whereby the boost reduces sharply resulting in an uneven EQ curve.

Figure 2.2 shows frequency cramping in action. This reduces both the presence and spatial resolution of sound and becomes more noticeable when employing processors or effects that emulate analog characteristics.

Analog modeled EQs, or distortion units will introduce harmonics either side of the frequency range being processed and these are highly susceptible to cramping. Notably, many professional plug-ins will up-sample (increase the sampling frequency) of the audio entering the plug-in and then down-sample it when leaving. This allows the producer to maintain a 44.1 kHz sampling rate in the project – saving on the computers processing power – while also avoiding frequency cramping effects.

MAGNITUDE

The second measurement taken by a converter is the magnitude of the signal. This is a measurement of the amplitude and determines the dynamic range. This is used to express the ratio between two extremities. In audio equipment, the dynamic range is the ratio between the highest possible signal level before distortion occurs and the noise generated by the apparatus (the quietest possible signal).

To measure the magnitude of a signal, the ADC/DAC takes periodic measurements using a series of bits along a vertical axis. As shown in Figure 2.3, each bit equates to 6 dB of dynamic range. Consequently, a 16-bit measurement would produce a **T**otal **D**ynamic **R**ange (TDR) of 96 dB (16 x 6 = 96) while 24 bits would produce a TDR of 144 dB (24 x 6 = 144). Twenty four bits is currently the highest resolution available in audio interfaces, and many DAWs will operate at 32- or 64-bit floating point. All of these far exceed the TDR of human hearing (140 dB SPL) and provide a substantial dynamic range in the workstation.

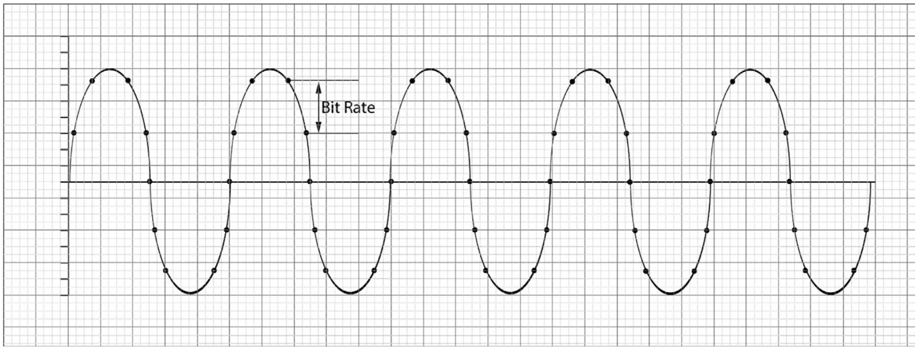

FIGURE 2.3
Bitrate measurements

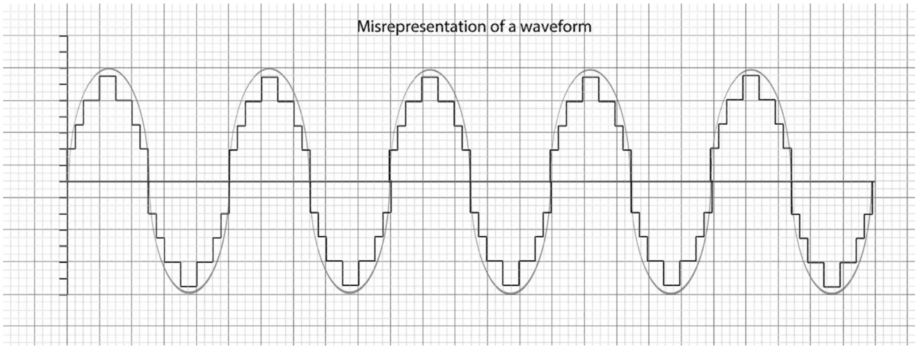

FIGURE 2.4
The effects of quantization error

While 24-bit ADC offers a dynamic range beyond that of human hearing, this measurement system is not infallible. Analog waveforms are freeform and continually change in magnitude whereas bit-rate measurements are fixed and only occur every 6 dBs. It isn't possible for an ADC/DAC to measure amplitude between two grid coordinates. Therefore, if the analog level experiences a change between the two coordinates of the bit-rate, it can't be represented accurately, and the ADC will round up to the next grid coordinate.

This mathematical rounding up can produce an inaccurate representation of the audio signals magnitude and results in quantization error. The effect of this is shown in Figure 2.4 and manifests itself as a form of digital noise. In many cases, this is imperceptible to human hearing, but as multiple audio tracks are summed together in the workstation, it can result in a mix that lacks stereo imaging and depth.

Sample and bit rate conversion occurs at both analog to digital and digital to analog stages. Therefore, even if you have no plans to record instruments or vocals, the conversion from digital to analog will also be influenced by the quality of the interface.

Internal soundcards supplied as standard on computers (Mac and PC) employ a monolithic chip design. These rely on small amplifiers built onto the microchips, termed op-amps. The voltage rails within these chipsets are limited to a few volts, and if driven hard, the voltage increases will overdrive these rails. This results in a signal deficiency at the outputs because the bit and sample rate are no longer correctly represented. The result is a reduction in the lower frequencies with smothering or masking of the higher frequencies.

In addition, soundcards that are part of the computer's motherboard often suffer from electrostatic interference and thermal noise. Both these will reduce the TDR, introduce noise, and artificially reduce the higher frequencies. This culminates in the audio output from the DAW becoming incorrectly represented.

You could compare this to working with a soft blanket covering your speakers. You do not receive a truthful representation of what's occurring inside the DAW. Since all your decisions are influenced by what you can hear, you will overcompensate for the lack of clarity by applying heavy-handed processing that may not be required.

AUDIO INTERFACE CONNECTIVITY

While there are PCIe cards available for audio work (these are specific cards that connect inside tower desktop systems or Thunderbolt enclosures), they are the mainstay of professional studios. As computers have grown smaller in form, and laptops (and Macbooks) are a favorable choice for artists who are always on the move, external audio interfaces now dominate the consumer market.

Most external audio interfaces will connect to the computer via USB, Firewire, Thunderbolt or AOIP (**A**udio **O**ver **I**nternet **P**rotocol). The decision on which to use depends on your finances, what you want to accomplish and the connections that are available on your computer. USB 1, USB 2 and Firewire are by far the slowest of the protocols and are less popular. In fact, you'll be hard pressed to find any manufacturer producing USB 1 or Firewire interfaces in the current climate.

USB3, Thunderbolt, and AOIP (Ethernet) are the most popular and fastest connections. Due to the connectivity speed, many are capable of simultaneous recording and playback of 30 or more audio channels, but the price is reflected by the number of connections they have available.

The number of input and output channels may or may not be significant depending on whether you plan to record multiple instruments or send numerous channels from your DAW into hardware effects and processors. While the quality of the AD and DA converters should always be the uppermost priority, very few interfaces offer only a stereo input and output, and many will feature a variety of connection formats.

Over the past 40 years, a large number of format and connection protocols have been developed. Of these, only a few have remained and been adopted by manufacturers. These are: analog balanced and unbalanced connections, MIDI, ADAT (aka Lightpipe or TOSLink), MADI, DSUB, S/PDIF, and AES/EBU.

When discussing both music and computers (and depending on your age), MIDI is the first connection protocol that usually springs to mind. It's the most standardized data transfer protocol and has maintained its original specifications for over 35 years. MIDI can be transmitted via USB (if the receiving hardware accepts a USB connection) or the ubiquitous 5-pin male DIN plug as shown in Figure 2.5.

MIDI can transmit data from a workstation to any MIDI capable device. These data consist of simple instructions such as note on and off, and velocity. They can, however, also be used to transmit more complex commands, including changing parameters in the receiving device through to dumping its entire memory into a DAW for recall at any time. Artists will use this form of MIDI dump to save instrument settings or presets when the synthesizer itself lacks the capability. By playing back this recorded MIDI file to the device from the DAW at a later date, any previous settings are easily recalled. This was an essential process before the introduction of the DAW and its capability to save and recall settings. In the past, the procedure was to program MIDI system exclusive (Sysex) information to each channel at the beginning of a song, so when playback was initiated, all connected hardware would recall its settings and play the song correctly.

FIGURE 2.5
A standard 'DIN' MIDI connection

Typically, an audio interface will feature a MIDI IN and a MIDI OUT port. The MIDI OUT of the interface is connected to the MIDI IN of a MIDI compatible synthesizer or sampler using a suitable 5-pin male DIN cable. Similarly, the MIDI OUT of the synthesizer/sampler would connect to the MIDI IN on the computers MIDI interface utilizing another 5-pin DIN cable. This arrangement permits a simple two-way communication from the workstation to the synthesizer and vice versa. Figure 2.6 shows this simple MIDI set-up.

Many producers begin with this small set-up, but over time increase the capability to include further equipment. Since most audio interfaces

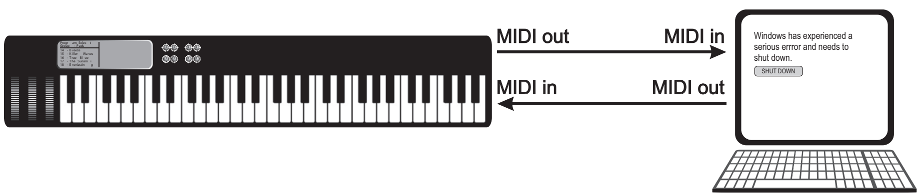

FIGURE 2.6
A simple MIDI set-up

feature only one MIDI IN port and one MIDI OUT port, means that to send data to multiple devices requires either a THRU ports or a third-party Multi-MIDI interface.

Some devices feature MIDI THRU ports. Using these – provided the device is configured correctly – any information received at the MIDI IN is repeated at the devices THRU port. In a typical arrangement, this THRU connection would be connected to the MIDI IN of the next device in the chain. Connecting and chaining devices in this manner is termed 'daisy-chaining' and it permits more elaborate set-ups as shown in Figure 2.7.

This arrangement is far from perfect, however. MIDI is a serial interface, and therefore messages are sent one after the other, so they take a finite amount of time to travel down the MIDI cable to the synthesizer. While this delay may not be discernible on the first few devices in the chain, if the data from your DAW are destined for the tenth device in the chain, they have to travel through nine other devices beforehand. This can result in the information arriving late, resulting in a noticeable delay. This delayed response is called *latency*.

Latency is a problem we all have to deal with whether employing MIDI or not because it also occurs with audio. All signals take a finite amount of time to travel around a system, so if you are recording a vocalist and listening to it on your computer, you will experience it, too. The audio signal has to travel from the microphone through the ADC into the DAW. The DAW then sends this signal back into the audio interface, and the DAC converts the signal to your speakers so you can hear it. This action results in latency, and is measured in milliseconds. Even as little as 20 milliseconds of latency can be discernible.

To avoid this, some audio interfaces offer zero latency monitoring. This works by letting you listen to the signal being recorded before it enters the ADC/DAC conversion process. If this is not available, then the only other option is to adjust the buffer size. This is the amount of time that is allotted for the audio interface and DAW to process the incoming information when recording. The shorter this

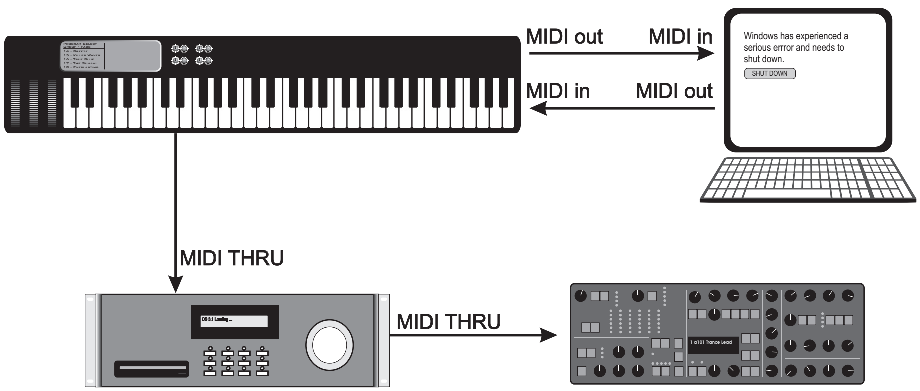

FIGURE 2.7
A more elaborate MIDI set-up (daisy-chaining)

time is, the less latency you'll experience but shorter times will put a more significant strain on your computer's processing power so it can be a delicate balancing act. The buffer size is usually accessible in your DAW's configuration settings.

Unfortunately, there is no such user-definable buffer size with MIDI. The only solution is to employ a multi-MIDI interface. These are external hardware interfaces connected to the DAWs host computer via USB and feature many individual MIDI IN and OUT ports. This can range from as little as two MIDI OUTS and MIDI INS through to four, but, more commonly, it is eight. By employing these, the DAW can utilize each separate MIDI output to feed different devices. This

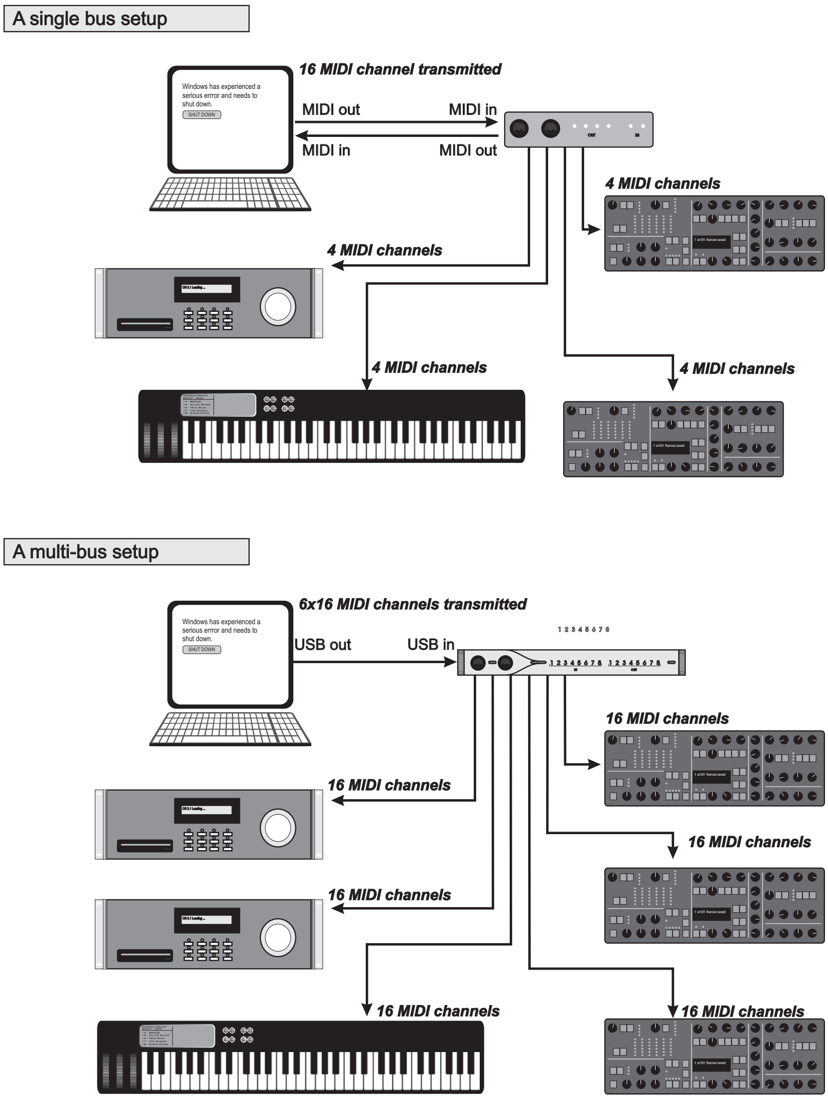

FIGURE 2.8
A multi-MIDI set-up

prevents the need to daisy-chain devices together but also permits each device to accept 16 channels of data if required.

Due to the vast resources available in computers today, most plug-in instruments are single channel. That is if you wish to use the same instrument again in your arrangement you can just open up another instance of the plug-in.

This same idea exists in many hardware synthesizers, although it occurs internally. They contain 16 instances of the same synthesizer engine and therefore it is possible to employ 16 different voices from the same synthesizer. This multi-timbral option permits a DAW to transmit 16 channels of MIDI to the synth, each destined to play a different sound. In this set-up, channel 1 could play a MIDI piano melody, and channel 2 could play a bass melody. The receiving device would have a piano sound set to channel 1 and a bass sound set to channel 2, so when the signals are received both piano and bass play together. Some plug-in instruments will also offer this multi-timbral operation.

AUDIO CONNECTIONS

All audio interfaces feature analog audio outputs or/and inputs in the form of unbalanced or balanced 1/4″ jack sockets, XLR or DSUB connectivity. The difference between a balanced and unbalanced cable is determined by examining the termination connections at each end of the audio cable.

Cables that are terminated with a mono jack or phono connectors are unbalanced. Those that terminate as a stereo TRS (**T**ip – **R**ing – **S**leeve) jack connection or an XLR connection will likely be a balanced connection – if used on a mono signal. Examples of TRS, XLR, and mono jack are shown in Figure 2.9.

A majority of professional hardware equipment – except most hardware synthesizers – employ balanced connections. This is because in many professional studio applications the equipment is spread around the room and requires long cable runs. This makes the audio signal traveling through the cables susceptible to **E**lectro **M**agnetic **I**nterference.

EMI is a result of electrical current flowing through a conductive device. As the current travels along the cable, it creates a magnetic field. The larger the current, the larger the electromagnetic field becomes. To further compound the issue, the higher the resistance of a cable, the higher the flow of the current must be to fight the resistance, which, in turn, increases the amount of electromagnetic interference. As resistance will increase with the cable length, or as the cable thickness decreases, it poses problems for many studios because cable runs are commonly bound together to run down a single conduit. The electromagnetic interference increases as it travels around instruments, through to earth and results in hiss, loss of high-frequency detail, and a low-frequency hum termed *ground hum*.

To prevent this, balanced cables contain three wires within a screen. One wire carries an earth that is grounded at both ends to avoid radio interference, while the other two are used to transmit the audio signal. Using two wires to transmit

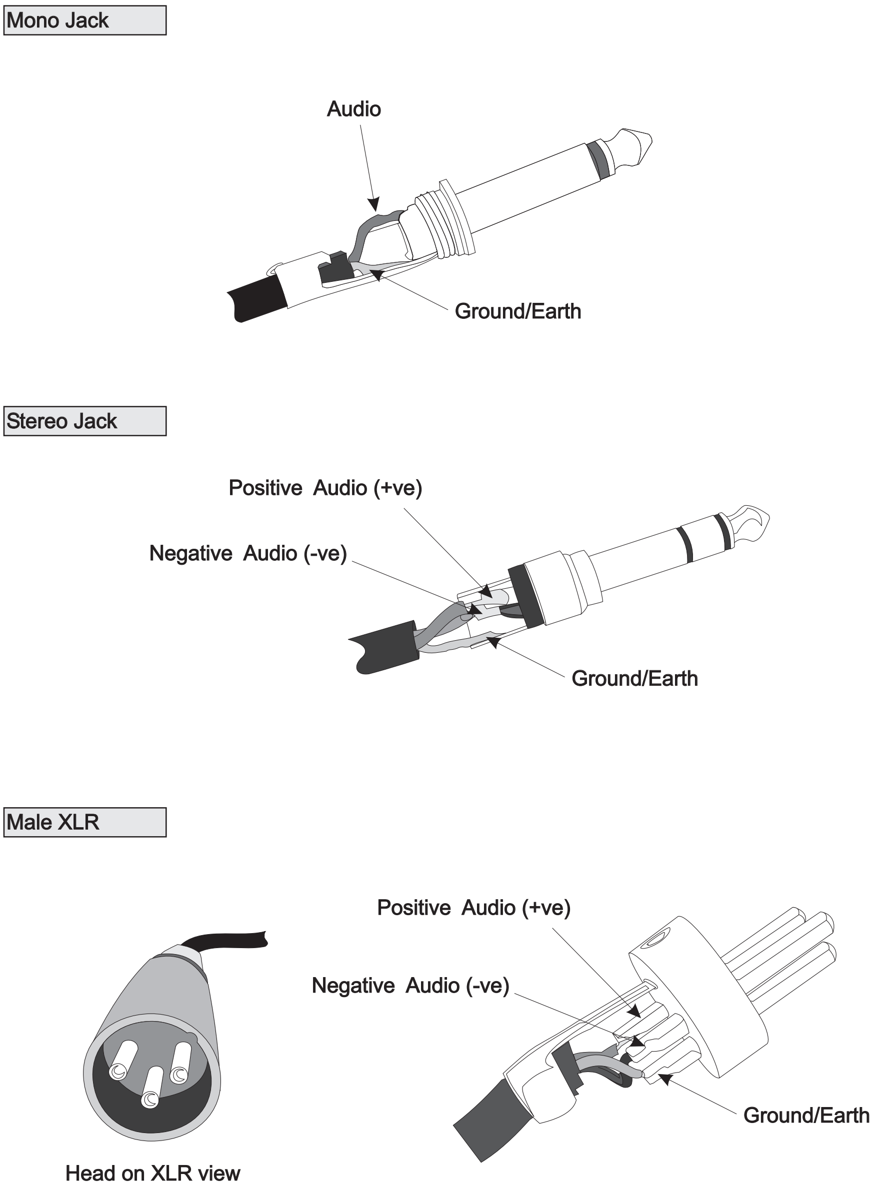

FIGURE 2.9
Mono, stereo jack, and XLR connectors

the audio signal, one can be phase inverted at the source so that when the signal arrives at the receiving device, it can then be put back into phase and summed with the original signal. By doing so, any interference that was introduced by the cable is canceled out. This is because when both the signals are put back into phase, any extraneous signal that may have been picked up while traveling through the cable is removed.

This interference cancellation does not occur with unbalanced connections. While unbalanced wires and connections do contain an earth terminated at both ends to prevent radio interference, only one wire is left to carry the audio signal. This makes them more susceptible to EMI and ground hum. Consequently, in any studio set-up that utilizes unbalanced equipment (such as many hardware synthesizers), they should be kept in proximity to the mixing desk or an audio interface to reduce the length of the cable run. They should also be kept away from any other possible sources of electromagnetic interference. This includes power cables (these create an even more significant electromagnetic field due to higher current), transformers, microphone pre-amplifiers, and loudspeaker amplifiers.

FIGURE 2.10
Tascam 25 DSUB connection

DSUB CONNECTIONS

DSUB connections differ slightly from the aforementioned balanced connections. These transmit eight balanced analog channels through a single hefty cable. Introduced by Tascam, the DSUB utilizes a 25-pin 'D' style male or female connection, and this is employed in multi-channel professional equipment to permit easier interconnection between equipment.

A typical example of this is with many multi-I/O professional audio interfaces and the inputs on many mixers. As these can employ multiple inputs or outputs, it is easier to use DSUB connections than row on row of jacks or XLR sockets. This not only helps reduce the size of the device but also makes multiple I/O connectivity easier to manage.

DIGITAL INTERFACING

Digital interfacing is more complicated than analog because the transmitted audio data must be encoded and decoded correctly by both connected devices. This means that the bit rate, sample rate, sample start and end points, and the left and right channels must be coded in such a way that the receiving device can make sense of it. Additionally, there are many digital formats available, and none of them is compatible with one another without the need for further digital conversion interfaces.

Nonetheless, digital connectivity does provide a preferable alternative to analog connectivity because it doesn't introduce noise, it is not susceptible to electromagnetic interference, and many of the formats are capable of transmitting and

FIGURE 2.11
Toslink connections (aka Lightpipe)

receiving multiple channels of audio simultaneously down a single cable. The most standardized digital connection that appears on many interfaces is ADAT Lightpipe.

Alesis developed Lightpipe for use with its digital multitrack recorders but over time it became a recognized industry standard. Lightpipe employs fiber-optic cables that terminate at each end with Toslink connectors. It's a format capable of transferring up to eight channels of uncompressed audio up to 48 kHz and at 24-bit resolution. Higher sample rates of 96 kHz are possible using a process called SMUX, but in this mode, the Lightpipe only transmits a maximum of four channels.

Similar to the USB interface, Lightpipe is a hotplug device. This means that the sending and receiving devices do not need to be switched off before connections are made. I would, however, strongly advise either muting both devices or turning the speakers off because connecting while these are on can result in substantial signal spikes.

Lightpipe is the most popular format in the consumer market because it offers a simple, inexpensive way to transmit and receive multiple audio channels. With just one thin cable, eight channels of audio can be transferred to or from a device digitally. Lightpipe, however, is a one-way communication protocol and cannot send and receive data in the same pipe; it is either one or the other. Many audio interfaces will feature two Toslink connectors. One connection will be used to output eight channels while the second connection will be used to receive eight channels, thereby permitting simultaneous I/O connectivity.

The ease and relatively cheap application of Lightpipe enables many manufacturers to add this connectivity to their interfaces. This is something to be cautious of, however. It can be tempting to purchase an interface providing 26 channels of I/O only to find that only two of the outputs are analog and the rest is made up of ADAT connectivity. While this format can be used to connect the interface directly to another digital device, if your equipment doesn't have ADAT, you will need to purchase further standalone converters, and this can substantially increase the cost.

S/PDIF

A digital connection similar to Lightpipe is S/PDIF (Sony/Philips Digital Interface). This is often mistaken for Lightpipe because it can employ the same TOSLink connection as Lightpipe. While the cable is the same, the format is different and only permits the transmission of two channels of digital audio (commonly a stereo L/R pair) or compressed surround sound formats such as Dolby Digital.

S/PDIF used to feature on many of the samplers and synthesizers in the early 1990s but more recently remains confined to smaller audio interfaces and consumer hi-fi and satellite products.

AES/EBU AND MADI

The audio professional alternative to S/PDIF is AES3. This is more commonly known as the AES/EBU connection. It uses a 3-pin XLR connection, similar to the balanced analog equivalent. In fact, you can use an AES digital cable to transmit analog audio between analog devices if you find yourself in a desperate situation. While it is also possible to use a standard analog XLR cable to transmit AES digital, it is not something I would recommend as it's likely to introduce faults.

Similar to the S/PDIF format, AES/EBU can only transmit two digital audio channels simultaneously, but it does provide a reliable connection over long cable lengths and was for a long time the preferred connection in many studios. However, being limited to only two channels, it again makes multichannel audio transfer challenging to employ in hardware devices, and therefore most multichannel devices now use the MADI format.

Multichannel **A**udio **D**igital **I**nterface (MADI) or AES10 was developed by Sony, Solid State Logic, Mitsubishi, and AMS Neve. MADI permits the transmission of multiple audio channels simultaneously at both high bit and sample rates. It is capable of transmitting up to 64 channels of audio at 24-bit and 96 kHz at lengths of over 100 meters. Consequently, it has become a standard in most professional studio devices and is available on many of the higher end manufacturers such as Allen & Heath, Solid State Logic, AMS Neve, Yamaha, Avid, and Ferrofish.

A PC or Mac installed with a MADI interface can be connected directly to a MADI interface via a small dual cable. This configuration can offer simultaneous input/output streaming of up to 64 audio channels to and from a DAW. This could be used to connect the DAW to a mixing desk, or to a high-end audio converter for connection to racks of hardware. The connection standard is, however, limited to professional equipment and therefore remains relatively expensive.

FIGURE 2.12
A BNC Connection

AOIP

Audio **o**ver **I**nternet **P**rotocol is a relatively new standard for the audio industry and permits transfer of audio via a standard CAT5e or CAT6

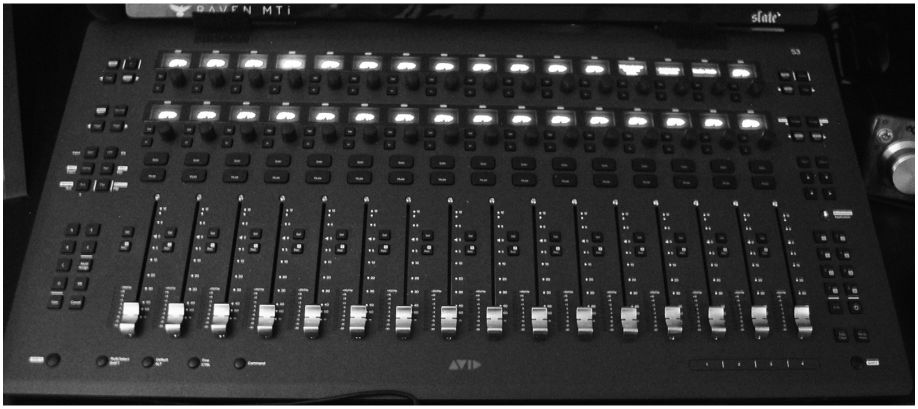

FIGURE 2.13
The author's Avid S3

Ethernet cable. Like MADI, AoIP can transmit and receive multiple channels of audio at various sample and bit rates but with the added benefit of also sending additional data. This could be channel name information or even various fader movements. The Avid S3 controller interface, for example, transmits four channels of audio to and from the connected computer alongside any moves on the faders or pan controls.

Because the protocol employs IP packets the number of channels that can be transmitted and received depends on the network speed. A 100Mbps network would only be capable of sending and receiving 32 channels of audio, while a 1000Mbps network could handle up to 64 channels of audio. The current leader in the field of AoIP is Dante from Audinate. However, like MADI, it is primarily aimed at the professional market and can be expensive to implement.

DIGITAL CLOCKING

Employing digital connectivity into any studio setup involves more than just interconnecting digital cables between compatible devices. For any two digital devices to communicate with one another, they must be *clocked together*. Clocking ensures that the devices communicate reliably, and on time with one another. For this, one device is configured as the master and all other digitally connected devices act as slaves.

With this arrangement, the master sends out a word clock signal that is received by all the slave devices. The word clock informs all receiving devices of the current clock rate, the left and right channels, and ensures they all remain in sync with one another. Without this clock, the devices become confused as to when a signal is expected. This results in jitter, whereby any number of audio artifacts may reveal themselves. This includes spurious spikes, a lack of stereo definition or detail, and sometimes an ear-piercing racket.

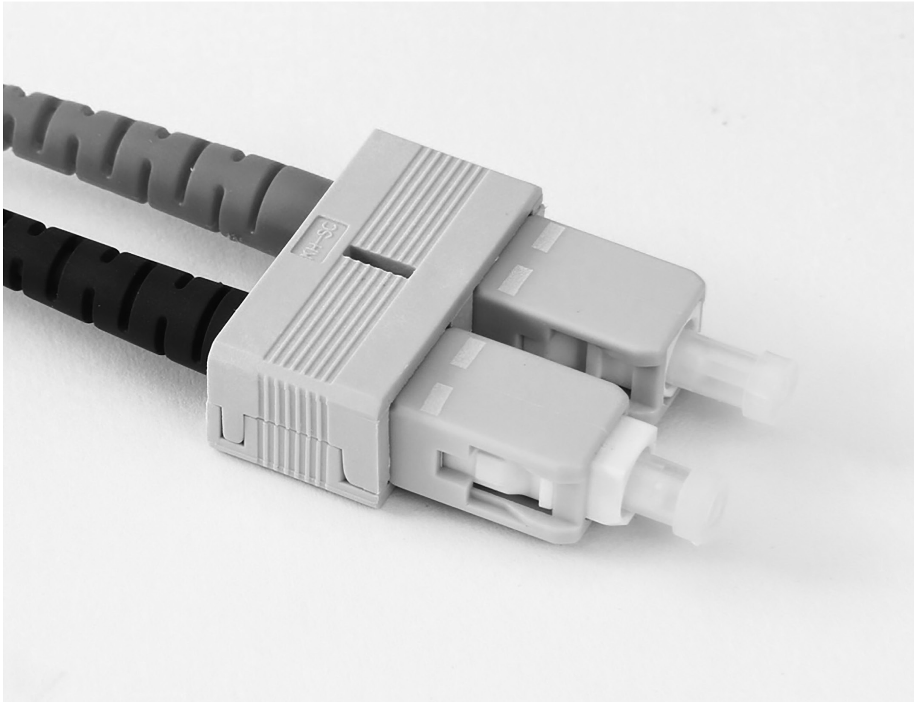

FIGURE 2.14
An Optical MADI Connection

Although you can commonly transmit clock alongside the audio channels via most digital cables, a preferred approach is to deliver the word clock via a separate designated cable because it's more reliable. This independent analog clock signal is provided via a cable fitted with BNC connections, and many interfaces will feature this connection format.

Some higher spec devices will feature two or three BNC connections, an input, an output and sometimes a through. The last allows mirrors of the word clock received at the input to the through port so that devices can be daisy-chained together.

If a device doesn't feature a through port, it is possible to employ a BNC splitter. These are designed to accept one input signal that is divided across a small number of outputs. Although these distribution units are typically designed for surveillance camera use, if the word clock signal is only to be distributed to a small number of devices, it's a cheap alternative.

It should be noted that a word clock signal grows sequentially weaker as it travels through different devices and long cable lengths. Moreover, the device sending the master word clock must be carefully chosen because not all word clocks are genuinely accurate. Minor amounts of deviation can result in jitter that, while not immediately perceptible to human hearing, can result in a weak or unbalanced stereo image.

If the studio relies heavily on digital communication and a number of devices are to be clocked together then a preferable alternative is to employ a word clock generator and distribution device. The clock signal generated by these is commonly far more accurate that other digital devices and they also feature multiple BNC outputs. In my experience, the best example of a standalone word clock generator/distribution device is Antelope Audio's 10MX or the Isochrone OCX.

CHAPTER 3

Monitors, headphones, and your room

If you can't hear it you can't fix it.

Rick Snoman

To create music, you need to hear it. As obvious as that statement may be it's riddled with challenges for the producer. We face multiple problems when it comes to listening to music in our studios because – regardless of the audio interface – the sound from our DAW isn't always going to be accurately represented by the speakers.

For example, if you're using a pair of speakers that exhibit a dip in the midrange frequencies, you'll naturally compensate for this when working on your music. And while the music will sound fine on your speakers, if it is played on a different pair at a different location, the midrange may then seem overly present.

To ensure we don't face this problem, we should employ speakers that exhibit a neutral frequency response. These are known as studio *monitors*. There is a bewildering choice of studio monitors available, and prices can range from as little as £150 to over £15,000 *per* monitor, so price vs. performance often determines the choice.

Monitors are available in nearfield and midfield designs. Midfields are intended primarily for professional studio installations and positioned approximately three to four meters away from the engineer's listening position. For these, you obviously require a sizeable working environment, but the room must also be accurately treated. If not, the sound leaving the monitors strikes the walls, and their subsequent sonic reflections have an adverse effect on what you hear.

Many semi-professional producers use nearfield monitors. These are positioned closer: approximately 1.5 to 2.5 meters away from the listening position. Being closer to the sound source eliminates more (not all) of the sonic reflections and provides a more accurate impression of what's going on in your mix. The downside is that they don't replicate bass as well as midfield monitors. You can increase the bass presence with the addition of a subwoofer, but these often create more problems than they solve.

A subwoofer is an additional mono speaker designed to handle all the low frequencies. The subwoofer will deliver all frequencies typically ranging from 20 Hz to 120 Hz leaving the main studio monitors to reproduce the rest of the frequency spectrum. While this can provide a great solution – particularly if the monitors do not have an extended frequency response – in a small, untreated room, the low frequencies produced by a subwoofer result in standing waves and interference. Standing waves are a result of two waves of the same frequency, amplitude, and wavelength meeting one another from opposite directions. This occurs because the waveform leaving the subwoofer meets the previous waveform reflected from another surface.

The effect is comparable to that of dropping a pebble into a fish tank. As the stone hits the water, waves propagate outwards in all directions. But, as soon as these waves strike the glass sides, they are reflected back where they meet other waves, producing a more substantial disturbance of the water. With sound, when reflected waves meet a successive wave travelling in the opposite direction, this results in an exaggerated response at differing frequencies. This manifests itself as a serious increase in bass frequencies that you will overcompensate for during production and mixing. Consequently, subwoofers should be avoided unless you acoustically treat your room to reduce these reflections.

ACTIVE OR PASSIVE?

When purchasing monitors, perhaps the most critical consideration is whether the monitors are active or passive. Passive monitors require the use of an external amplifier. The audio signal leaves your audio interface and enters a standalone amplifier that increases the signal level before it is passed onto the monitors.

With passive systems, it is essential to match the amplifier to the monitors correctly. It should be evident that if the amplifier is overpowered, you could damage your monitors, but the same is true for an amp that is underpowered. If the amp runs out of available headroom, it will distort, and this distortion can result in more severe damage to your monitors.

It's for this reason that many producers opt for active monitors. These have matched amplifiers built into the monitor itself, so there is no requirement for a standalone amplifier. The audio signal can leave your audio interface and run directly to the monitors. The disadvantage of this approach, however, is that they are more expensive, and each monitor requires power from a wall socket.

Many active monitors are 2-way ported (aka reflex). These consist of two speakers, a woofer to reproduce the low frequencies, and a tweeter to replicate the high. There are also 3-way ported speakers that consist of three speakers, a woofer, midrange, and tweeter and while these can often outperform 2-way, they're also considerably more expensive. Both models have a hole in the rear or

FIGURE 3.0
2-way ported vs
3-way ported

front of the cabinet called a *port*. These ports are installed and tuned to control the frequencies that emanate from the back of the speaker.

BAFFLES AND PORTS

Sound is generated by the speaker cone moving forward and backward. However, sound doesn't just emanate from the front of the cone, it equally emits at the rear, too. If these front and rear signals mix with one another, it results in the phase problems discussed earlier.

The ideal solution would be to enclose the rear of the speaker in a sealed cabinet. This would prevent sounds from escaping at the back of the speaker and produce an accurate representation of the audio signal from the front. Although there are some sealed cabinet monitors like this, such as the infamous Yamaha NS10M, this *infinite baffle* system isn't an ideal solution.

In a sealed system, when the speaker cone moves, the air inside the speaker cabinet will go through a compression and rarefaction cycle. The air inside the cabinet will be compressed when the cone moves inwards, and will decompress as it moves outwards. The heavier the bass signal, the further the cone needs to move, and if there is too much air resistance inside the cabinet, the movement of the cone is restricted. This means that the reproduction of low frequencies – although accurate – is limited.

An alternative solution is to port the cabinet. This prevents air compression from restricting the speaker's movement. Moreover, the speaker can expel controlled frequencies at the front (or rear) of the cabinet. These frequencies mix with the

FIGURE 3.1
Genelec 2-way ported monitor – one of the more popular studio nearfields

sound from the speaker mounted in front of the cabinet resulting in exaggeration of bass frequencies. By doing so, smaller speakers can reproduce a lower bass frequency than they could unaided.

The problem, however, is if the port is poorly tuned as they are on many budget monitors. If there is too much or too little air resistance, it results in poor time dominance effects or excessive phasing. The result is an inaccurate reproduction of the lower frequencies. Tuning and development costs are not cheap, and this is why ported monitors can cost more than £1000 each.

MONITOR POSITIONING

The positioning of your loudspeaker monitors is just as important as the model you choose. Placing monitors close to a corner or wall will increase the perceived bass by 3, 6, or 9 dB because of the boundary effect. As discussed, sound does not just emanate from the front of the speaker; it also comes from behind (whether rear ported or not). If a speaker is positioned close to a wall or corner, the sound leaving the rear of the speaker will reflect from the wall and mix with further signals leaving the speaker. These reflections lead to phase cancellations at differing frequencies producing an inaccurate response.

Ideally, monitor speakers should be positioned at least 12″ (about 30 centimeters) away from any hard surfaces, and, if they're placed on a desk, isolated with absorption panels such as Auralex or Primacoustic recoil stabilizers. This is because as the speaker monitor vibrates with sound, the vibrations are transferred onto the desk. The desk then resonates and acts as a sounding board, producing an increase at the lower to mid frequencies. This is known as *vibration loading*.

It is equally essential to position the monitors to form an equilateral triangle between you and the speakers. That is, the distance between each monitor should be the same as the distance from each monitor to the listener's ears. The monitor should never face directly to the rear of the room but instead, be angled so that the speakers are aiming towards your ears when you sit in the 'sweet spot' (listening position). The height of the monitors should also be adjusted so that the tweeter is at ear level. Higher frequencies contain more directional information than low, and if your ears are at the same height as the woofers, you will be more likely to mix the bass lighter and the higher frequencies heavier. The ideal monitor arrangement is shown in Figure 3.3.

FIGURE 3.2
Primacoustic recoil stabilizer

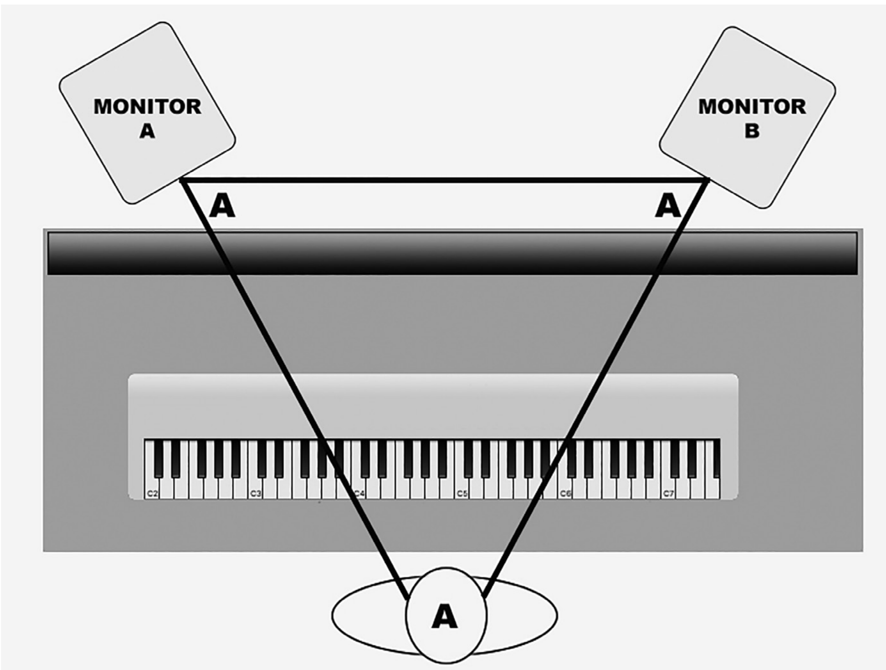

FIGURE 3.3
Monitor positioning

MONITOR CALIBRATION

The above guidelines should produce the best starting position for monitoring, but you should also calibrate the speakers. Calibration requires an SPL meter, but if you own a smartphone, there are plenty of SPL apps in the Android and App Stores. To begin, you'll need to generate a test tone. Many DAWs have a test tone oscillator available as a plug-in, and you can use any oscillation. I prefer to use pink noise because it is not as ear piercing as many of the other oscillations.

Playing the test tone, hold the SPL meter at ear height where your head would typically be. Mute the left speaker and measure the right speaker. Make a note of the measurement and then mute the right speaker and take a measurement of the left speaker.

The SPL reading should remain the same for both monitors. If it isn't, you'll need to adjust the volume control on each monitor. If the monitors are active, each will have a volume parameter somewhere on the rear. If they're passive, you may have to adjust your DAW or audio interface output in order to level the two monitors. This will ensure that when mixing, your stereo field is accurate and evenly weighted.

With the stereo field correctly weighted, the next calibration should be to ensure consistency in gain from your DAW. To begin, you'll need to set up a pink noise test tone and ensure that the output meters on your DAW are registering this signal at 0 dBFS. Set the outputs of your audio interface to unity gain. This is commonly the midway setting or level at which the signal from the audio interface isn't attenuated or boosted.

Having an assistant hold an SPL meter in your listening position, turn off the left speaker and decrease (or increase) the volume of the right speaker until the SPL meter reads 76 dB SPL (C weighted). Now do the same with the left speaker. Turn the right speaker off and adjust the volume of the left speaker until the SPL meter reads 76 dB SPL. Turn on both speakers and the SPL should increase by 3 dB to 79 dB SPL. Your speakers are now calibrated.

> The standard calibration level is 85 dB SPL, but this is often too loud for a home studio.

ROOM ACOUSTICS

A final hurdle we must overcome with monitoring is the room itself. The problem we face with mixing in a typical room is that every object and surface in that room will influence the sound from the monitors. As discussed previously, the effect a room can have can be compared to wave propagation from dropping a pebble in a fish tank.

Sound waves propagate outwards from the monitors, and these will strike the walls, ceiling, floor, windows, and any furniture. These sonic waves then reflect back, and where these reflected waves meet further waves emanating from the speakers, there will be a peak or dip in frequencies. This creates all sorts of problems from differing phase relationships through to exaggeration of the bass, mids, and highs to everything in between. To reliably monitor your mix, the room should be adequately treated to prevent any wayward acoustics from influencing your decisions.

Acoustic treatment of a room is a complex and multifaceted subject. It's a topic that could digest any number of books (and often does) and even then, you cannot rely on the theory alone because the formulas for calculating the problem 'modes' within a room are based on the walls, floors, and ceilings exhibiting specific acoustic properties. So, what follows should be considered a broad generalization of how to reduce the most common modal problems you'll experience.

Dealing with reflections consists of more than sticking a few egg boxes to your walls. Indeed, just randomly placing acoustic panels, foam panels, or egg boxes around a room will likely make the situation worse. If you plan to put any acoustic treatment tiles on the walls, I strongly recommend you consult a professional or research the topic. But before contemplating covering the walls with treatment, a number of the problems can be reduced with some forethought.

First, you should cover or remove any hard reflective surfaces from the room. If there are large windows, the glass will be highly reflective. Therefore, it is advisable to cover them with large heavy curtains. The heavier the curtains, the better, as they will absorb and reduce the reflections returning to the room. Similarly, any mirrors or pictures on the walls should be removed, and if you have a hardwood or laminate floor, this should be covered with a large rug.

Small rooms are considered to have fewer modal problems than larger rooms because, in smaller areas, the lower frequency waves do not have the time to develop fully. Larger rooms are more likely to suffer from effects such as standing waves. This means the more extensive the room, the more acoustic treatment will be required.

The first step should be to add bass traps. Low-frequency waveforms are long and require time to dissipate adequately. Any corners will capture the bass and produce an inaccurate low-end response in the room. Placing quality bass traps in every corner of the room (front and back) will provide the most immediately noticeable results as the bass begins to come under some control.

After bass traps, you should direct attention to the mid and high frequencies by taking care of the *first reflection* points. You can find these by sitting in your listening position with an assistant holding a mirror on the right wall. Have the assistant slowly move the mirror along the wall, moving further and further away from the monitor. When you can see the monitor in the mirror – that's your first reflection point. This process should be repeated on the left-hand wall, and in most cases, you'll find that they occur in the same position. Finally, have the assistant move the mirror across the ceiling. When you can see both monitors in the mirror, that's the first reflection point on the roof.

In each of these reflection points, it is advisable to mount some acoustic tiles or panels. The area to be covered should roughly be around 2ft x 4ft (61 x 122 cm), and the tiles should be 50 to 75mm thick. By applying acoustic deadening to these areas, it should reduce flutter echoes and produce a reliable response from the monitors. Just these few necessary steps are commonly enough treatment to receive an acceptable sonic response from a room. But, as with everything in audio, you can go as far as your wallet (or partner) permits.

If your studio happens to be in a bedroom or a spare room, then your partner/housemate/landlord may not be tolerant of gluing 'foam all over the newly decorated walls.' In this case, the only alternative would be to employ room correction software such as Sonarworks Reference.

This is software that comes supplied with a measurement microphone. The software generates some test tones from your monitor speakers. These are measured by the microphone that is placed in multiple positions around your room for each test. Once the analysis is complete, the software calculates the resonances and modal problems in your room and generates an EQ profile. You can then place this EQ plug-in on your master DAW output, and it reproduces a flat frequency response. I've used this in the past and, while not the ideal solution, it's certainly better than nothing at all.

HEADPHONES

With all the problems associated with monitors, you could be forgiven for thinking that headphones offer the best solution. While they do eliminate some of the issues related to your room, they also introduce deficiencies. Like monitors, headphones come in different designs and price ranges. And like monitors, good headphones are not cheap. When choosing a pair of headphones, the most significant decision you'll face is whether to go for circumaural (over the ear) open or closed back headphones.

FIGURE 3.4
Closed back headphones (Sennheiser HD280s)

Closed back headphones are designed with a sealed outer shell. This directs the sound directly to your ear. These eliminate outside noise and produce a quiet listening environment so you can hear the low-level detail. This is because like monitors, the air behind the speaker goes through a series of compression and rarefaction cycles that results in a defined cone movement. Just like infinite baffle monitors, however, they have a limited bass response.

Typically, closed back headphones are the preferred choice for musicians or talent in the vocal booth as they have little to no sound leakage.

However, they are not ideal for monitoring or mixing due to the smaller soundstage and limited bass response.

Open back headphones do not have a sealed outer shell and instead employ a grill. On some headphones, it is possible to see the rear of the speaker through this grill on the outside of the headphone. Because of this open back design, there is no air compression, and so they produce a more robust soundstage with a lower bass response. However, like ported monitors, if they are not accurately tuned, you will receive an inaccurate low-frequency response. Nevertheless, a good pair of open back headphones are the number one choice for many producers for monitoring and mixing duties.

While headphones eliminate problems with room modes and standing waves from a monitoring environment, they suffer from a different assortment of issues.

The same as our two eyes produce depth perception; our two ears create an effect known as binaural cross-talk. This effect occurs when the sound from the left monitor speaker reaches our right ear, and the right speaker reaches the left. Our brain deciphers this cross-talk as music emitting not only from the front left and right but also the center. This is known as the *phantom center*, because there is no speaker in the center of a stereo field. We lose this perspective with headphones because sounds only occur directly to the left and right. Without the cross-talk, it makes it extremely difficult to determine the spatial area and depth of the music accurately.

FIGURE 3.5
Open back headphones (AKG K712)

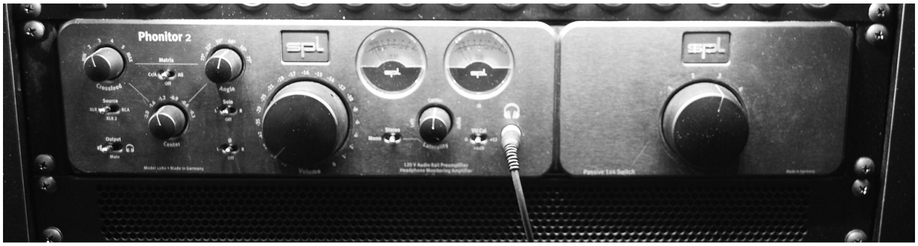

FIGURE 3.6
The author's SPL Phonitor 2

To combat this problem, SPL produced the Phonitor 2. This is a monitoring solution that features a cross-talk matrix system. This attempts to emulate the binaural effect, allowing you to configure the listening angle and adjust the cross-talk amount between the left and right channels. (The SPL matrix calculates time, level difference and frequency response in order to mimic speaker playback.)

A secondary problem we face on headphones is the bass response. The speakers in headphones are only small, and although some models can replicate bass with a fair degree of accuracy, they do not exhibit the power experienced from a larger monitor. Therefore, it is easy to overcompensate for the bass, and this results in a bass-heavy mix.

To further compound the situation, when wearing headphones, there is a difference in pressure created within the ear canal. While this can prove beneficial for lower frequency sounds, as the frequency increases above 500 Hz the outer ear begins to resonate. The closeness of the headphone cup results in further pressure changes within the ear producing a different perception of frequencies and their respective loudness.

Because of this, the ideal situation is to use a mix of both monitors and headphones. By switching between the two on a project, the headphones can make up for the weakness in the monitors and the monitors will make up for the problems with the headphones.

EAR TRAINING

Aside from having a great sounding room, monitors, and audio interface, the single most crucial factor in producing professional music is the ability to listen critically. This is a necessary skill that is severely understated and undervalued.

It is a topic that few like to discuss because there are no cliff notes or shortcuts. It's a skill that must be learned, and that takes time and dedication. Many plug-in manufacturers take advantage of this by producing visual aids to mixing or mastering, some that can help a little, but *none* of them will *ever* replace a good pair of trained ears. Just as a musician spends years learning to master an instrument before they can play anything that sounds good, with audio production you can produce results almost immediately but they won't sound professional until you develop your hearing.

The first challenge is to develop your critical listening skills and improve your auditory memory. This is the ability to identify frequencies and how long these frequency responses remain in your memory. It is more than the ability to listen to a piece of music and remember the melody or the lyrics. It's the capacity to listen to a piece of music, make a note of the frequency balance of instruments, the effects, and processing, and commit it to memory. So much so, that when repeated, you're capable of picking out subtle frequency or tonal differences between the two. As simple as this may sound, it's particularly tricky and for many the length of this form of memory will be less than one second.

The second challenge is to develop your analytical memory. This is the ability to hear, identify, and introduce feeling and meaning into music. Just as we have developed skills to interpret the meaning and emotion behind spoken language, we must do so with music. For example, 'what are you up to?' could be translated differently depending on how it's delivered. If it is said in a relaxed manner, it is received as a polite friendly interest, but if it's delivered sternly, it will be translated as an aggressive accusation.

The meaning and emotional content of a piece of music is the essence of music production. It's our job to select the appropriate effects, processors, microphones, pre-amplifiers, and synthesizers that deliver the right feel suited towards the music. And if we are unable to successfully accomplish this, we either bore or confuse our listeners. For example, an energetic deep house track requires a different sonic approach than a soft R&B ballad.

> If you're unsure on your auditory skills, you can test it using the files on the DMM resource kit available to download from http://www.dancemusicproduction.com.

Developing both auditory and analytical memory is accomplished through repetition. Just as you have learnt communication skills over the years, you must do the same with all other areas of music.

The emotion of music is often a result of the timing, colour, and dynamic behaviour of the instruments. Listening closely, and repeatedly, to many different genres of music will increase your awareness of how a sounds tonal colour and dynamic manipulation establishes an emotional picture in a listener. For example, a fast paced melody with short notes, occurring close in pitch to one another, treated to distortion and excessive audio compression will produce a sonic soundscape that appears more aggressive than a slower melody without distortion and no compression. But, as we all perceive music differently, it is important to develop your own perception of what contributes to the emotion of the music. As you produce music over the years, this will naturally begin to improve. It is often a result of your hearing developing and the ability to hear what makes a sound alongside other skill sets advancing.

It is challenging to develop hearing for processors and effects until you begin to use them in context of a project. And, for some processors, such as compression, you *will* experience difficulty hearing what it does until you've developed an excellent auditory memory for dynamics. This takes time and is very often one of the last skills developed by many engineers (even though it contributes to the emotion of the music).

Learning frequencies is more straightforward, however. There are many software applications designed to help you learn the different frequencies. My favorite is *Train Your Ears*. Although the software is not free, it should be considered a worthwhile investment in your future as a producer. You can save an additional 20% off this product using the code available in our resources package.

In the meantime, you can start training your ears with little more than listening intently to music everywhere. Listen beyond the lyrics and melodies and to the individual sounds. Pick out an instrument or vocal and ask questions such as what effects are applied to the voice; does it sound close or far away? Does the voice change in tonal character? Is it bright? Is it dull? Does it delay? Does it change in the *drop* or *breakdown*?

While these are simple questions, if you ask them every time you listen to music and for different instruments, you are self-training to listen critically to music until eventually, it occurs on autopilot. You'll listen to all music with a more critical ear and begin to make comparisons between professional music and your own. This *will* produce the single most significant difference to your music production skills.

AUDITORY PRACTICE

Before moving on let's put our listening skills to practice. We'll examine the track 'Lean On' by Major Lazer. I've chosen this track because its proved itself to be popular in both the dance music scene and pop. It's instantly identifiable and an easily accessible source. You can find the record on YouTube, Spotify, or it can be purchased from iTunes. Listen intently to the song a number of times and jot down anything you notice about the production of the song. Then, let's compare notes …

The track is composed in G Minor and with a tempo of 88BPM. The entire track consists of just four chords with a repeating rhythmical pattern (Eb/F/Gm/Bb). The first four bars are the first two chords of the progression (Eb/F) and the next eight bars consist purely of the chorded stab and vocals. Note the use of lengthy reverb and delay on the vocals, and also the pitched down vocal sample occurring in the background throughout the intro.

When the bass is introduced (after the eight bars), it's a triangle or sine wave (I suspect a triangle) played low in the octave that holds the root note of the chord for each bar rather than follow the rhythmical pattern of the leads.

The frequency of the kick ensures it sits above the bass and remains 4/4 for the first verse, pre-chorus, and chorus of the song. However, in the second verse, the kick changes pattern. Also, the kick changes in *tonality* between the first pre-chorus and chorus. In the chorus, the kick becomes more substantial with a stronger transient.

The hi-hats and side-stick, when introduced, display a slight 1/16th swing in the groove, while the finger snap is treated to a reverb that changes in length throughout the song. The vocals also go through a series of different effects. In the first verse, they're treated to room reverb with a small amount of delay but for the words 'Blow a kiss, fire a gun' and 'we all need somebody to lean on,' they're doubled with harmonies, further reverb, and stereo delay pushing the vocals wider in the stereo image.

The real magic behind the track, however, is the amount of vocal manipulation and how the drums change their rhythm throughout. There are a significant amount of chants and shouts occurring throughout, weaving in and out of the vocals in the verse and the chorus acting as instruments.

They're obviously sampled and pushed to extreme pitch ranges to behave in a similar way to instruments. This is particularly evident with the 'hey' in the chorus; it almost acts as an offbeat reggae guitar to complement the dancehall rolling hi-hats that occur there too. All these shouts are treated with different effects, some experience reverse reverb, others are washed in reverb, and all these effects continuously change over the course of the record. The break is also unusually complex, involving multiple vocal chants, all experiencing different timed effects that evolve as the sound progresses.

Of course, this is not a comprehensive breakdown of the track but is, instead, acting as an example of what we need to listen for. While on casual listening, the track appears fundamentally simple, it's actually particularly complex. With music production, the devil is in the detail, but without this detail the track would not be the success that it is. This is why it is important to develop your listening skills. If you can't hear these smaller, finer details, you won't know what your music is missing or how to apply them.

CHAPTER 4

Hearing protection

Color is to the eye what music is to the ear.

Louis Comfort Tiffany

Equally important as training your ears is *looking after them*. Many producers don't give their hearing a second thought, and often monitor at loud levels or attend clubs without hearing protection. These practices will result in permanent hearing damage and even armed with the best audio interface and monitors, if you're hearing is damaged, they cannot act as a replacement. So, in this chapter, we're going to look at the importance of hearing protection and how hearing loss and tinnitus will affect you and your work.

As I'm not an expert in this field, I asked my audiologist to contribute this chapter. Alex is a qualified audiologist who has worked in both the NHS and the private sector; she has a BSc (Hons) in audiology and regularly holds seminars on hearing, tinnitus, and hearing protection.

Our world is full of noise, from natural sounds to electrical appliances to traffic and music. While we are regularly subjected to noise, any that is more than 85 dB SPL is considered liable to cause permanent damage if the exposure length is excessive. When we consider that traffic noise alone can often reach and, indeed, exceed 85 dB SPL and concerts/parties/personal listening devices, and home studios can quickly reach in excess 100 dB SPL, it is essential to consider how much you could be permanently damaging your hearing.

Hearing damage is a combination of volume and the duration of the exposure. There are guidelines on noise exposure and international regulations that state the amount of exposure suitable for an individual. It is reported that, when at work, the loudest noise that a person should be exposed to over an eight-hour day is 85 dB, but if this sound level is reduced by 3 dB, it is possible to double the exposure time.

As an artist and producer, you will train ears your ears so that you can recognize frequency, compression, and many other artifacts in the music. But to

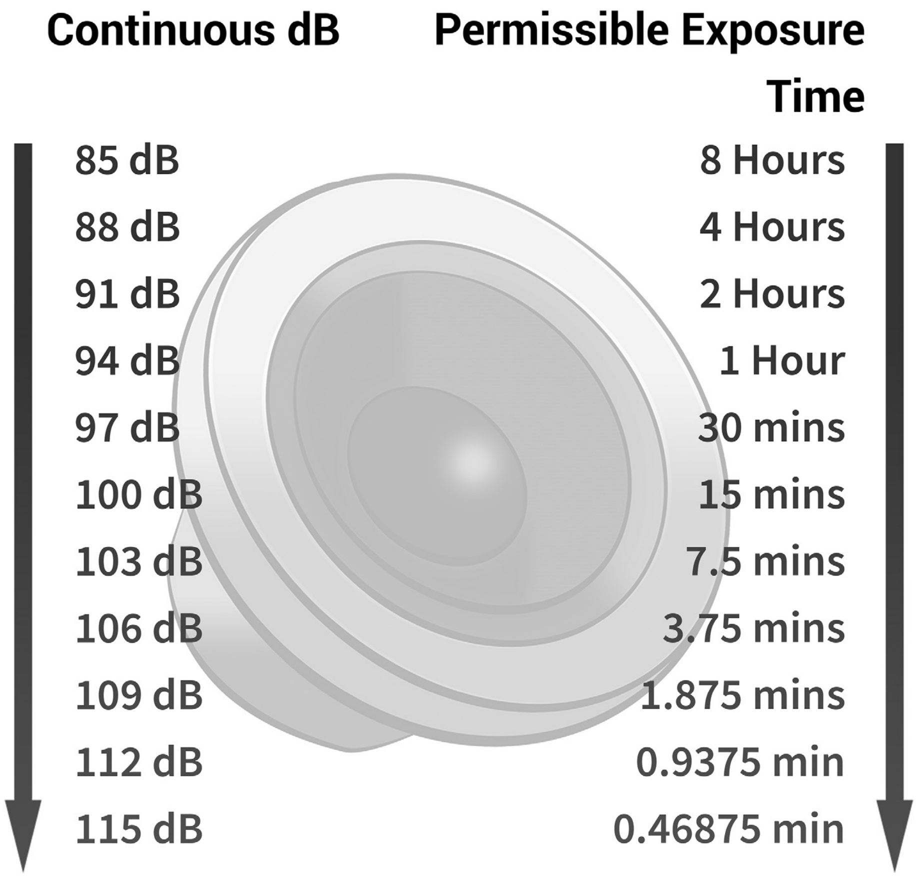

FIGURE 4.0
Levels of sound and length of time for safe listening

accomplish this, it is essential to wear the appropriate hearing protection when subjected to loud noises or music. There are many different styles of hearing protection. It is important to always read the manufactures specification and be aware of the current standards, legislation and recommended exposure times:

- Basic ear plugs/foam ear plugs. These are a one-size-fits-all product, although they are sometimes available in small, medium, or large, and are easy to obtain at minimal cost. They fill the ear with foam and cut out excessive harmful sounds, but at the same time, they also prevent the wearer from being able to detect speech. They are designed to be a used once and then disposed of.
- Generic fit plugs with filters. These are similar to the foam plugs mentioned above, available as a one-size-fits-all, however, these contain a filter. This enables you to maintain the ability to hear both music and speech at normal conversation levels but prevents loud sounds from causing damage. Depending on the brand, there are several different filters available, and it is important to select the appropriate one.

- Custom-filtered hearing protection. These products require an impression or scan of your ears. These impressions must be taken by a professional audiologist and can be expensive, but they are the most comfortable to wear for long periods of time. These have a bore in them to insert a filter, and the filter to use depends on the activity. They are available to cover a wide range of activities such as motorsport, shooting, heavy industry, and music (attending a party, concert ,or playing an instrument). There are also filters that are designed specifically for musicians. These have a flat response attenuation, which performs in the way as an un-occluded ear (not blocked ear). This means that their sound is the same as an open ear but attenuated in gain, thus far less likely to result in hearing damage. These are available often with interchangeable filters with values of, but not limited to, 10 dB, 15 dB, 25 dB, and 30 dB.

Regarding the uses of filters for musicians and engineers:

- 10 dB– is recommended for light noise levels similar to a vocalist or an acoustic guitar.
- 15 dB– is recommended for medium noise levels, and is the most common filter.
- 25 dB– is recommended for heavy noise levels, for example highly amplified instruments, percussion or bass players.

CUSTOM-MADE MONITORS

These perform a dual function: they block out unwanted noise and also act as a single in-ear headphone, so the musician may listen to a stage mix while performing.

The range of products available to protect hearing is varied but not complicated. An audiologist is always worth a visit to first obtain a baseline in hearing thresholds and to discuss what would be the most appropriate product to purchase.

Impressions

When having the molds taken for custom products, it is essential that the audiologist ensures the impression is deep enough. Custom ear protectors must sit deeply enough within the canal of the ear so that there is no or little occlusion experienced. When you speak, you not only hear through air conduction but also through bone conduction. We don’t typically notice this occurrence as the ear canal isn’t occluded, so any excess sound therefore escapes.

We are often unaware of what our voice sounds like because we hear it partly through bone conduction. However, when you are wearing hearing protection, the sound is unable to escape, so you hear your voice. Therefore, the hearing protection needs to extend past the cartilage in the ear canal so that this isn’t experienced.

HEARING LOSS AND TINNITUS

If you regularly attend clubs, parties, or monitor your music at excessive levels (about 90 dB SPL) and do not protect your ears, you will suffer from hearing loss. This affects more than 11 million people in the UK alone, that equates to one person in six who experiences hearing loss, and this figure is growing. Over 40% of people aged over 50 have some form of hearing loss. Tinnitus is experienced by one in ten and is best described as a constant *ringing* in the ears. This figure is not age dependent, and it is possible for tinnitus to come and go over time. Tinnitus is often associated with a hearing loss, but this is not always the case. You may experience tinnitus yet suffer from no hearing loss. Tinnitus is also variable for each individual, thus the sound and the effects experienced may differ. Treatment for tinnitus is in the form of amplification, if a hearing loss is detected, and followed with therapy/education.

As an artist or engineer, you should have your hearing checked/screened every five years once you are over the age of 40 and every two years over the age of 60. A hearing test/screen should be viewed as a health check as the professional you see will also check the condition of the external ear.

When the hearing is tested, the results are recorded on an audiogram displayed in dBHL (decibel hearing level). dB SPL is not used in audiology because it produces a curved, wavy line that is not easy to interpret. Therefore, the curved line of the SPL scale of normal hearing is normalized so that it becomes a straight line, indicating normal hearing as 0 dB HL.

If your results for a hearing test are –10 dBHL to 20 dBHL, you are considered to have normal hearing. Below the 20 dBHL line on the audiogram is considered

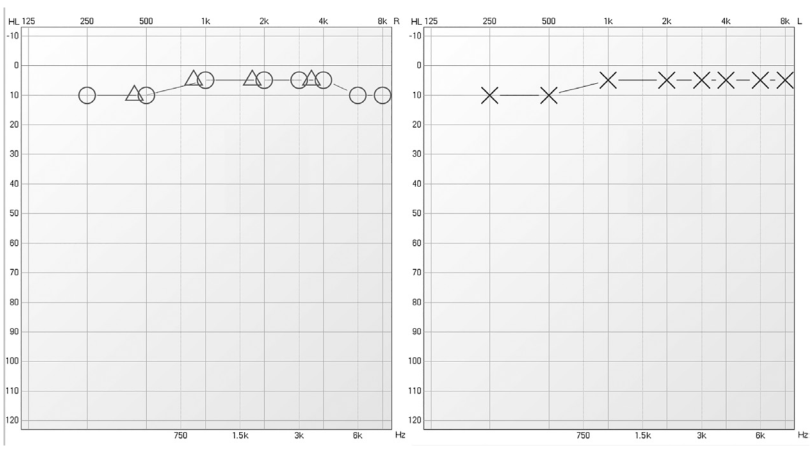

FIGURE 4.1
Normal hearing (right ear = 0, left ear = x)

hearing loss. Audiometry measures **A**ir **C**onduction (AC) hearing ability at 0.25, 0.5, 1, 2, 3, 4, 6, and 8 kHz, and, depending on the results, it may sometimes be necessary to also measure at 0.75 and 1.5 kHz, and **B**one **C**onduction (BC) at 0.5, 1, 2, 3, and 4 kHz. These are measured since audiologists are looking for what type of speech sounds you are unable to hear.

Often the speech banana is also included on the audiogram enabling the level of hearing to be related to the ability to understand speech sounds at a reasonable conversational level. Depending on the clinic you attend, the audiologist may perform many different tests including, but not limited to, tympanometry, speech tests or **A**uditory **B**rainstem **R**esponse (ABR).

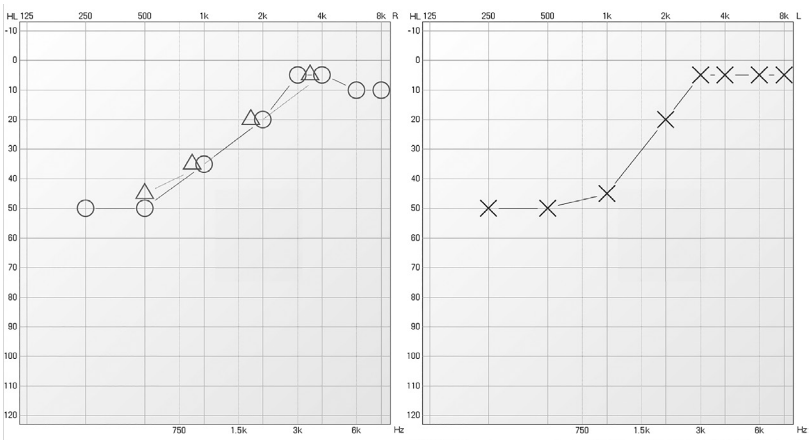

FIGURE 4.2
Low-frequency hearing loss (right ear = 0, left ear = x)

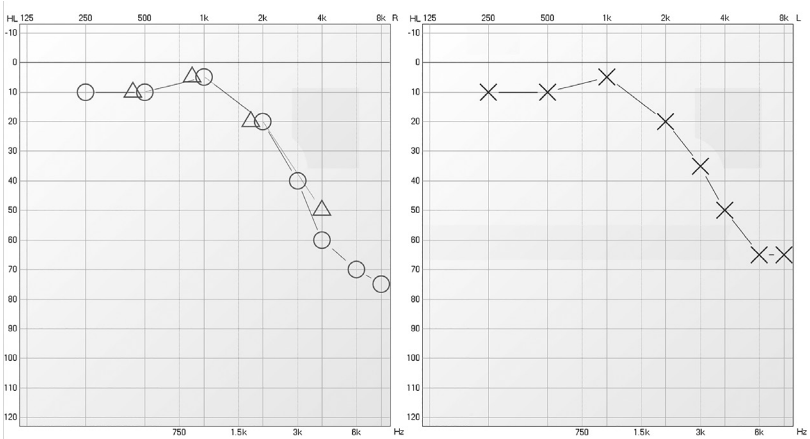

FIGURE 4.3
High-frequency hearing loss (right ear = 0, left ear = x)

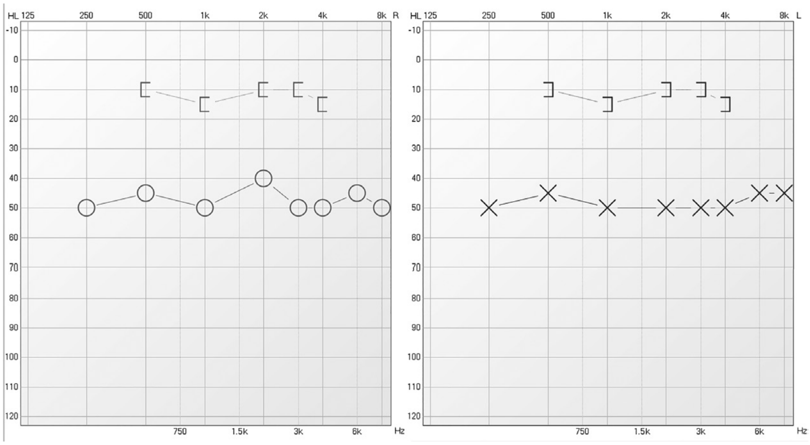

FIGURE 4.4
A conductive hearing loss (right ear = 0, left ear = x)

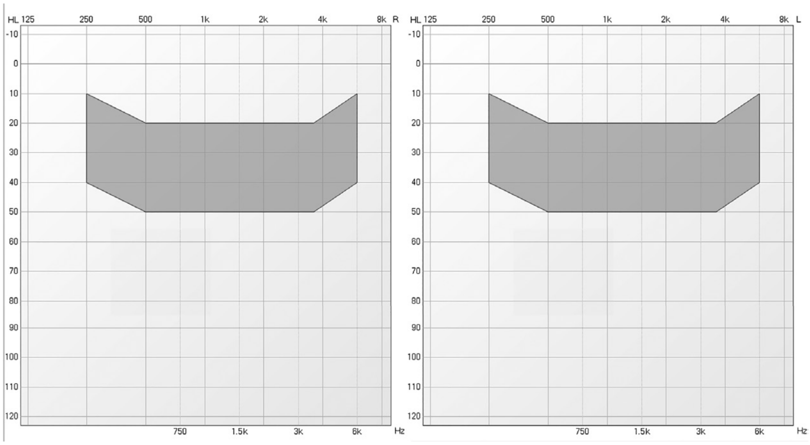

FIGURE 4.5
Speech banana

You probably won't realize that your hearing is deteriorating as the process occurs very slowly over time and our brain compensates with auditory memory. We tend to think that others are not speaking clearly as sounds start to become muffled. As time passes, you might begin to withdraw from social situations as you begin to struggle forevermore to both hear and communicate effectively. It is generally at this stage that you will seek help.

CHAPTER 5

The science of frequency and amplitude

If you want to find the secrets of the universe, think in terms of energy, frequency and vibration.

Nikola Tesla

Before we progress any further into music production, we must understand the fundamentals of acoustic science and engineering. After all, this is *electronic* dance music. It is a style of music that has developed from the use and abuse of electronic audio equipment. Whether this equipment is hardware or virtual renditions within an audio workstation, it relies on an understanding of the science behind it. The first step in this journey is to understand a few basics of acoustic science.

When an object vibrates, surrounding air molecules are forced into vibration and move outwards in all directions. This outwards momentum of air molecules could be compared to the reaction when an object is dropped into a pool of water. The moment the object strikes the water, a series of small waves spread spherically outwards. Each motion of the wave, whether through water or air, follows the same principle and consists of two states: compression and rarefaction.

When a tuning fork is struck, the vibrating tines cause the surrounding air to move through these two states of compression and rarefaction. As the tines move closer to one another, air is compressed between them. When they move outwards, air molecules surrounding the tines are pushed in an outward movement. This continual back and forth motion continues until the tuning fork come to rest.

The numbers of rarefactions and compressions, or *cycles*, completed every second are measured in Hertz (Hz). This is named after German physicist Heinrich Rudolf Hertz who first documented the propagation of sound through various media. If the compression and rarefaction cycle completes 300 cycles per second, it has a frequency of 300 Hz while if there are 3000 cycles per second, it exhibits a frequency of 3000 Hz (or 3 kilohertz (kHz)).

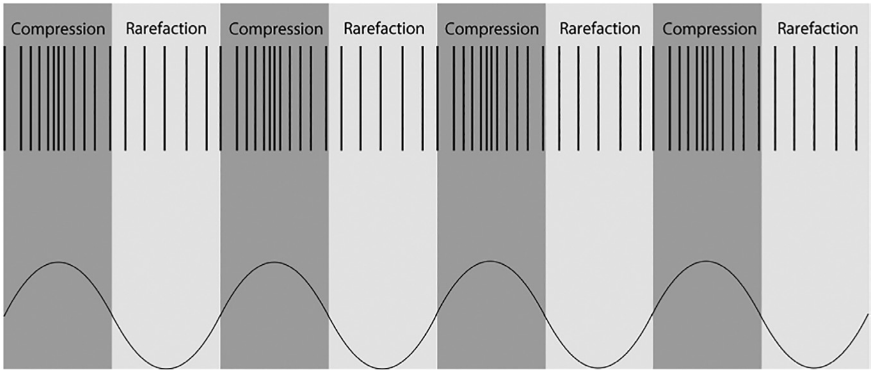

FIGURE 5.0
Compression and rarefaction of air molecules

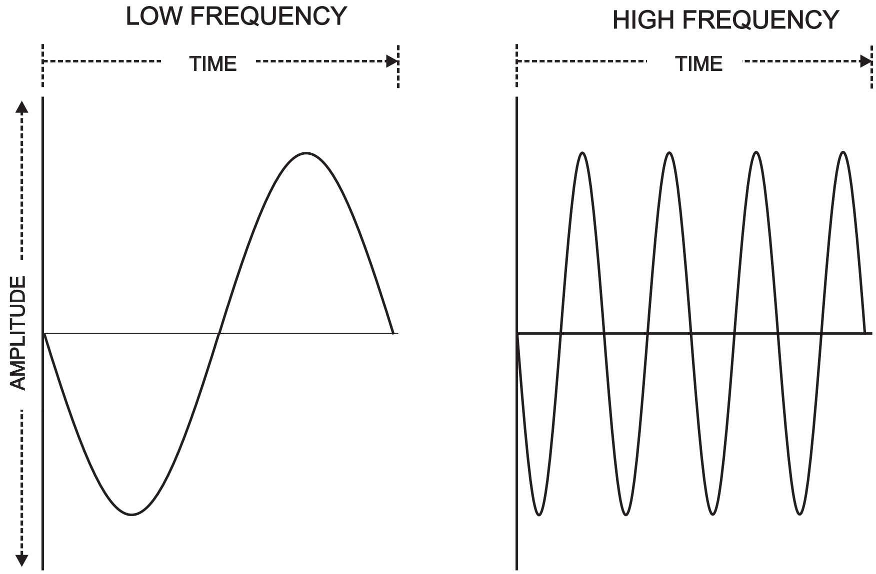

FIGURE 5.1
Low and high frequencies

For music, frequency determines pitch. The more quickly an object *oscillates* (vibrates), the shorter the cycle between compression and rarefaction and the higher the pitch becomes. An example of this action is shown in Figure 5.1.

When an object oscillates, it repeatedly passes through the same positions as it moves back and forth through its cycle. Any one particular point during this cycle is termed the 'phase.' Phase is measured in degrees, similar to the measurement of a geometric circle. As shown in Figure 5.2, each cycle starts at zero, passes through 'zero crossing,' and eventually returns to zero, thus a complete cycle passes through the 'zero crossing' three times.

If we recreate an oscillation in scientific conditions, it would consist of only one fundamental frequency. This is a single frequency and determines the pitch.

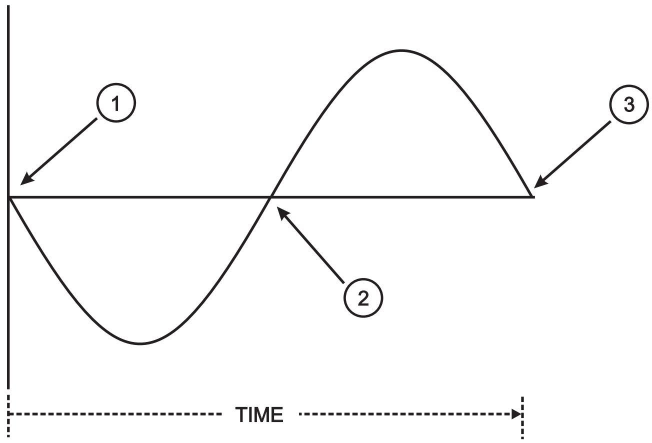

FIGURE 5.2
The zero crossing in a wave form

It is only possible to hear a single wave in controlled conditions because in the natural world many waves at differing frequencies are always occurring and interacting with one another.

If two oscillations occur at different frequencies, both may begin at the same zero point, but the higher frequency would complete more cycles than the lower in the same amount of time. Provided that the oscillations of both continued, they would eventually both reach zero point simultaneously before repeating the process all over again. This constant cycle of drifting in and out of sync is termed *beating* and results in any number of additional tones called *harmonics*. It is these harmonics that contribute to the sounds we hear.

If you throw this book at the wall, besides upsetting the author, it will create a fundamental frequency as it strikes the wall. The air molecules set into motion from this event would force other objects in close vicinity also to resonate and produce fundamental frequencies of their own. As these all combine, many harmonic overtones are generated. However, since these are all random frequencies, it rarely results in a pleasing tone.

Musical instruments are designed to produce harmonics that occur at specific integers of the fundamental to provide a pleasing sound. The Greek philosopher Pythagoras (of the math triangle geometry fame $a^2 + b^2 = c^2$) discovered this harmonic relationship. He noted that the harmonics generated by any musical instrument would always occur at specific intervals of the fundamental frequency. The first frequency would occur an octave above the fundamental, whereas the next would be at 3:1, followed by 4:1, then 5:1, and so forth.

Because the harmonic content or 'timbre' (the French word for color) of a sound can be particularly complicated, conceptualizing the resultant waveforms proves difficult. It took Jean Fourier, a French scientist, in 1822 to theorize that, no matter how sophisticated a sound may be, it could be broken down into a series of frequency components and by using only a given set of harmonics it would be possible to reproduce a waveform. To quote directly:

> Every periodic wave can be seen as the sum of sine waves with certain lengths and amplitudes, the wavelengths of which have harmonic relations. (Jean Fourier, 1827)

The mathematical theorem is complex and beyond the scope of this book, but it is based on the concept that any sound can be determined by the volume relationship of the fundamental frequency to its associated harmonics and their evolution over time. The development is shown in Figures 5.3 and 5.4.

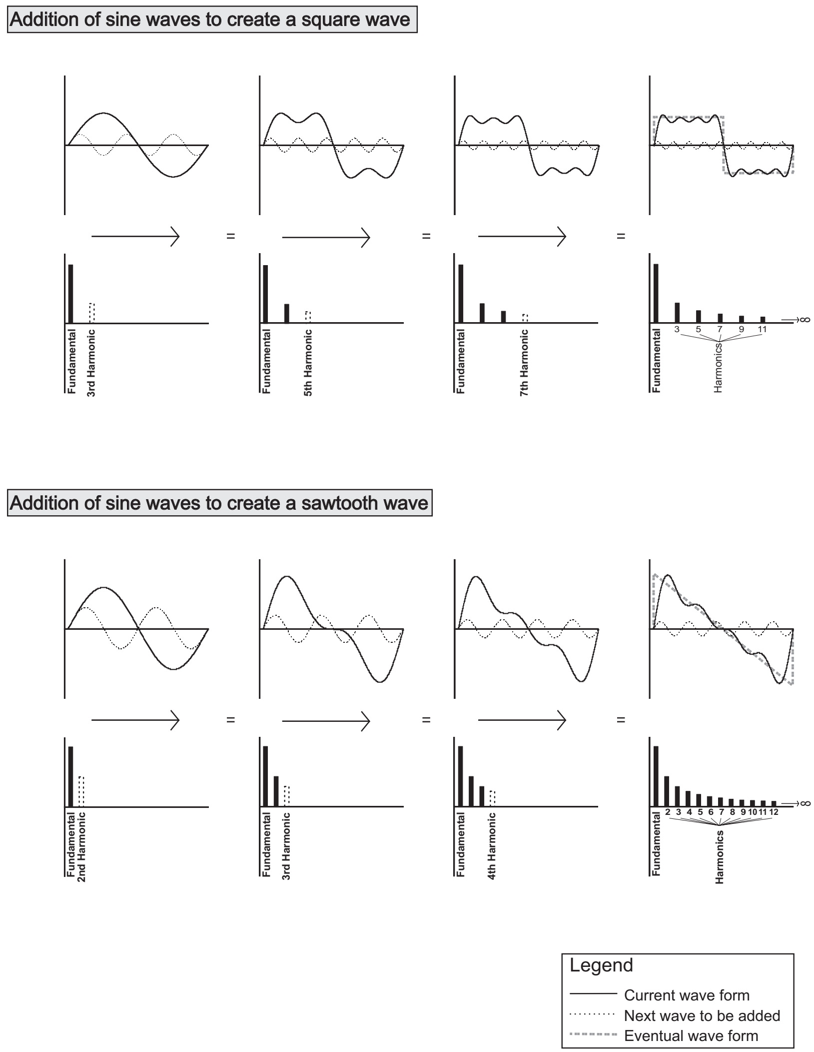

FIGURE 5.3
How multiple sound waves create harmonics (1)

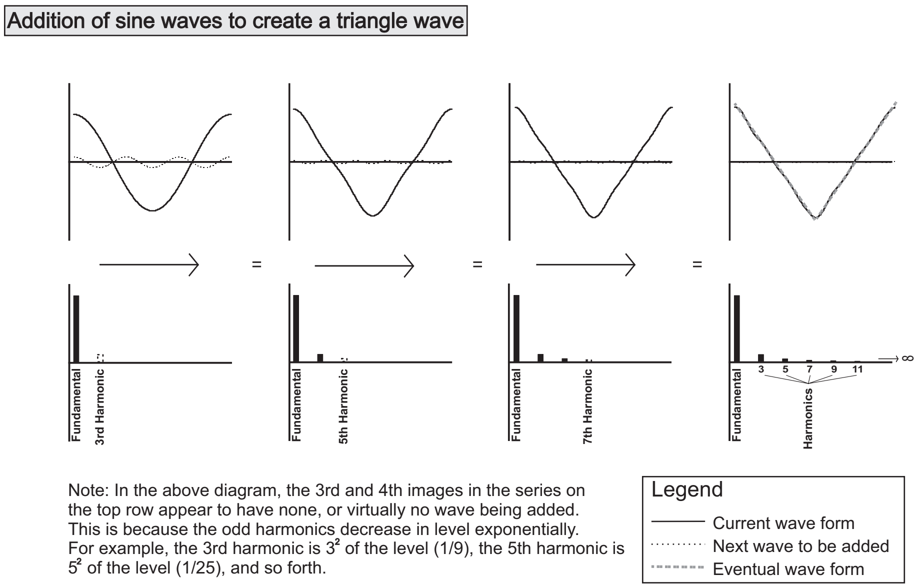

FIGURE 5.4
How multiple sound waves create harmonics (2)

AMPLITUDE AND GAIN

Alongside pitch and timbre, a final character of sound is *amplitude*. Changes in amplitude are a result of the number of air molecules an oscillating object displaces. The more air an object displaces, the greater the amplitude becomes.

Amplitude is measured by the degree of motion of air molecules that correspond to the extent of rarefaction and compression that accompanies the wave. In many musical instruments, the amount of air displaced is too small for the sound to be audible and therefore they employ different forms of amplification. In acoustic instruments, this is accomplished through forced vibration. On a piano, when the wire is struck by the mallet, a large wooden board located beneath is also set into motion. This sounding board is forced into motion via the displaced air molecules, and since this board moves a higher amount of air, the sound is amplified. Similarly, with wind instruments, the body of the instrument is designed to increase the resonance of the air traveling through it.

Amplification is measured by its gain. This is the ratio of output to input and is measured in decibels. This is a unit of measurement first conceived by Bell Laboratories. It was originally used to express how much power was lost when sending voltages over vast lengths of telephone cable. The loss of power was measured in *Bells* but being such a significant measurement when it is applied to audio, it must be subdivided into 10ths of a bell, hence the term *deci*bel.

DECIBELS

Decibels are an appropriate measurement for audio because they accurately represent the non-linearity curve of human hearing and also permit for a sizeable audible range. Understanding decibels requires an understanding of logarithmic scales and in particular working with base 10. An example of this is shown in Figure 5.5.

The lower numbers represent the sound intensity, and each respective number is to the power of 10. For example, 10 to the power of 2 is 100 (10 x 10 = 100) while 10 to the power of 3 is 1000 (100 x 10 = 1000) and 10 to the power of 4 is 10,000 (1000 x 10 = 10,000), and so forth through the scale.

Using Figure 5.5 as a reference, a movement from 10 dB to 20 dB would be equivalent to a 10-fold increase in intensity (to the power of 10). Similarly, 30 dB would be 100 times more intense than 10 dB and 40 dB would be 1000 times more intense. This type of scale offers an accurate representation of how human hearing behaves.

Decibels are not as direct a measurement as many suggest, however. Decibels express the ratio or difference between two *different* levels – a reference level and the current level. This is why any decibel measurement is (or should be) followed by a suffix. The suffix that follows dB will determine the reference level.

If you measured noise levels from your studio, then **s**ound **p**ressure **l**evel (dB SPL) would be the reference level. With this reference, 0 dB is the lowest threshold of human hearing, and anything above this measurement would be considered an approximate representation of its loudness. I say approximate here because loudness, unlike gain, is unquantifiable and relative. It depends on your age, the frequency, and your perspective.

In theory, the human ear can detect frequencies from as low as 20 Hz up to 20 kHz, but this depends on many factors. Indeed, while many are capable of hearing frequencies as low as 20 Hz, the ability to perceive higher frequencies changes with age. Most teenagers are capable of hearing frequencies as high as 20 kHz while the middle-aged may struggle with frequencies above 14 kHz.

Human hearing has developed in such a way that we will perceive some frequencies to be louder than others even though they are at the same gain. This response is part of our survival mechanism and developed for us to single out the human voice among a series of other sounds or situations. Someone shouting

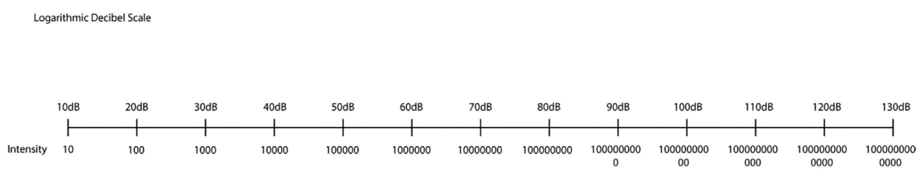

FIGURE 5.5
The logarithmic scale

'there's a hungry looking bear behind you' … to be able to hear that above any other sounds is a definite survival advantage.

The human voice centers at 3–4 kHz so this is where our ears have become most attuned. At conversation level, our ears are most sensitive to sounds occupying the midrange and any frequencies higher or lower than this must be physically louder for us to perceive them to be at the same volume. In fact, at normal conversation levels, it's 64 times more difficult to hear bass frequencies and 18 times more difficult to perceive the high range. Interestingly, this observed relationship changes as the gain is increased.

If the gain increases beyond normal conversation level (approximately 60 dB), the lower and higher frequencies become perceivably louder than the midrange. In 1930, two researchers Fletcher and Munson from Bell Laboratories were the first to experiment and measure this uneven hearing response of the ear. This is known as the Fletcher Munson contour control (see Figure 5.6).

To put this into perspective if you were to measure a 60 Hz sine wave at 30 dB with an SPL meter, despite the measuring equipment stating there is a 30 dB sound present, it would be inaudible. Therefore – from a hearing perspective – this particular measurement would be completely useless. To counter this problem, the **I**nternational **E**lectrotechnical **C**ommission (IEC) introduced weighted filters.

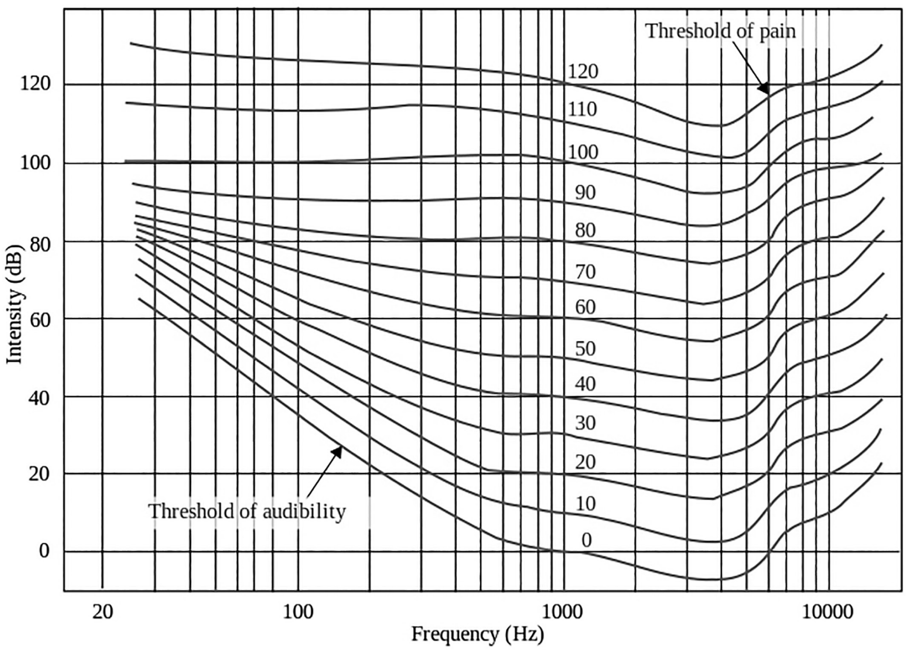

FIGURE 5.6
The Fletcher Munson contour control

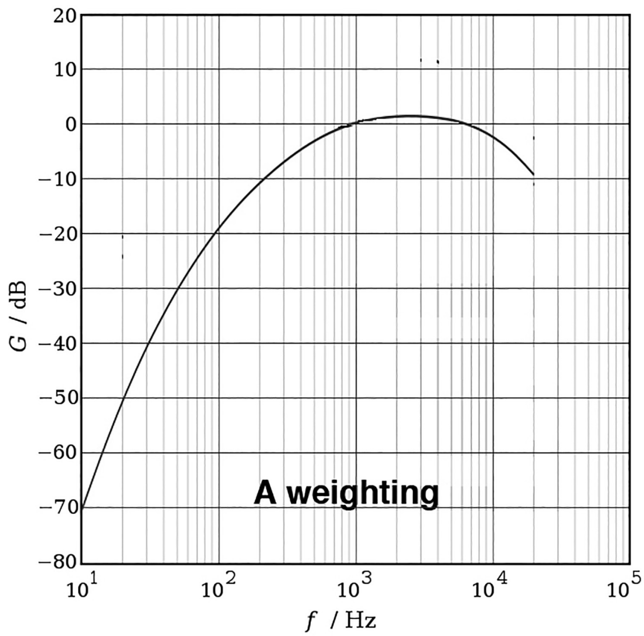

FIGURE 5.7
A-weighted filters

Weighting filters are a number of different filters that reduce or enhance specific frequencies along the sonic spectrum. These (currently) consist of A, B, C, D, K, & Z. In most audio hardware applications, *A-weighting* is the most popular because this is believed to represent human hearing at low levels accurately. With A-weighting, the filter gradually opens allowing more frequencies through until it reaches its apex at approximately 2–3 kHz where it then begins to close again. Figure 5.7 shows this weighting filter's response.

When weighting is used, the filter type is applied to the end of the decibel measurement. For example, many microphone pre-amplifiers and processors may rate their **S**ignal to **N**oise **R**atio at 120 dBA (**SNR 120 dBA**). This is the ratio between the noise created by the device and the level of the desired signal.

The signal to noise ratio is one of the most important properties to factor in when employing any production hardware. It is used by many manufacturers to quantify the amount of noise that will be generated by a hardware device processing a signal. The lower the SNR, the more noise the unit is likely to introduce into a low-level signal.

CHAPTER 6
Synthesizers

Play the music, not the instrument.

Anonymous

Despite having developed significantly over the past 25 or so years due to developments within the DAW, the humble synthesizer remains as the single most influential musical component in the production of electronic dance music.

Although there are various forms of synthesis, the three most favored by dance artists are subtractive analog, wavetable, and frequency modulation. These three all consist of combining some primary sound generators to create a timbre that is rich in harmonics. This timbre is then sculpted through the use of filters, modifiers and further time-based modulation effects to create interesting motions in a sound.

The approach is different depending on the form of synthesis in use, but an understanding of analog subtractive synthesis goes a long way to comprehending all of them so we'll begin this chapter by investigating analog subtractive synthesis.

Any subtractive analog synthesizer consists of four essential components:

- Oscillators. These are combined and mixed to produce a timbre that is rich in harmonics.
- A filter, employed to sculpt the resulting harmonics.
- A time-based amplifier to define the overall level of the sound.
- Modifiers to add time variant modulation/movement to the timbre.

OSCILLATORS

The oscillator section is the heart of *any* synthesizer. These produce the fundamental tones of the instrument. In many early synthesizers, there were only three oscillator waveforms available: square, sawtooth, and triangle waveforms. However, over the years further waveforms have been introduced including sine, noise and pulse waves.

The most basic waveform is the sine wave.

FIGURE 6.0
The DMP Sapphire analog-modelled synthesizer

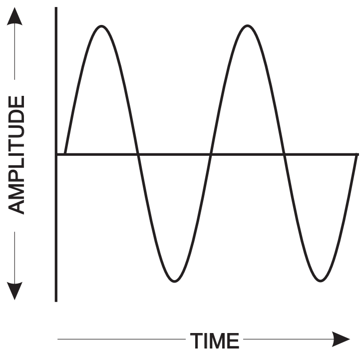

FIGURE 6.1
A sine wave

The sine wave

A sine wave is the mathematical sinusoidal function. A sine wave consists of only a fundamental frequency and does not contain any harmonics; therefore, it is audible as a single fundamental tone, comparable to a whistle. They are not ideally suitable for subtractive synthesis and this is the reason they did not feature in the first analog synthesizers. After all, if the fundamental is removed, there is no sound. As a result, the sine wave is only used on its own to create sub bass, and is more commonly mixed with other waveforms to introduce body or power to a sound.

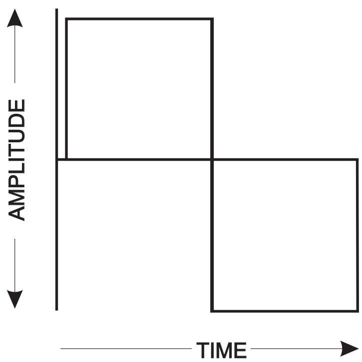

FIGURE 6.2
A square wave

The square wave

A square wave is the most uncomplicated waveform that can be produced by an electrical circuit because it exists in only two states: high and low. This wave provides a number of odd harmonics that are best described as mellow, hollow, or woody. The square wave can make an excellent bass if played low on the keyboard but is often mixed with other oscillators to add width or body to a sound.

The pulse wave

A pulse wave is a square wave, but the width of the high and low states are adjustable thereby varying the harmonic content of the sound. It is unusual to see both square and pulse waves featured on a synthesizer. They will either employ one or the other. The option to adjust the width of the pulse results in many complex sounds from deep, hollow timbres through to thin reedy style sounds. A favorite technique is to use cyclic modulation on the width of the pulse wave, so it moves through its different states over time, creating a complex yet exciting tone.

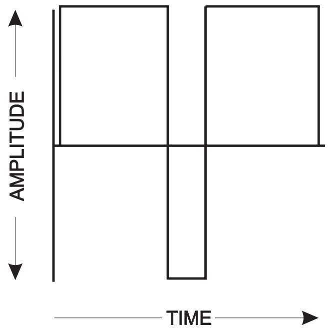

FIGURE 6.3
The pulse wave

The triangle wave

A triangle wave features two linear slopes and, like the square wave, consists only of odd harmonics. As with the square wave, this produces a timbre that best can be described as thin and partially hollow. Typically, this type of waveform is often mixed with a sine, square, or pulse wave to add a sparkling or bright effect to a sound.

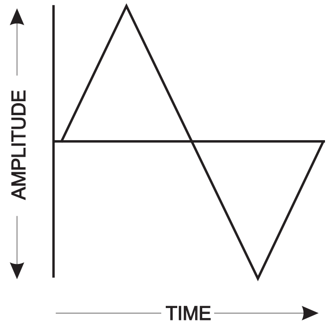

FIGURE 6.4
A triangle wave

The sawtooth wave

The sawtooth wave is the most commonly used oscillator in synthesizers. It produces a proportionate number of even and odd harmonics and results in a timbre with the most substantial harmonic content. This harmonic content is particularly suitable for sweeping and sculpting with filters. Typically, a saw wave is used in the construction of any timbre that requires a full, bright sound.

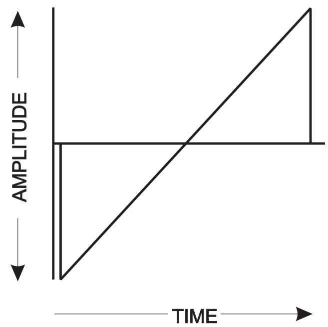

FIGURE 6.5
A saw wave

The noise wave

Noise waveforms are a random mixture of all frequencies rather than actual tones. Noise waveforms are commonly 'pink' or 'white,' depending on the energy of the mixed frequencies they contain.

White noise contains the same amount of energy throughout the frequency range and is comparable to radio static. Pink noise contains differing

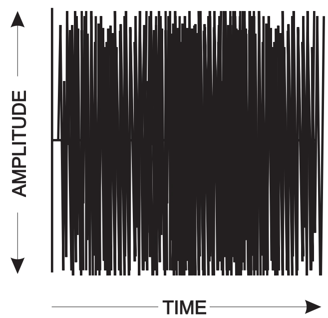

FIGURE 6.6
Noise waveform

amounts of energy at different frequencies, and therefore we perceive it to produce a more substantial, deeper hiss. Indeed, the darker the color of noise, the duller the frequencies it will appear to exhibit.

Noise is most useful for generating percussive sounds and was commonly used in early drum machines to create snares and handclaps. It is also often used in bass and lead timbres to add brightness and air to tones.

FURTHER OSCILLATORS

With advanced computer processing, the number and style of oscillators have increased substantially from the early days of synthesis. Many plug-in synthesizers now offer a variety of waveforms ranging from tri-saw, pulse, variable pulse, and tri-pulse to many differing wave shapes sampled from different sources.

Many of these waveforms are based on the three conventional analog oscillators: the saw, the pulse, and the triangle. They are employed to prevent having to mix various basic waveforms, a task that would reduce the number of available oscillators. For example, a tri-saw waveform is a sample-based oscillator consisting of either three sawtooth oscillators blended at different pitches or a triangle mixed with a saw wave (depending on the manufacturer of the synth). Either of these waveforms produces a sound that is rich in harmonics and saves employing two or more oscillators to create the same tone.

Wave shapes may also fall into the sampled category. These are samples of real-world sounds and instruments rather than oscillations. Some wave shapes may consist of the attack stage of a piano repeated while others may consist of a sampled guitar string or series of noises.

Oscillator's pitch

In traditional hardware analog synthesizers, a voltage determined the oscillator's pitch. Each note on the keyboard would produce a different control voltage that would force the oscillator to vibrate at a particular frequency. For this to work, the control voltages must remain precise to prevent the oscillator's pitch from drifting out of tune. Often the synthesizers were regularly serviced to ensure they remained in tune. For many vintage aficionados and dance musicians, this drifting is part of their natural charm and contributes to the classic *analog sound* that is favored by many producers.

A similar pitch to oscillator function applies today but rather than employ control voltages, each note is awarded a number. From the number transmitted, the synthesizers CPU determines the note depressed and reproduces the correct pitch oscillation.

Regardless of how the oscillations are produced, the foundations for programming analog subtractive synthesizers remains the same. We mix the available oscillators, detune them from one another to add further harmonics, and then we modify or modulate the harmonics with the available modulation parameters on the synthesizer.

FIGURE 6.7
The oscillator sections detune parameter

Detuning of oscillators is accomplished via a *detune* parameter that is commonly featured next to each oscillator. Typically, detuning occurs in octaves, semitones, and cents (100ths of a tone).

When detuning oscillators from one another there is obviously a limit whereby the oscillators will separate from one another and create two distinct individual pitches. This typically occurs if the detuning exceeds 20 Hz but this may be less depending on the manufacturer's choice of detuning algorithms.

Detuning so that both oscillators feature independent pitches can produce interesting tones if the ratio of detuning remains at specific integers. For example, detuning by an octave will result in the oscillator's harmonics increasing in gain due to the fundamental of the detuned oscillator matching the 2:1 harmonic of the first. This has the effect of thickening the sound and can be particularly useful for adding weight to a timbre. Further detuning of 3:1, 4:1, and 5:1 can also produce harmonically pleasing results.

Synthesis tip

Detuning in odd integers rather than even between oscillators will produce more immediate results. Detuning by even amounts will introduce harmonic content that often mirrors the harmonics already provided in the first oscillator. The result is the harmonics that are already present are amplified rather than the process introducing new harmonics. This is not to suggest detuning by even amounts should not occur, however, since even tuning can aid in solidifying the current timbre rather than making it appear busier.

Further control may be offered for syncing oscillators together. This permits two oscillators cycles to be synced to one another. When employed, many synthesizers will synchronize the secondary waveform to the first oscillator's cycle. This means that no matter where in the period the second oscillator currently is when the first begins its cycle again, the second oscillator is forced to restart its cycle too.

Here, if two square oscillators are detuned from one another by –7 semitones, every time the first oscillator restarts its cycle so, too, will the second, regardless of the position in its period. This produces a timbre with a continually changing harmonic structure and is a particularly robust programming strategy. Moreover, if the second oscillators pitch is modulated via an external source or the synthesizer's own internal engine, then as the pitch changes the timbre changes dramatically creating a strong distorted style lead.

Synthesis tip

In the early years of dance music, a favorite approach was to employ a filter envelope to modulate the pitch of the first oscillator as well as the filter. By doing so, as the filter changed with the envelope motion, so too did the sync. This produced a timbre typical of classic early Rave. This classic timbre is making a comeback in genres such as Tech House and Techno.

Once the signal leaves the oscillators, it is routed into a mixer section. Here, the output gain of each oscillator can be adjusted. Using the mixer, you can change the volume of each oscillator and combine them to produce a cohesive whole.

Alongside volume control, some synthesizers may feature additional parameters such as ring modulation. Using this, the frequencies of two oscillators (commonly oscillator one and oscillator two in a multi-oscillator synth) are input into the modulator where the sum and difference of the two frequencies are output.

For example, if one oscillator produces a signal frequency of 440 Hz (A4 on a keyboard) and the second produces a frequency of 660 Hz (E5 on a keyboard) the frequency of the first oscillator is subtracted from the second:

$$660\ \text{Hz} - 440\ \text{Hz} = 220\ \text{Hz (A3)}$$

Then the first oscillator's frequency is added to that of the second:

$$660\ \text{Hz} + 440\ \text{Hz} = 1100\ \text{Hz (C\#6)}$$

Based on this example, the difference of 220 Hz provides the fundamental frequency while the sum of the two signals, 1100 Hz, results in a 5th harmonic overtone. Both of these frequencies are mixed together and then output from the ring modulator.

While this may appear to produce simple tones it does become infinitely more interesting with more complex oscillations such as a saw wave. Here, every harmonic generated from the two oscillators would be subject to the ring modulation resulting in complex timbres. The resulting tones from ring modulation are often described as metallic or bell like, which is a reasonably accurate description.

The option to add noise may feature in the oscillator's mix section. Noise is often used to introduce additional harmonics and brightness to a timbre. Adding noise to multiple saw wave oscillators is a proven technique for trance artists who want to achieve robust, cutting lead lines.

FILTERS

After the mix section, the signal enters the filter stage of the synthesizer. If the oscillator's signal is a piece of wood that is yet to be carved, the filters are the hammer and chisels to shape it. Filters chip away parts of the original signal until a rough image of the required sound remains.

This action makes the filters an essential element of any subtractive analog synthesizer. If the filters are inferior, then few sound sculpting options will be available, and it will be difficult to create quality timbres. Generally, you can determine the quality of a filter by reducing the cut-off parameter from fully open to fully closed. The sound the filter imparts as it sweeps through the timbre defines its character. On poor filters, it will exhibit a stepping quality and sound mainly uneventful. On a quality filter, the sound will exhibit an extra motion or fluidity as the filter sweeps the available harmonics.

The most common filter in basic subtractive synthesis is the low-pass filter. This removes frequencies above a defined cut-off point. The effect of any filter is progressive, meaning that more frequencies are removed the further the parameter is increased.

It is termed a low-pass filter because the high frequencies are removed first, permitting the low frequencies to pass through the filter. As the low-pass filter is reduced, an increasing number of higher harmonics are extracted from the signal until eventually, only the fundamental frequency remains. If the cut-off parameter is increased further, the fundamental is removed, resulting in silence.

Low-pass filters are the most common filter because they prove to be the most useful. Since the fundamental frequency determines the pitch of the note it is not something you want to remove first.

It should be noted that when employing any filter, harmonics that lie above the cut-off point are not attenuated in a 90° fashion. Instead, the upper harmonics are attenuated gradually. The degree of this gradient attenuation depends on the transition band of the filter. This is shown in Figure 6.8.

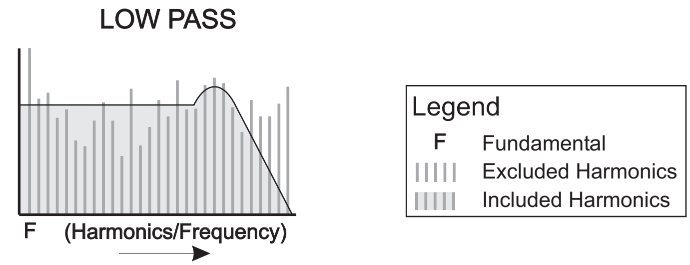

FIGURE 6.8
The action of a low-pass filter

The transition band defines the sound of any one particular filter. If the transition is steep, the filter is said to be 'sharp,' whereas if the transition is gradual, the filter is said to be 'soft.' This is why two different filters performing the same function can sound very different. One may create a beautiful tonal sweep, while the other is entirely uneventful. The reason filters create such an attenuation slope is due to the limitations of electronic resistor-capacitor circuitry and beyond the purpose of this book.

Filter transitions are measured in decibels, and the smallest transition is 6 dB. This results in 6 dB of attenuation per octave of frequencies. It may also be known as a single pole filter because when the action is plotted on a graph, it looks like a tent draped over a single pole.

Single pole filters are the mainstay of EQ because of the short transition period, and although synthesizers may feature them, it is more common to employ 12 or 24 dB filters. These are sometimes called 2-pole and 4-pole filters respectively. 2-pole filters attenuate 12 dB per octave, and 4-pole filters provide 24 dB per octave attenuation.

Because 4-pole filters attenuate 24 dB per octave, they create substantial changes to a sound and tend to sound more artificial than a 2-pole filter. Therefore, it's important to decide what transition period is chosen. If a 24 dB filter is used to sweep a harmonically rich sustained sound, it will result in substantial attenuation throughout, while a 12 dB may create a more natural fluidic sound.

If there is more than one filter available, you may be able to combine them in series or parallel. By doing so, two 12 dB filters can be summed together to produce a 24 dB transition. Or one 24 dB filter could be used in isolation for aggressive tonal adjustments with the following 12 dB filter used to perform real-time gentle sweeping movements.

Although the low pass was the only filter to feature on the very early synthesizers, there are numerous variations of filters on modern synthesizers such as high pass, band pass, notch, and comb. These employ the same transition periods as the low-pass filter, but each has a wildly differing effect on the sound.

A high-pass filter has an effect opposite to that of a low-pass filter. It begins by first removing the low frequencies, gradually moving toward the highest.

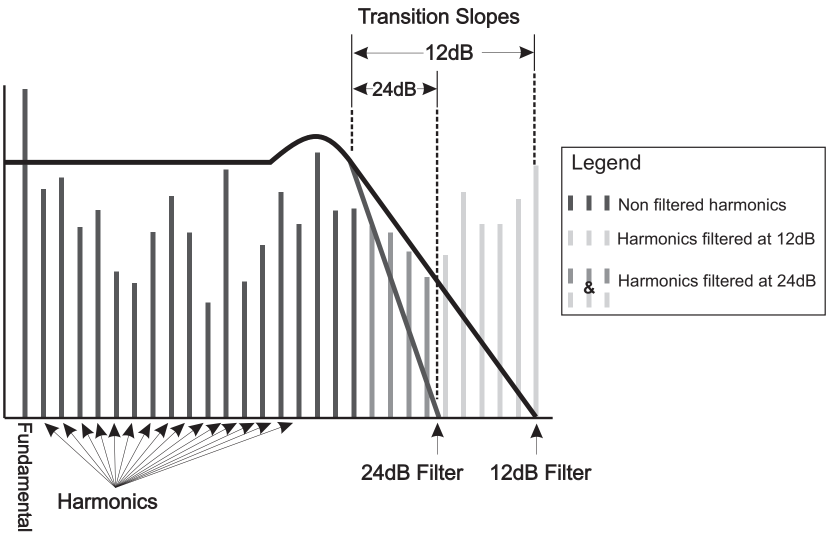

FIGURE 6.9
The difference between 12 dB and 24 dB slopes

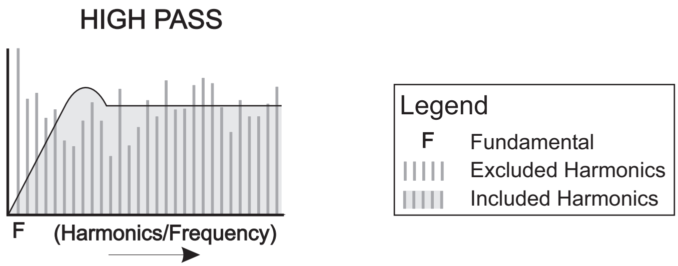

FIGURE 6.10
The action of a high-pass filter

This is less useful than the low-pass filter because it eliminates the fundamental frequency of the sound, leaving only the harmonic overtones. Because of this, high-pass filters are rarely used in the creation of instruments and are predominantly used to create effervescent sound effects or bright timbres that can be laid over the top of another to increase the harmonic content.

Synthesis tip

A favorite technique in dance music is to layer some synthesized tones from different instruments on top of one another to create broad, exciting sounds. When doing so, it is good practice to use a low-pass filter on one synthesizer to maintain the fundamental but use high-pass filters on all other layers. This prevents the timbre from becoming too muddy as a consequence of stacking together many fundamental frequencies.

In both remixing and dance music, it is commonplace to run a high-pass filter over a mix to eliminate the lower frequencies, creating an effect similar to a transistor radio or telephone. By reducing the cut-off control, gradually or immediately, the track morphs from a thin sound to a more significant, heavier sound. This produces a dramatic effect in the right context.

If both high- and low-pass filters are connected in series, it is possible to create a bandpass or band-select filter. These permit a set of frequencies to pass unaltered through the filter while the frequencies either side of the two filters are attenuated. The frequencies that pass through unaltered are known as the 'bandwidth' or the 'bandpass' of the filter. If the low pass is set to attenuate a range of frequencies that are above the current high-pass setting, no frequencies will pass through, and no sound at all is produced.

Bandpass filters, like high-pass filters, create timbres consisting of mid- to upper harmonics. They can be used to determine the frequency content of a waveform because sweeping through the frequencies each harmonic can often be heard. Since this type of filter frequently removes the fundamental, it is commonly used as the basis of sound effects or to create thin sounds that will form the basis of sound effects.

While bandpass filters are used to thin a sound, they should not be confused with band-reject filters, which can be used for a similar purpose. Band-reject filters, often referred to as *notch filters*, attenuate a selected range of frequencies effectively creating a notch in the sound – hence the name – and usually leave the fundamental unaffected. This type of filter is handy for scooping out frequencies, thinning out a sound while leaving the fundamental intact. This makes

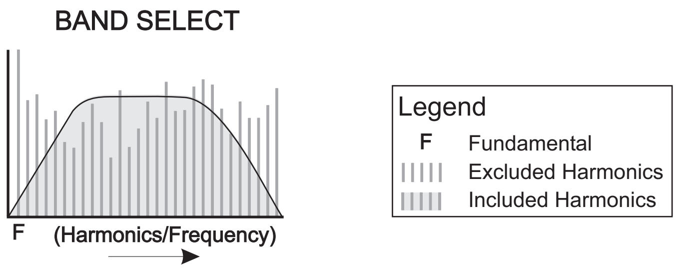

FIGURE 6.11
The action of a bandpass filter

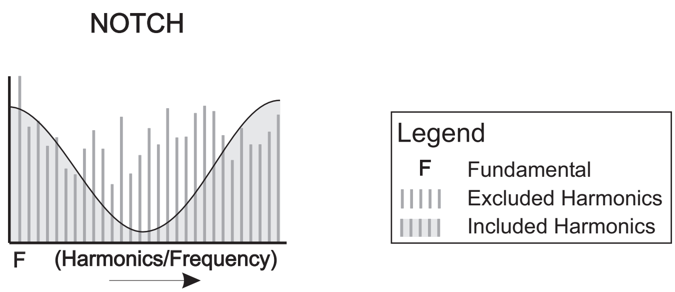

FIGURE 6.12
The action of a notch filter

them useful for creating timbres that contain a discernable pitch but do not have a high level of harmonic content.

Another filter is the *comb filter*. With these, some of the samples entering the filter are delayed in time while the output is fed back into the input of the filter to be reprocessed. With this approach, sounds are tuned to amplify or reduce specific harmonics based on the length of the delay and the sample rate. This makes it useful for creating complex sounding timbres that cannot be accomplished any other way.

For example, if a 1 kHz signal is run through the filter with a 1-millisecond delay, the signal will result in phase because 1 millisecond is coincident with the inputted signal, equaling one. However, if a 500 Hz signal with a 1-millisecond delay is employed instead, it would be half the period length and so would be shifted out of phase by 180°, resulting in a zero. It is this constructive and deconstructive period that creates the continual bump and dip in harmonics, resulting in a comb-like appearance when represented graphically, as we see in Figure 6.13.

This method applies to all frequencies, with integer multiples of 1 kHz producing ones and odd multiples of 500 Hz (1.5 kHz, 2.5 kHz, 3.5 kHz, etc.) producing zeros. The effect of using this filter is best described as highly resonant, and its use is limited to extreme sound design rather than the more basic sound sculpting.

One final element of sound manipulation in a synthesizers filter section is the resonance control. Often called a *peak*, or more simply *Q*, it controls the amount of feedback into the filter. By feeding the output of a filter back into its input, it emphasizes frequencies at the cut-off frequency. This creates a peak at the filters cut-off point.

Resonance directly affects the filter's transition period and creates a brighter, almost ringing tone at the cut-off frequency. Increasing the resonance produces a more dramatic brighter sound and is particularly useful when used in conjunction with low-pass filter sweeps.

On many analog and DSP analog-modeled synthesizers, if the resonance is increased high enough, it will feedback on itself. As more and more of the signal feeds back, the signal is exaggerated until the filter breaks into *self-oscillation*.

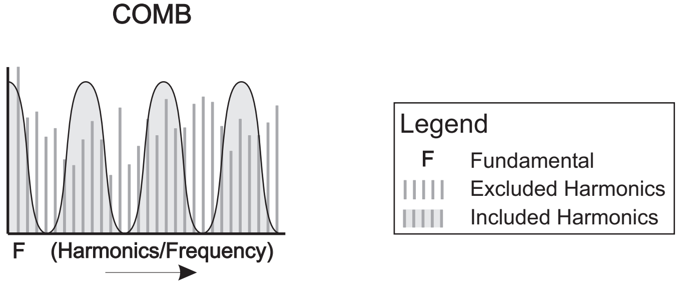

FIGURE 6.13
The action of the comb filter

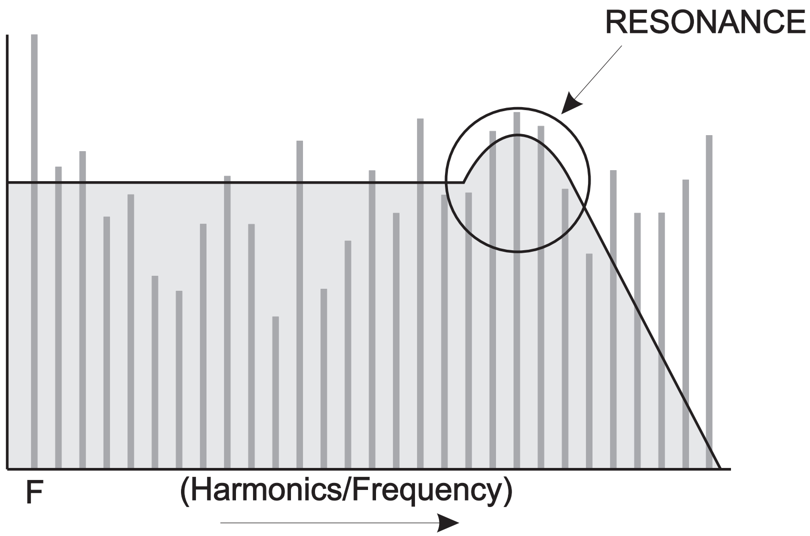

FIGURE 6.14
The effect of resonance

Self-oscillation produces a sine wave at a frequency equal to that of the filters cut-off frequency. By adjusting the cut-off, it is possible to tune the sine wave to any frequency. In the early days of synthesis, this provided a great way to create a sine wave if the synthesizer didn't feature one. In the past, producers would tune the cut-off to produce sub bass style sine waves.

Today, self-oscillation isn't that useful but is still expressed as if it were relevant to a synth. Every synthesizer today features a sine wave oscillator and so self-oscillation serves no real purpose. Plus, digital filters do not self-oscillate at all; they're merely replaced with a sine wave.

A keyboard's pitch can also control the action of a filter by employing *pitch tracking*, *keyboard scaling*, or *key follow*. On many synthesizers, you can adjust the depth of this parameter so you can determine how much or how little the filter should follow the pitch.

When this parameter is set to its neutral state (neither negative nor positive) and a note is played on the keyboard each note is subjected to the same level of filtering. If this is employed on a low-pass filter, the filter setting remains fixed so as progressively higher notes are played, fewer and fewer harmonics will be present in the sound. This results in the higher notes appearing mellower than the lower notes.

If the *key follow* parameter is set to positive, higher notes are awarded a higher cut-off frequency. So the further up the keyboard you play, the more the filter opens, resulting in the timbre remaining bright. Alternatively, if the key follow parameter is set to negative, higher notes will lower the cut-off frequency. So the further up the keyboard you now play, the more the filter will close resulting in higher notes sounding progressively duller.

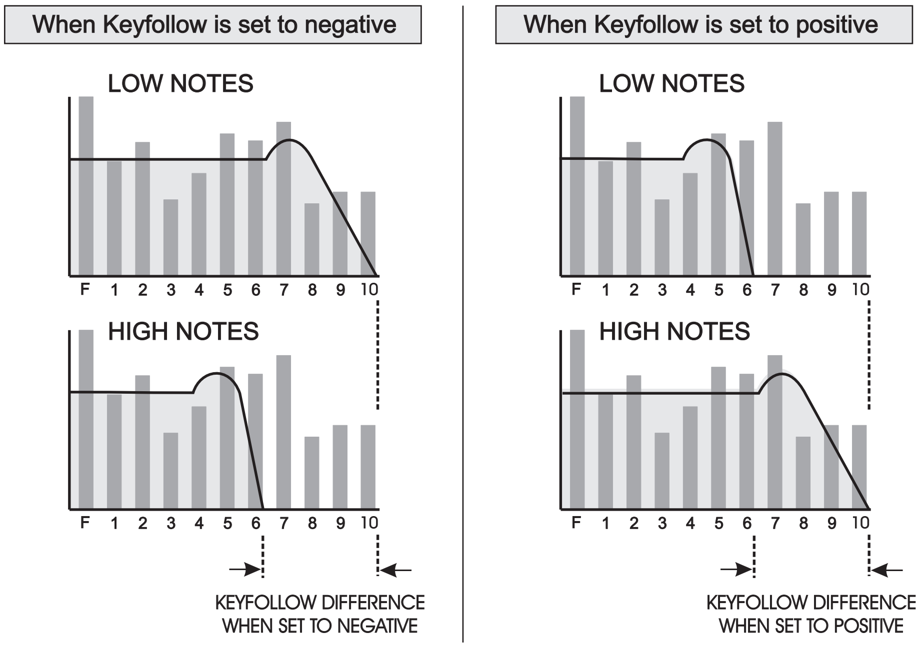

FIGURE 6.15
The effect of filter key follow

Key follow is useful for recreating real instruments such as brass, where the higher notes are often mellower than the lower notes but can be equally valuable on intricate bass lines that extend over an octave, preventing them from clashing with leads or vocals.

Some filters may feature a saturation parameter to act as a filter overdrive. By overdriving a filter, it introduces distortion that can add character and thicken timbres. The process of adding more harmonics to the signal will create rich sounding leads or basses.

THE AMPLIFIER

Few, if any, instruments start and stop immediately. It takes a finite amount of time for the sound to reach its full amplitude and then decay to silence; thus the **E**nvelope **G**enerator (EG) – a feature of all synthesizers – can be used to shape the volume of a timbre over time.

An amplifier envelope determines whether a sound starts instantly or builds up gradually and can also be used to control how the sound dies away (quickly or slowly) when the key is released. These usually comprise of four sections: **A**ttack, **D**ecay, **S**ustain, and **R**elease (ADSR). Each determines the shaping that occurs at specific points on the duration of a note. An example of this is shown in Figure 6.16.

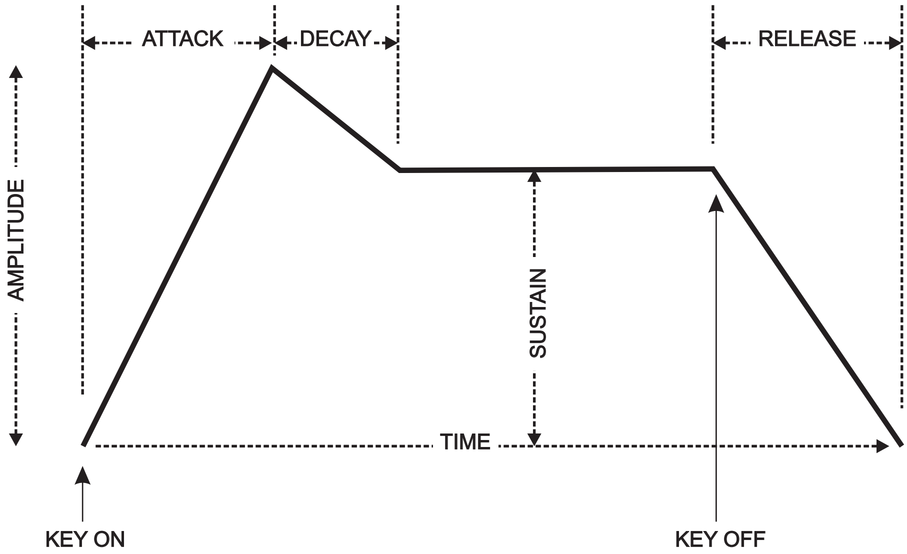

FIGURE 6.16
The ADSR envelope

Attack

The attack parameter determines how the note starts from the point when the key is depressed, and the period it takes for the sound to go from silence to full volume. If the period is extended, the sound will gradually 'fade in,' slowly increasing the gain. If the period is short, the sound will start the instant a key is depressed. Many instruments utilize a very short attack time as more extended attacks can make an instrument appear distant.

Decay

After a note has begun, it may decay in volume. For example, a piano note starts with a very loud, percussive transient, but then drops quickly to a lower volume and sustains while the note is held. The time the note takes to fade from the initial peak at the attack stage to the sustain level is the *decay period*.

Sustain

The sustain determines the volume of the note while a note is held. If the sustain level is set to maximum, then the decay period will be ineffective. This is because during the attack stage the volume rises to its maximum, so there is no level to decay down too. Conversely, if the sustain level is set to zero, the sound peaks at the attack period and will then fade to nothing even if you continue to hold down the key. In this instance, the decay time determines how quickly the sound decays down to silence.

Release

The release period is the time it takes for the sound to fade from the sustain level to silence after the key has been released. If this is set to zero, the sound will stop the instant the key is released. If set to a high value, the note will continue to sound, fading away, after the key has been released.

Although ADSR envelopes are the most common, there are subtle variations, such as **A**ttack–**R**elease (AR), **T**ime–**A**ttack–**D**elay–**S**ustain–**R**elease (TADSR), and **A**ttack–**D**elay–**S**ustain–**T**ime–**R**elease (ADSTR). As there is no decay or sustain elements in drum timbres, AR envelopes are the norm. They may also feature on more economical synthesizers because AR parameters are considered to have the most significant effect on a sound and therefore considered a requirement.

TADSR and ADSTR envelopes are implemented on more complex synthesizers. With the additional period, T (time), in TADSR, for instance, it is possible to set the amount of time that passes before the attack stage is reached.

Not all envelopes offer linear transitions. The attack, decay, and release stages will not necessarily consist entirely of a linear action as shown previously. On some synthesizers, these stages may be logarithmic or exponential, and others may allow you to control whether each envelope stage is linear, logarithmic, or exponential. The differences between the linear and exponential are shown in Figure 6.18.

Alongside an envelope to control the amplitude of the final sound, many synthesizers will feature an envelope that is hard wired to the filter. By doing so, it is possible to employ tonal changes to the note.

This movement in sound is fundamental to producing exciting tones that can maintain repeated listening. A sound that does not change or develop in textural quality doesn't occur in the real world. Therefore, if a sound doesn't develop texturally, listeners will switch off.

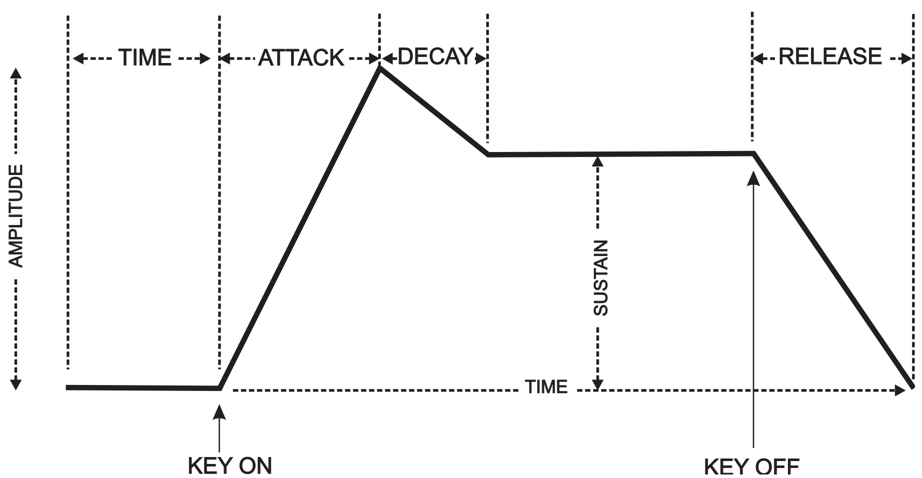

FIGURE 6.17
The TADSR envelope

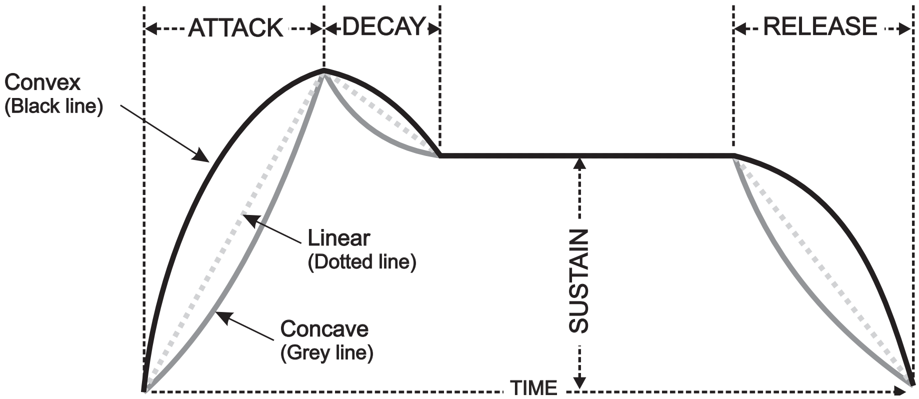

FIGURE 6.18
Linear and exponential envelopes

MODIFIERS

Because of all the above, textural modulation is necessary and all synthesizers offer methods for manipulating sound in the form of modulation sources and destinations. Often termed as a *modulation matrix*, these make it possible to employ one parameter to modify or modulate another independent parameter.

The modifiers available, and the destinations they can modulate, depend on the synthesizer. Many synthesizers feature additional envelope generators alongside the filter and amplitude envelope so you can control further parameters. With these, you could employ an envelope to change the pitch over time or perhaps the pulse width. Alternatively, the synthesizer may only feature a filter and amplitude envelope but allow these to additionally modulate other destinations, too.

In addition to modulation envelopes, there are commonly further oscillators whose only purpose is to modulate other parameters. Whereas a synthesizer's oscillator produces an audible frequency (within the 20 Hz to 20 kHz range), modulation oscillators provide a relatively low frequency that is inaudible to the human ear (in the range 1 Hz to 10 Hz). These **L**ow-**F**requency **O**scillators (LFO) can modulate other parameters to introduce additional movement into a sound.

For instance, you can use an LFO to modulate the pitch of a conventional oscillator. By doing so, the pitch of the oscillator will rise and fall according to the speed and shape of the LFO waveform, and an effect similar to vibrato is generated. If a sine wave is employed for the LFO, then it will result in an effect similar to a police siren.

Alternatively, if this same LFO is used to modulate the filter cut-off, then the filter will open and close at a speed determined by the LFO. And if it were used to modulate an oscillator's volume, it would rise and fall in volume recreating a tremolo effect.

The LFO features depend on the synthesizer but most commonly employ sine, saw, triangle, square, and sample/hold waveforms. The last type of waveform consists of a randomly generated noise waveform that momentarily freezes every few samples before beginning again.

Synthesis tip

A classic effect is to employ an LFO sample and hold waveform to modulate a sine wave oscillator. With a slow LFO speed setting, the result is a series of beeps and pings that were a staple sound of the computers in early sci-fi films.

An LFO must also offer a parameter to determine how much it will augment the destination (usually termed *LFO amount* on a synthesizer), a rate control to manage the speed of the LFO, and some may also feature a *fade-in parameter*.

The fade-in parameter determines how quickly the LFO affects the waveform after a note has been depressed. By employing a long fade in, the LFO will gradually increase its modulation on the source. An example of this effect is shown in Figure 6.19. This style of effect is used to emulate wind instruments because vibrato commonly appears after the initial onset of a tone. With electronic dance music, it is widely used on long strings with the LFO modulating either pitch or filter to introduce fluctuations as it continues.

The LFOs on more capable synthesizers may have access to their own envelopes. This lets you set the LFO's performance over a specified period permitting it to fade in after a key has been pressed and also employ decay, sustain, and release. This is rare, however, and typically most will only employ fade-in and fade-out parameters (an AR envelope).

The destinations an LFO can modulate depend on the synthesizer. Some may just permit LFOs to modulate the oscillator's pitch and the filter. Others may offer multiple destinations and more LFOs. Obviously, the more LFOs and destinations that are available, the more creative options you will have at your disposal.

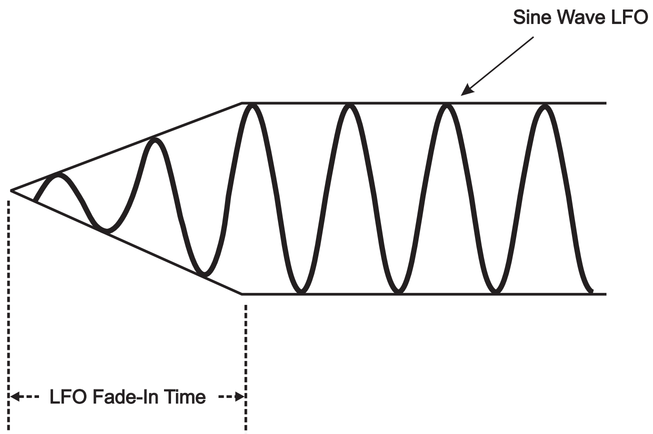

FIGURE 6.19
LFO fade-in

CHAPTER 7

FM and wavetable synthesis

At least it has a ton of presets.

Yamaha salesman

While subtractive analog remains the most popular form of synthesis for many dance musicians, both wavetable and frequency modulation synthesis are increasing in popularity. Much of this status can be traced back to Native Instruments and the introduction of Massive (a wavetable synthesizer) and FM8 (a frequency modulation synthesizer). In fact, genres such as *Dubstep*, *Future House*, and *Mainroom House* rely heavily on the timbres produced via FM and wavetable synthesis, forming part of the distinctive sound.

Frequency modulation was invented in the early 1970s by Dr. John Chowning of Stanford University. It was later developed into a synthesizer by Yamaha, leading to the release of the legendary DX21, DX7, and TX81Z synthesizers. These were a prevalent source of sounds (notably the *solid bass* preset) for many dance musicians for over 20 years.

Unlike analog, FM synthesis produces sounds through operators rather than oscillators. An operator is a sine wave that features its own amplitude envelope. Sounds are generated by routing the output of one operator onto the pitch input of another. By doing so, the pitch of the second operator is modulated by the first. This is similar to the operation of an LFO discussed in the previous chapter but here, operators function at much faster frequencies. This permits the synthesizer to produce more complex tones.

The operator that is used to modulate is called a *modulator* and the operator that is modulated is called the *carrier*. Any operator can function as a modulator or a carrier, or both. In early Yamaha FM synthesizers, there were a total of six operators and the modulator/carriers were preconfigured into a series of set algorithms. These contained preset combinations of modulator and carrier routings and were included in an attempt to make programming an FM synthesizer easy. It didn't. A typical algorithm from the DX7 is shown in Figure 7.0.

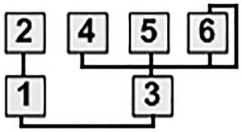

FIGURE 7.0
FM algorithms

The DX7 featured 32 algorithms, but the approach to programming remained the same. You would call up one of the algorithms and then modify the pitch frequency, modulation amounts, and envelope data of the operators to create new sounds. The problem with this approach, though, was the somewhat random nature of the results.

With only a series of fixed algorithms consisting of operators employing pitch modulation on one another, it was difficult to predict the results with any degree of certainty. Moreover, due to the digital nature of the synthesizer, the hardware fascia panel consisted of very few real-time controllers. Instead, numerous buttons adorned the front panel forcing you to navigate and adjust any parameters through a small two-line LCD. This style of interface is like trying to paint a hallway through the letterbox.

This resulted in the DX7 being infamously challenging to program, and so for many artists, it became little more than a preset machine. But these presets would find their way into countless records throughout the 1990s' dance scene.

The introduction of virtual instruments changed this approach due to the significant screen estate available. However, because of the DX7s notorious history with programming, few manufacturers took to producing FM specific synthesizers. The most famous example is Native Instruments' FM8, a powerful FM synthesizer that can accept system exclusive messages from the original DX synthesizers but also features a sophisticated engine allowing it to produce a more extensive range of timbres.

With the FM8, there are no fixed algorithms, and the artist has the freedom to chain the operators together in whatever form they choose. Here, you can use a modulator to modulate a carrier, which, in turn, modulates another carrier before modulating a modulator that modulates a carrier that is modulating itself. Even if you didn't follow along with all that, it produces a rich sound that can be modulated further with an LFO, filter, and amplitude envelopes in the same way as analog subtractive synthesis.

While this approach might appear to make programming FM even more difficult, it's the exact opposite. The ability to arrange the operators in custom configurations removes much of the random results and makes it much easier to predict outcomes. It just takes a basic understanding of how fast pitch modulation affects a carrier.

It is best to consider the FM synthesizer as similar to a subtractive analog synthesizer. It features the typical ADSR envelope generators for the amplitude and the filters and also features some LFOs for modulation. The only significant difference is that rather than have square, triangle, and saw oscillators immediately available for use, you must first mix the FM operators to create these essential waveforms. This process to produce these is shown in Table 7.0.

Using the techniques discussed in Table 7.0 to create the typical oscillator waveforms, you can see that they can be mixed and modulated with the parameters

Table 7.0

Waveform	Created by
Sine wave	*You can use a basic operator because all operators are sine waves*
Saw wave	*Feedback an operator on itself. As you increase the self-modulation, it will produce a tone consisting of even and odd harmonics, much like a saw wave. If you continue to self-modulate the operator, it will move beyond a saw into a denser, almost distorted harmonic tone containing noise*
Triangle and square	*If the modulator is set to twice the ratio of the carrier (double the pitch) and the modulation amount is increased, it will produce a hollow timbre, typical of a triangle. As further modulation is employed, it will transform into a tone similar to a square wave*

that adorn all subtractive analog synthesizers. This makes the initial approach to programming any FM synth much easier.

The benefit of this approach is that you can not only predict results, you can also modify the waveforms in ways that are not possible on subtractive analog. Rather than just have access to the basic oscillators, you can feedback an operator on itself to create a basic sawtooth through to distorted stacked saws. Or by modulating an operator with a 2:1 ratio of differing amounts, it is possible to morph through a triangle into a square and beyond.

This approach is only the tip of the iceberg, since each operator features its own independent amplitude envelope. Using these opens up a further range of possibilities. For example, you could modulate a carrier with a 2:1 ratio to produce a square wave. However, if the modulator amplitude envelope were set to a long attack stage, the resulting timbre would begin as a sine wave, and as the amplitude of the modulator increased, the tone would gradually evolve into a square wave.

FM IN PRACTICE

To better understand how FM works, we will program a timbre in the FM8. For this, we'll reproduce the somewhat classic house *donk* style bass that's currently making the rounds again, this time in *Deep House*.

> You can download a demo of FM8 from https://www.native-instruments.com/ a preset of the sound we'll reconstruct, and a video tutorial building the sound is included in the resources package available from http://www.dancemusicproduction.com.

Despite being a staple in dance music, the *donk* bass is perhaps one of the most straightforward sounds to program in FM consisting of nothing more than a single modulator and carrier. An operator with a fast AD envelope is employed as a modulator set at twice the frequency of the carrier. Finally, some overdrive and shelving EQ is used to add some character to the bass.

With the FM8 on a new patch and in the *expert* view, the FM matrix on the right shows the six available operators, named A through F. Currently operator F is the only activate operator (it is illuminated), but further operators can be switched on or off by right-clicking (*ctrl-click* on a Mac) on the operator's letter.

In the current configuration, operator F is currently fed to the output of the synthesizer at a volume of 80. This is shown in Figure 7.1.

Since all the operators are sine waves and only operator F is fed to the output of the synthesizer, if a note is played you will hear operator F's sine wave. Turn on operator E by right-clicking (*ctrl-click* on a Mac) on the letter E and then click and drag in the light gray space underneath E but directly to the left of F. As you click and drag, the amount of operator E modulating operator F increases. This is shown in Figure 7.2.

Currently, since operator E is at the same ratio as operator F if the modulation amount is set to approximately 25, the synth will produce a sound similar to a soft-edged saw wave. However, if operator E is set to twice the ratio of F, it will create a timbre similar to a square wave.

With the ratio of the modulator now set to twice the frequency of the carrier, by increasing the modulation amount it's possible to create a more substantial sounding square wave. For a donk bass, a modulation setting of approximately

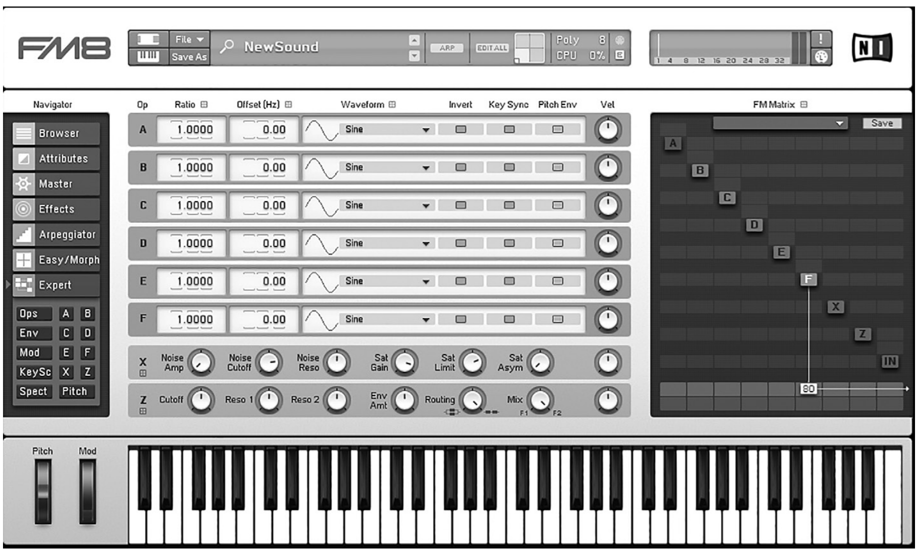

FIGURE 7.1
The FM8 basic patch. Note how operator F has a line to the output of the synthesizer (the light grey line at the bottom of the FM matrix)

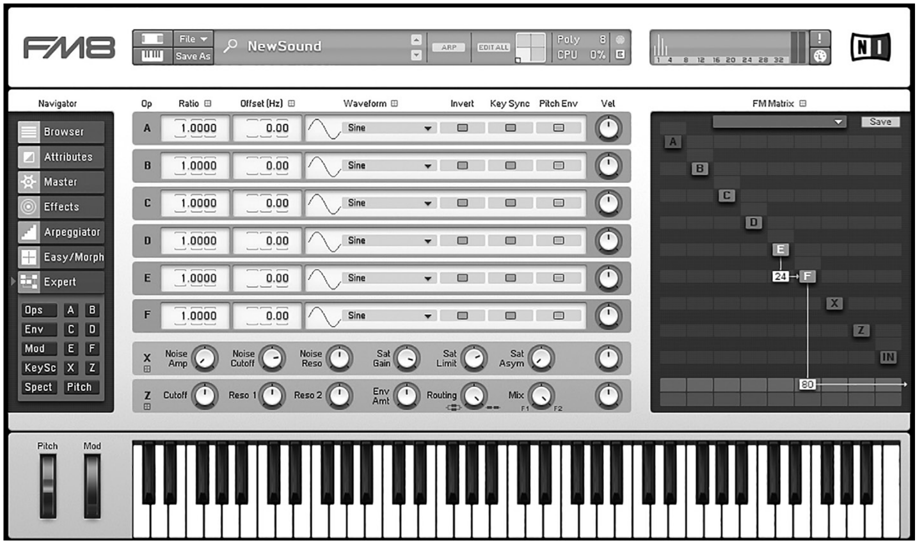

FIGURE 7.2
By clicking and dragging, you can increase the modulation of the carrier

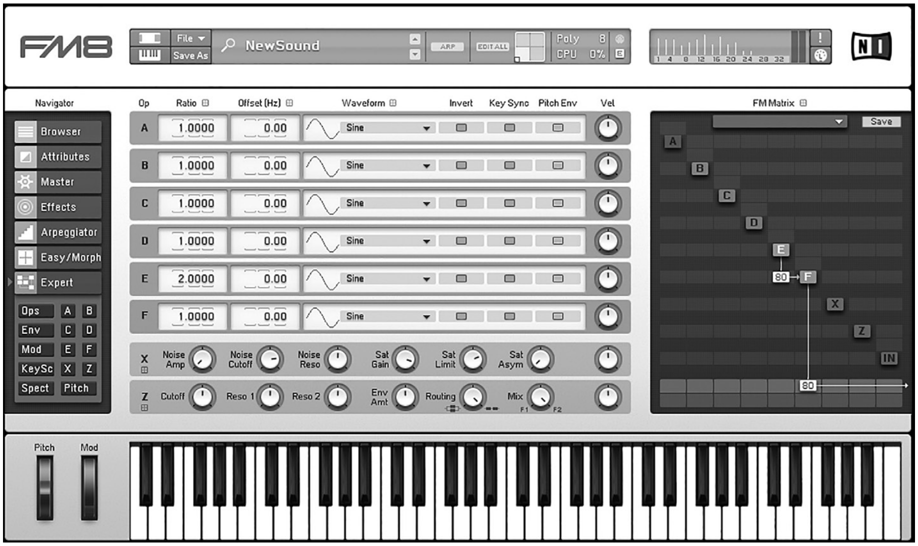

FIGURE 7.3
By doubling the ratio of operator E, it produces a square wave

80 is suitable. This produces a sophisticated, dense-sounding square that will serve as the basis of the sound.

To create some movement in the sound, we can modify the amplitude envelope of the modulator. This needs only be a simple AR envelope employing an immediate attack and a swift release.

Finally, some distortion and shelving EQ can be applied to add some character to the bass. If played at A0, it produces the classic donk bass.

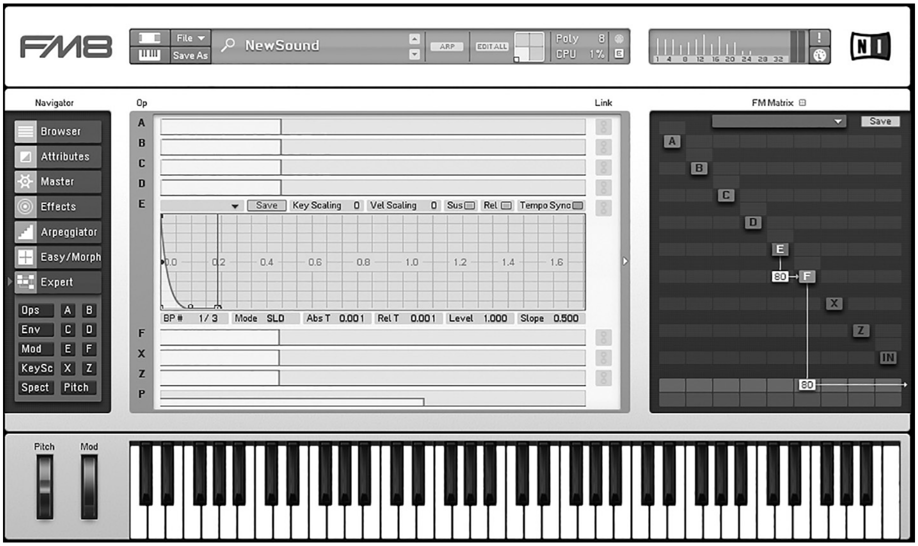

FIGURE 7.4
Employing a fast attack/release envelope on the modulator

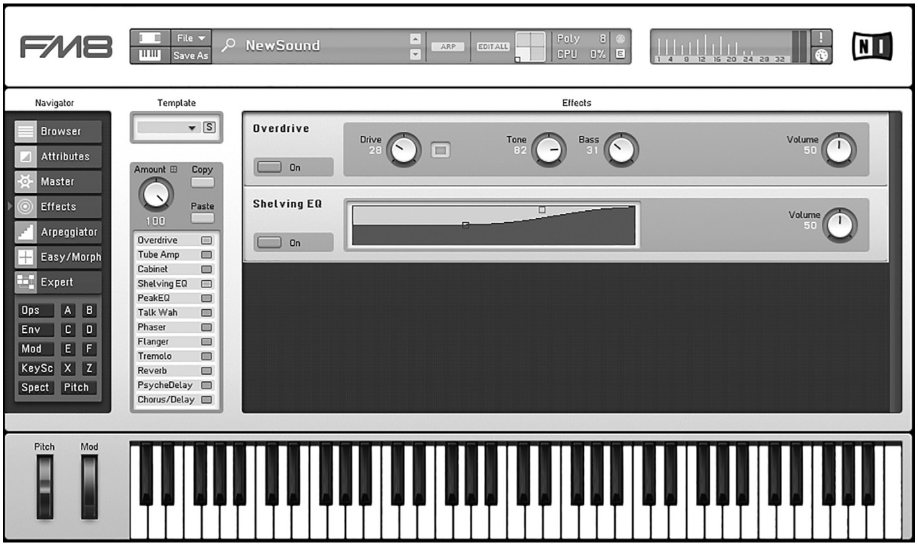

FIGURE 7.5
Applying distortion and shelving EQ

With the primary sound now created, it is worth experimenting further with the timbre. For example, operator D could be routed directly to the output of the synthesizer. This would allow you to mix a sine wave oscillator in with the output of operator F, or by self-modulating F, the carrier will introduce even harmonics into the signal, producing a mix of square and saw.

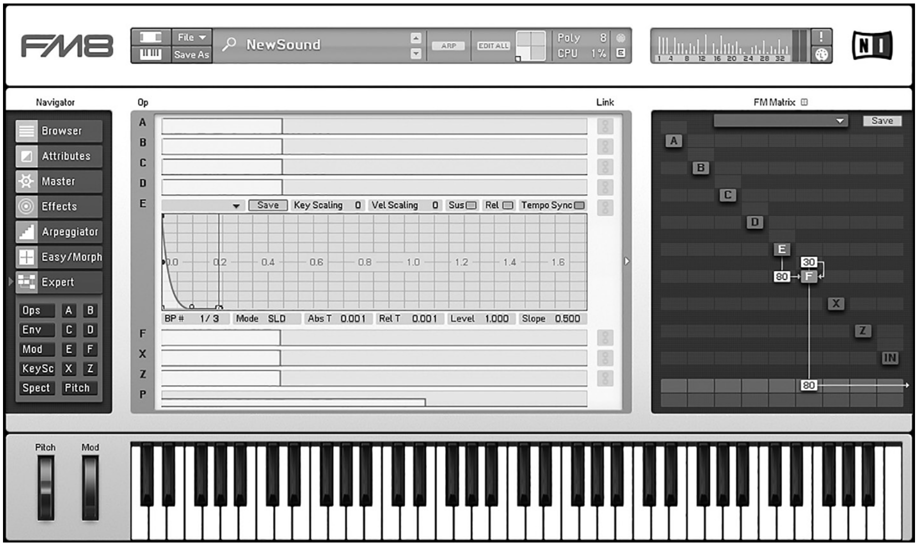

FIGURE 7.6
Self-modulating the carrier to produce even saw-like harmonics

This simple process can be applied when approaching any FM synthesizer. You can begin by producing the basic wave-shapes of subtractive analog and then experiment further through the use of each operator's amplitude envelopes, ratios, and modulations.

WAVETABLE SYNTHESIS

Wavetable synthesis is another digital form of synthesis, this time employing single cycle samples of waveforms and storing them in what are known as *tables*. An example of wavetable is where just *one individual cycle* of a saw wave is sampled from an analog subtractive synthesizer into a table. When a note is played on the wavetable synthesizer, it would merely repeat the recorded cycle of the saw wave over and over again, reproducing the sound from the analog synth. This approach saves on memory in the synth since it only requires a small amount to store the oscillations.

What makes wavetable such an exciting form of synthesis is that any single table can store more than one wave sample and in many cases can store anywhere from 64 to over 400. These individual waves can all be 'scanned' through using a *wavetable position* parameter on the interface. With this arrangement, wave 1 could be a saw wave, and wave 64 could be a square wave, with all the waves in between these gradually changing in harmonics. By turning the wavetable position parameter, it would be possible to morph progressively between the saw and square.

Alongside manually scanning through the wavetables, it is also possible to employ an LFO or an envelope to modulate the wavetable position. This permits you to quickly create a variety of automated harmonically changing, sweeping sounds.

Wavetable synthesizers often contain a large number of wavetables, each employing anywhere up to 400 individual cycle samples. While some of the tables will provide harmonically related samples, others include a variety of complex, harmonically unrelated waveforms allowing the producer to create complex, random, and unpredictable changes in the sounds. Some of the latest wavetable synthesizers, such as Xfer Serum, even permit you to import your audio files into wavetables.

Wavetable synthesis, particularly the latest incarnations such as Xfer Serum, Vengeance Avenger, and Native Instruments Massive are commonly responsible for the sounds heard in Dubstep, Future House and EDM (Mainroom).

Native Instruments' Massive, in particular, features a collection of growling bass waveforms contained in many tables. When the wavetable position and low-pass filter are modulated via a modulation envelope, it results in some of the most popular and recognizable sounds employed in some of the modern dance genres.

WAVETABLE IN PRACTICE

To better understand how wavetable works, we will program a simple timbre in Xfer Serum. For this, we'll create a harmonically rich and deep bass that we can sweep with some envelopes of filters. The sound could be typical of Techno, Tech House, or even Electro.

> You can download a demo of Serum from https://www.xfer.com/ a preset of the sound we'll reconstruct, and a video tutorial of building the sound is included in the resources package available from http://www.dancemusicproduction.com.

The first stage is to initialize the preset, so we're working from a blank canvas, you can do this by selecting *init preset* from the Menu.

Because we're creating a bass sound, we can take advantage of the sub bass oscillator offered by Serum. We need to activate this and choose a waveform that is used for the sub. You can select any waveform, but for this example, I'm going to use a triangle wave. I prefer to use triangle waves rather than sine waves for sub frequencies because it features more subharmonics that can add a sense of depth to the low end. Tune this sub-oscillator down by –2 octaves to add depth.

The first oscillator in Serum defaults to a saw wave. While we could use this, we could just use an analog synthesizer to achieve the same results, so let's pick a more exciting wavetable. There is no one particular wavetable suitable for this style of sound and it's worth just scanning through and listening to the different wavetables to find an interesting tone. I came across *gritty* that can be found under the *digital* menu.

We can use this wavetable for both oscillators in Serum, so switch on the second oscillator and use the same wavetable for this. Because we're creating a bass sound, we want to tune both oscillators down, so let us change the first oscillator to –2 octaves and the second to –1. Now it is just a case of scanning both wavetables to find something interesting in the table, I settled on wavetable 170 for oscillator 1 and wavetable 200 for oscillator 2.

At this stage, you can experiment with unison, detuning, blend, phase, and all the parameters available in the oscillators to produce a pleasing tonal quality. To keep this sound relatively simple for the book, I chose not to apply anything to the oscillators beyond the detuning.

Now we have an exciting harmonically rich sound, we can filter it. My favorite filter is the German low-pass filter that is hidden away in the *other* menu of the filter. The sweep of the filter is particularly resonant and sounds warm, yet controlled, especially when a small amount of resonance is applied.

FIGURE 7.7
Start by adding the sub-oscillator to Serum for the quintessential low end

FIGURE 7.8
The gritty wavetables are detuned by octaves, and then you can scan through to find an exciting sound in both

FIGURE 7.9
Filter the oscillators; ensure that both wavetables are filtered by clicking on both A and B in the filter section

This produces the basic bass timbre but it requires modulating to add some excitement. Using Serum, the sound can modulate in numerous ways but for many dance genres, it's worth modulating the amplitude to produce a pulsing side-chain effect. We can do this by choosing the first LFO and drawing in an interesting envelope with the mouse.

Once you have an envelope for the LFO, it can be used to modulate the amplitude in the matrix page. In here, select *LFO 1* as the source and *amp* as the destination, then set the modulation amount to around 75%. This produces the pulsing sound typical of many genres.

We can add further modulation to sweep the filter and the wavetable position. For the filter, I used LFO 2 with a different envelope and modulated the cut-off by

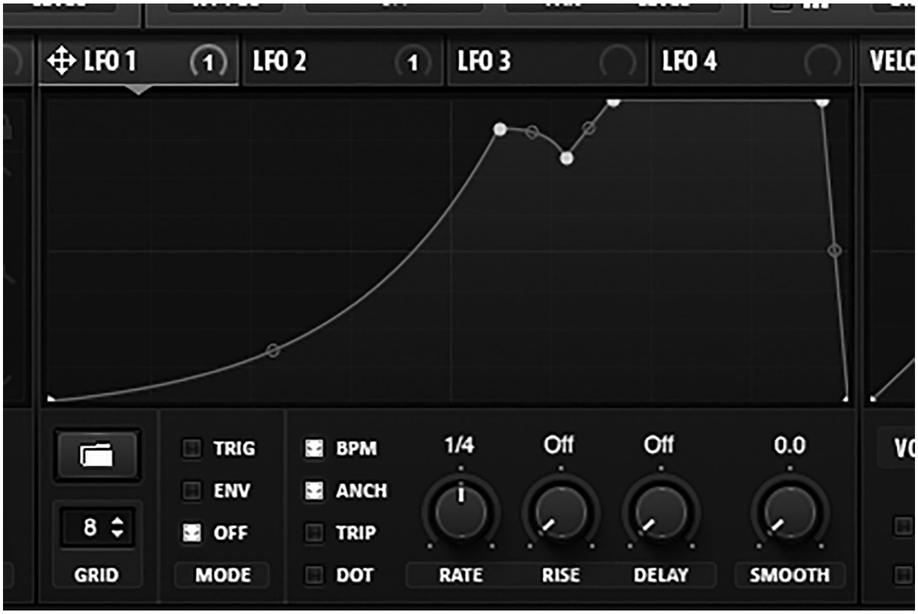

FIGURE 7.10
Draw an envelope for the LFO to modulate the amplitude

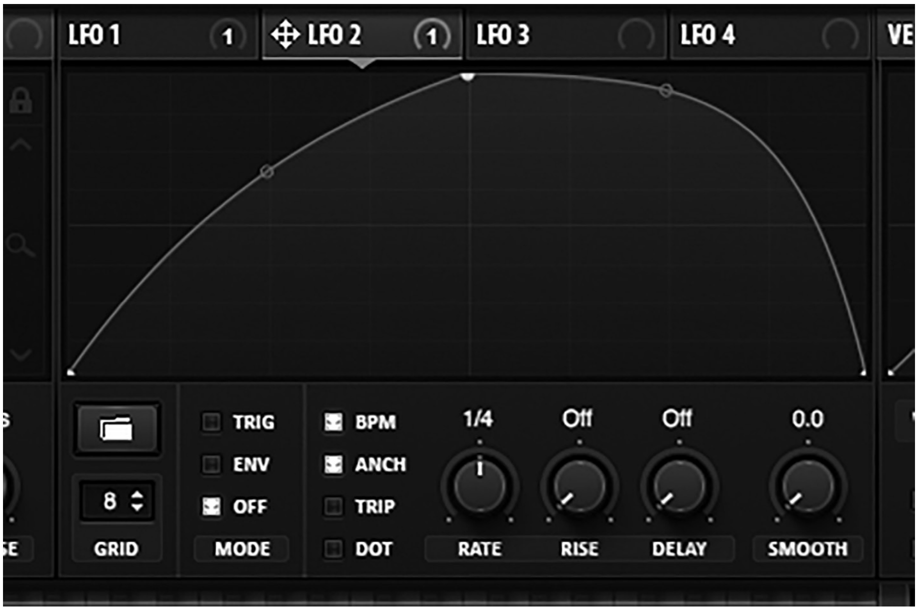

FIGURE 7.11
Draw an envelope for the LFO to modulate the filter

approximately 32. You can modulate the filter either in the matrix of Serum or by clicking and dragging the *arrowed plus symbol* next to the name of the LFO and dropping it on the filter cut-off. You can increase or decrease the amount of modulation by clicking and dragging on the circle on the top left of the filter cut-off control.

Modulation is the fundamental principle behind any exciting sound so you can experiment further using envelopes or further LFOs to modulate other parameters. I used another LFO to modulate the wavetable position of oscillator one. By doing so, the sound morphs and changes in exciting ways as the wavetables are swept by the LFO.

Finally, we can add some effects to the sound. In the example, I employed some tube distortion, a small amount of controlled reverb, and some EQ.

You can adopt this process when creating any sounds within any wavetable synthesizer. Just select a new wavetable, detune, apply filters, and modulate different parameters until it produces an interesting sound. Serum, in particular, lets you also import your audio samples so it's always worth experimenting with dropping in percussive sounds, vocals, and sounds designed on other synthesizers to then further modulate and effect in Serum's engine.

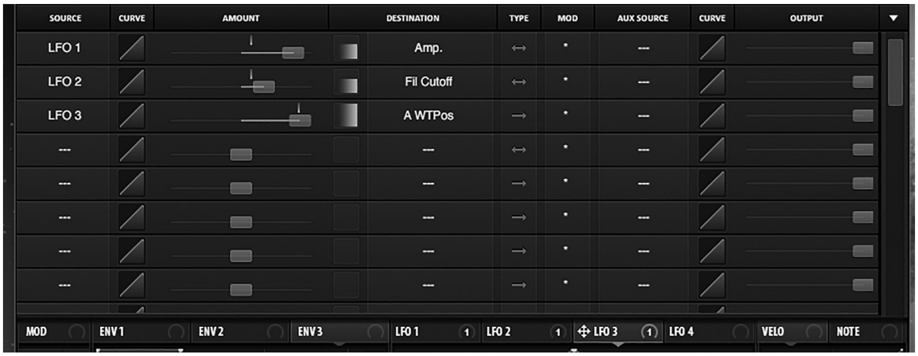

FIGURE 7.12
Draw an envelope for the LFO to modulate the wavetable

FIGURE 7.13
Adding effects

CHAPTER 8

Modular synthesis

Got maths…?

Every Eurorack owner

Over the past 10 years, modular synthesis has enjoyed something of a resurgence, with artists such as DeadMau5, Junkie XL, and Scanner all employing it in their music. The revival of the format shows no signs of abating either, with more and more dance musicians entering the fray to create tones and textures that just aren't possible any other way.

Modular synthesis was born in the 1960s, a result of the work of Bob Moog on the east coast and Donald Buchla on the west coast. Both were simultaneously developing instruments for the electronic musician. Bob Moog's approach was conventional. He would ask musicians what they wanted in a synthesizer because he wanted to create a synthesizer that was immediately playable and would appeal to musicians. Donald Buchla, by way of contrast, took a far more experimental approach. He was more interested in producing machines (he didn't like to call them synthesizers) that could be used to create new and innovative generative styles of music.

Both, however, employed the same principle. Several modules such as oscillators, filters, envelopes, sequencers, and LFOs were all connected to a single power supply situated in a case. The inputs and outputs of every module were featured on the front panels and had to be connected via 1/4″ patch chords by the operator. While this may appear counterproductive to music, it offered the artist an entirely new experimental soundscape.

However, while some musicians took to this new modular format, for many, it proved to be confusing and complicated. This, combined with the expense of these synthesizers marked their eventual demise. In 1970, the Minimoog Model D put the final nail in the coffin. The Model D's connecting circuitry was hard wired, and the synthesizer produced sound without the need to connect modules together. The first-ever 'plug and play' synthesizer went on to become adopted by musicians everywhere.

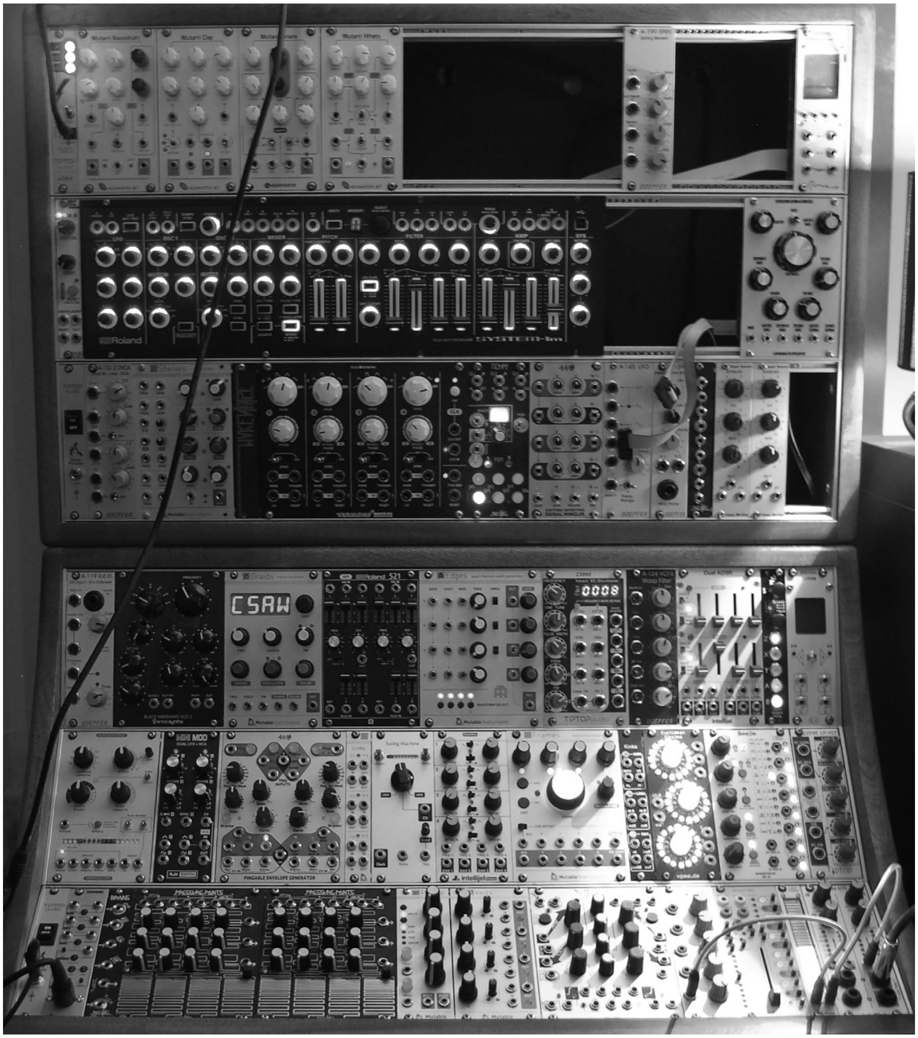

FIGURE 8.0
The author's modular rig

It wasn't until 1995 when Dieter Doepfer released the A-100 analog modular synthesizer and the *Eurorack* format that modular began to regain its popularity. Although there were numerous modular formats available such as Buchla and Serge, various manufacturers quickly adopted Doepfer's standard, and Eurorack became the most popular.

The Eurorack standard consists of modules that all share the same power requirements and transmit and receive the same control voltages (CV) via 3.5mm patch cables. The mass appeal of this format is, most likely, because there is a varied selection of oscillators, filters, VCAs, envelopes, etc. available, and many are modeled on old analog synthesizers. As these can be connected in any way you see fit, you could plug an analog oscillator from a Moog into a filter from a TB303 and into envelopes and LFOs from a Prophet 5. Moreover, if you have the necessary soldering skills, you can even purchase kits and build DIY modules.

If you want to enter into the world of modular, you first require a case and a power supply. Eurorack cases are arranged in rows of three units high and are measured horizontally via horizontal pitch. So when purchasing a case, you need to decide on the height and the width. While modules are all the same height, the width is not standardized. Even though this gives manufacturers free rein to lay out their interfaces however they please, it is equally annoying because as you fill a rack with modules, you're often left with a small gap at the end that no module will fit into.

Power for the case is available as a distribution bus board or as a flying distribution bus. The fixed bus board is the preferred option for many enthusiasts because it is a solid circuit board that attaches to the rear of the case and a single power supply can then be used to power up to three or four of these bus boards fitted in the same rack.

A flying bus is more suited towards starter kits because each bus requires a power supply. They are also susceptible to noise interference, and you have the disadvantage of powered ribbon cables hanging around the back of the case. In both cases, bus boards are powered via a 12 volt plug-in transformer.

FIGURE 8.1
The 'Peter' 12U modular case from www.synthracks.com

FIGURE 8.2
A fixed bus board, a flying bus board, and a power transformer

Modules connect to the bus board via ribbon cables, and many modules require +12v DC to operate. However, some digital modules also need an additional +5v. Some bus boards supply both +12 and +5v to modules that require it, while others may only provide +12v. In these instances, you will require an additional power supply for any modules that also need the extra +5v.

Connecting modules to the bus board is a simple case of connecting the two via a ribbon cable supplied with the module. This is a simple affair, the only thing to note is that the −12v connection should always remain at the bottom. Most (but not all) manufacturers help with identification by color coding the −12v link on the ribbon cable in red.

While each bus board will commonly feature 14 or more power connections, you must consider the load placed on the board by your modules. Each module will have a specific power consumption rated in milliamps (ma). Although many bus boards are rated high enough to supply power to multiple power hungry modules, it is something you should always confirm before fitting. If a module doesn't receive its required power, it can behave in strange and unexpected ways. Many manufacturers will display the power consumption of their module in the specifications.

Once you have the case and the power supply, you can begin purchasing and connecting modules to your rig. The choice of modules is extensive, and you'll need to investigate and read about each to decide on what would suit you the best. However, to create a basic synthesizer, you will need at an absolute minimum of the following:

- an oscillator
- a VCA
- a filter
- an envelope
- an LFO
- a gate trigger
- an output module
- a collection of 3.5mm mono cables.

Modular synthesis consists of two signal types: audio and control voltages. Audio signals are similar to control voltages; the only difference is the polarity and the voltage. A **C**ontrol **V**oltage is commonly 5 volts or under and is a bipolar signal. This means that the voltage will swing from +5 volts to –5 volts. This bipolar action makes it suitable as a modulator and so CVs are often the output from LFOs and envelopes to modulate other modules. This is not always the case, however. In some cases, a CV may exhibit a frequency fast enough to produce an audible oscillation! This is one of the many exciting capabilities of modular: audio can sometimes be used as a control voltage, and a control voltage can sometimes be applied as an audio signal.

To better explain modular in context, we will build a simple synthesizer using a modular rig consisting of the modules mentioned above. I'll be using the free software called *VCV rack* to show the basic operation of a modular system. You can download this for free at https://vcvrack.com. It is available on Linux, Windows, and Mac OSX and I would strongly suggest downloading the software and following along to understand the basic principles of modular.

We should begin by installing all the modules we'll require to create a fundamental synthesizer. Once VCV is open, right-click anywhere in the main rack window and choose the following modules:

- Core > MIDI to CV
- Core > audio interface
- Fundamental > VCO-1
- Fundamental > VCA
- Fundamental > VCF
- Fundamental >ADSR
- Fundamental > envelope-1.

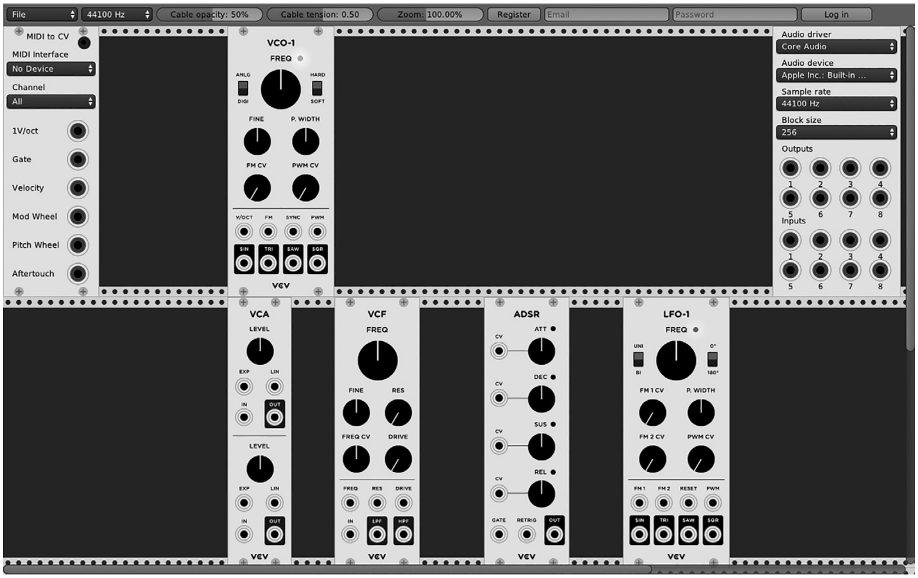

FIGURE 8.3
The basic arrangement

THE OSCILLATOR

Modular synthesis starts with a sound source: the oscillator. On this, you can see the functions you would expect from a typical analog-based oscillator. It features frequency and fine-tuning parameters, pulse width, and, at the very bottom, the synthesis waveforms discussed in a previous chapter: sine, triangle, saw, and square. These are the signal outputs from the oscillator.

We cannot connect these outputs direct to an audio interface or monitor speakers, however. The signals within modular are far higher than the audio signal expected by studio devices such as audio interfaces and monitor speakers, so we require an output module. This will convert the high voltages of modular into a much smaller voltage suitable for an audio interface while also turning the 3.5mm into a 1/4″ jack.

With just these two devices, the most straightforward connection we can make is to connect the output of the oscillator to the input of the output module (the output of which is connected to the audio interface). In this example, we'll connect the output of the square wave to the input of the output module.

The moment we make this connection, we hear the square wave. And, by adjusting the frequency, the fine-tuning and the pulse-width modulation, we can affect the oscillator's sound. The problem, however, is that the oscillator's sound is continuous at the output. A typical synthesizer will only make a sound when we press a note on its piano keyboard, so we need the same arrangement here. To do this, we require a VCA.

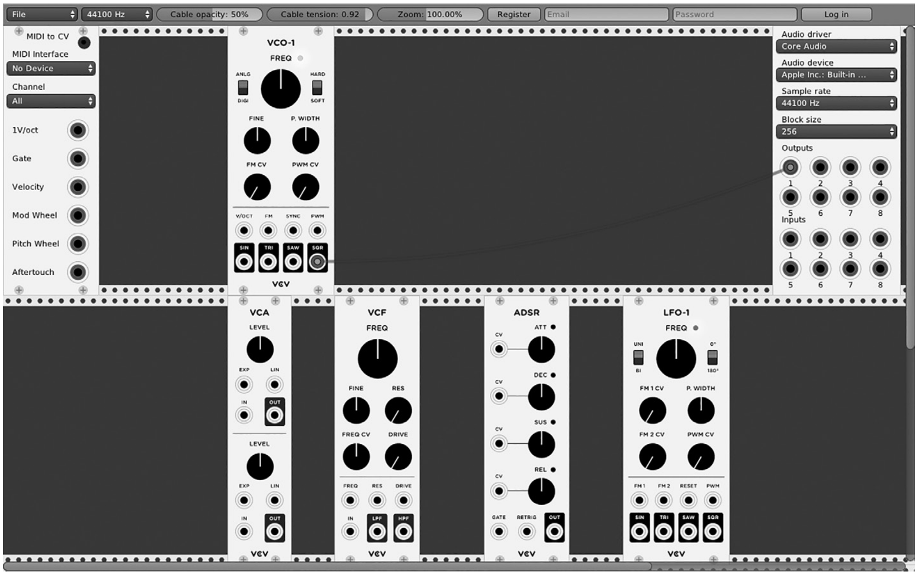

FIGURE 8.4
The oscillator connection

VCA

A VCA is a **V**oltage-**C**ontrolled **A**mplifier that, in modular, also acts as a gate. To introduce the VCA into our set-up, the square wave output of the oscillator is connected to the input of the VCA, and the output of the VCA goes into the output module. By doing so, the large dial on the VCA now acts as a volume control for the oscillator.

Note, however, that even with this arrangement, the oscillator remains continuous and audible. We must configure the modular system so that the oscillator is only audible when a note on an attached controller keyboard is pressed. To do this, we'll use the MIDI to CV converter.

The MIDI to CV converts incoming MIDI information from a piano keyboard or the DAW into a series of control voltages. The first connection we're interested in is the gate output of the MIDI-CV. The gate output is a square wave voltage, so when it receives a MIDI message, it sends a control voltage, and when the note is released, this voltage stops.

By connecting the MIDI-CV gate output to the linear or exponential input of the VCA, the VCA will react to the voltage it receives. When the VCA does not receive a control voltage from the MIDI-CV, the gate between the oscillator and the output is closed. This makes the oscillator inaudible. However, when the VCA receives a control voltage the gate opens, and we can then hear the oscillator.

This arrangement has now produced a simple note on/note off procedure. There is silence until we press a note on the controller keyboard. If, however, we want to shape the amplitude of the sound over time, we must introduce an envelope.

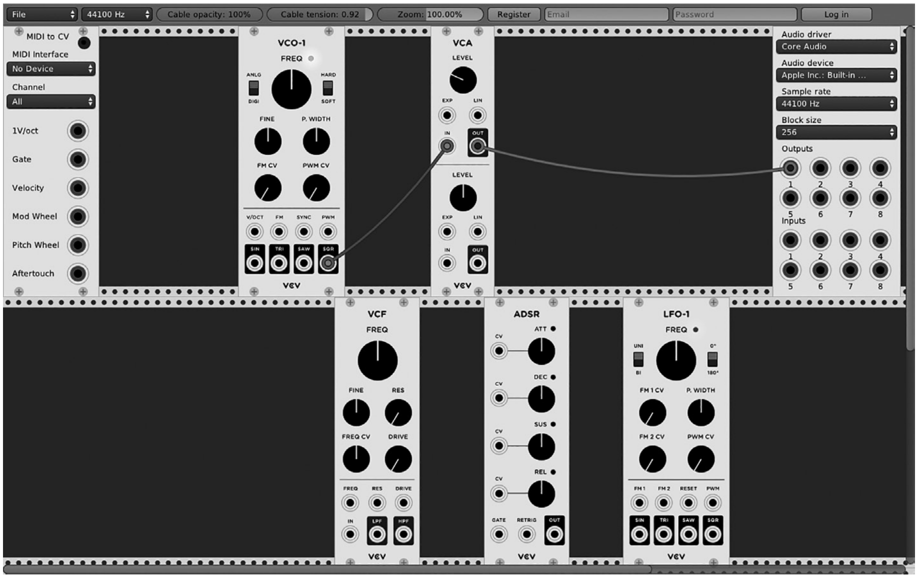

FIGURE 8.5
Adding a VCA

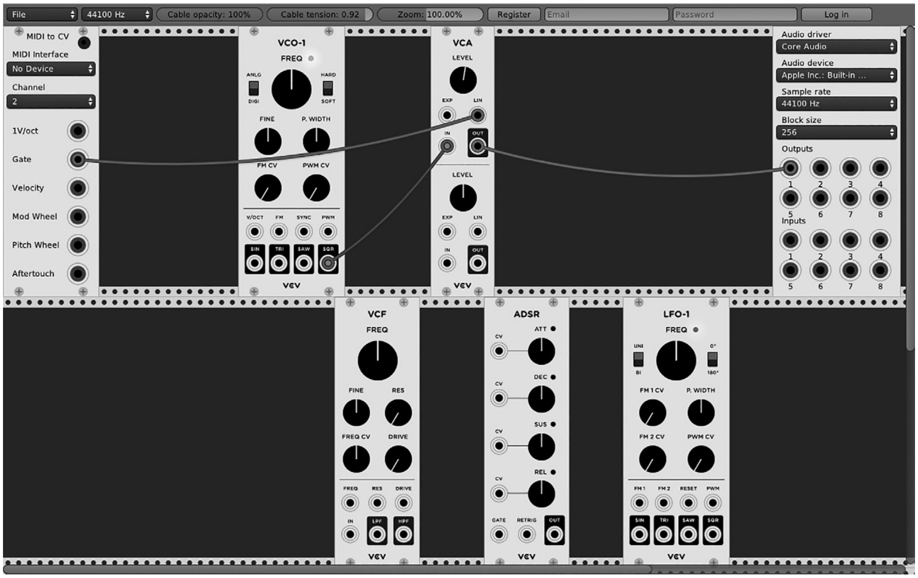

FIGURE 8.6
Adding MIDI to CV

An envelope creates control voltages, but unlike a gate that exhibits a simple on/off action, an envelope sends continuously changing voltages that are determined by the envelope parameters. If there is a long attack stage on the envelope, the envelope will gradually increase the voltage.

An envelope requires a trigger signal for it to begin its function. For this we can take the gate trigger output from the MIDI-CV and rather than have it connect directly to the VCA, we can attach it to the envelope instead. This way, whenever we press a note on the controller keyboard, the gate signal starts the envelope cycle and this will continue for as long as the note is held. The control voltage output of the envelope must now enter the VCA.

In this configuration, a MIDI note triggers the gate to the envelope. Provided the envelope receives a constant voltage (i.e., you keep the note pressed), it will produce different voltages depending on how the envelope is configured. These voltages modify the action of the VCAs, gradually opening and closing it, that as a result, reshapes the dynamic content of the square wave oscillator.

You may have noticed that regardless of the pitch played on the controller keyboard, the pitch of the oscillator remains constant. To remedy this, we must send a control voltage to the oscillator to determine the pitch. We can do this by connecting the 1V/Oct output of the MIDI to CV into the V/Oct input of the oscillator.

With this connection, the different notes on the attached controller keyboard are converted into differing voltages by the MIDI-CV. And for every additional volt, the oscillator will increase by an octave; hence the *1v per octave* printed on many modular oscillators.

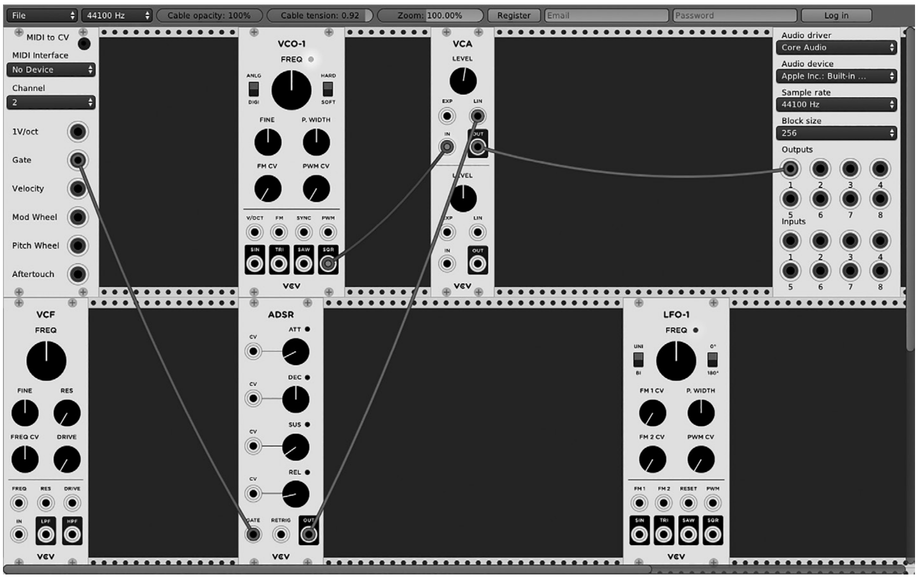

FIGURE 8.7
Adding an envelope

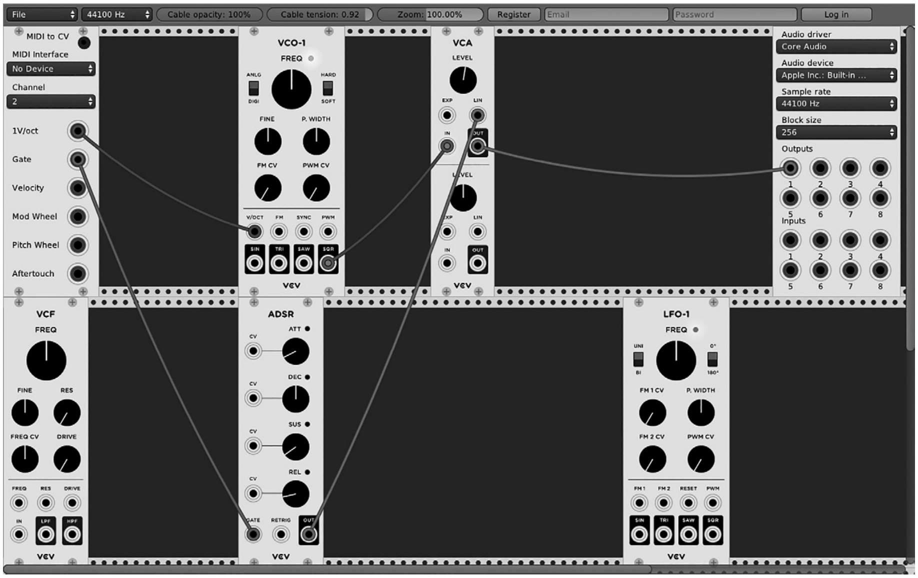

FIGURE 8.8
Adding pitch

FILTERS

We now have the beginnings of a synthesizer but it lacks any tonal modification options so let's now add a filter to our synthesizer. We want to filter the harmonics of the oscillator so the most straightforward connectivity would be to send the output of the oscillator into the filter. To do this we connect the square output of the oscillator into the input of the filter, rather than the VCA, and as we want to use a low pass filter, we connect the LPF (**L**ow-**P**ass **F**ilter) output to the input of the VCA.

Now, the signal leaves the oscillator and enters the filter, and the filtered signal now enters the VCA that is controlled by the envelope. Using the filter, we can remove the upper harmonics from the sound, and we have a rudimentary synthesizer. It consists of a single oscillator, a filter, and an envelope to sculpt the sound behavior over time.

Static sounds are uninteresting, however, so let's add some modulation for further interest. Many modular LFOs will modulate quickly enough to be used as a sub-oscillator, but here we're going to use it for its original intention, to modulate any parameter that accepts a CV input.

The most straightforward configuration here would be to use the LFO to modulate the pulse width of the square, producing a pulse-width modulation effect. We can do this by connecting sine wave output of the LFO to the PWM input of the oscillator. By now increasing or decreasing the rate of the LFO, we can control the speed of the pulse-width modulation.

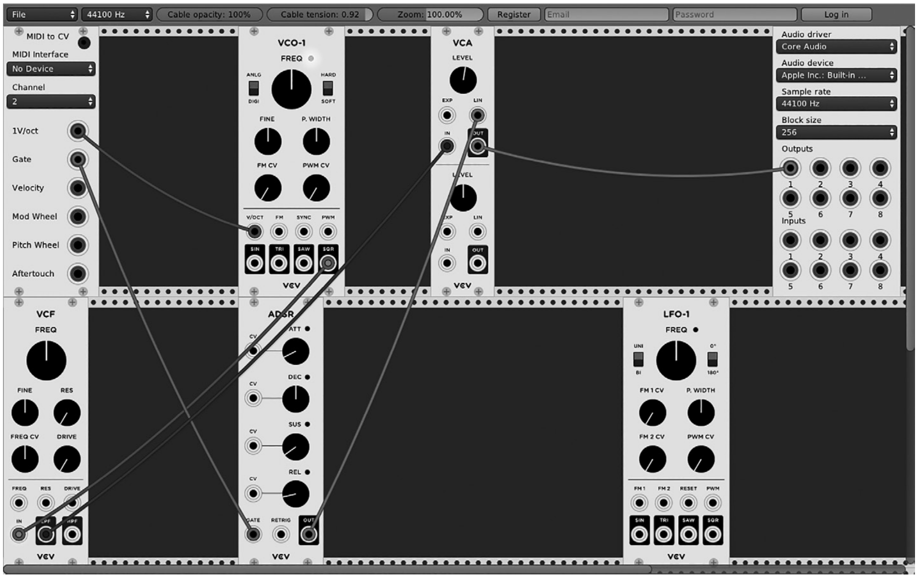

FIGURE 8.9
Adding a filter

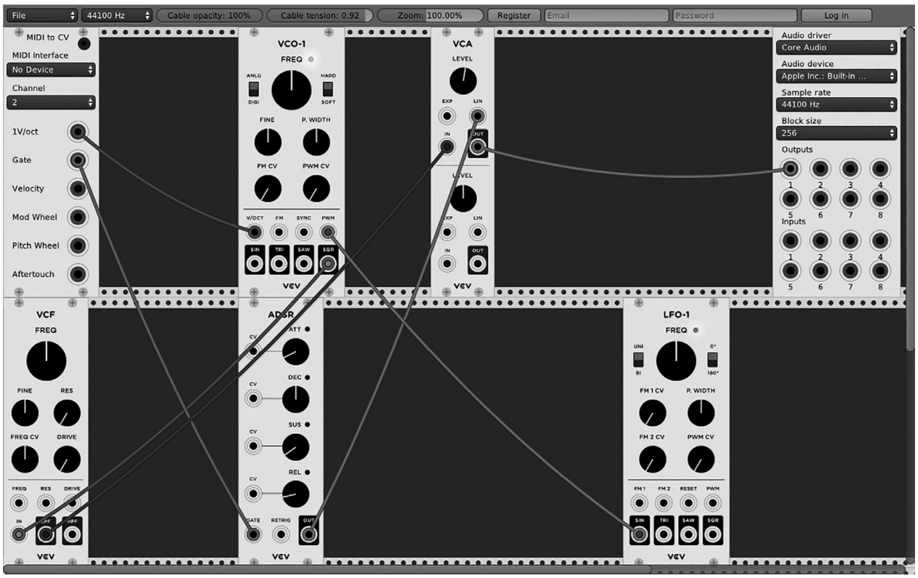

FIGURE 8.10
Adding some modulation

Because the LFO has multiple outputs – one for each waveform just like an oscillator – we can use the same LFO with a different waveform to modulate another parameter. Here, we could use the saw wave to modulate the cut-off frequency of the filter. We can accomplish this by connecting the saw wave output of the LFO to the *frequency* input of the filter.

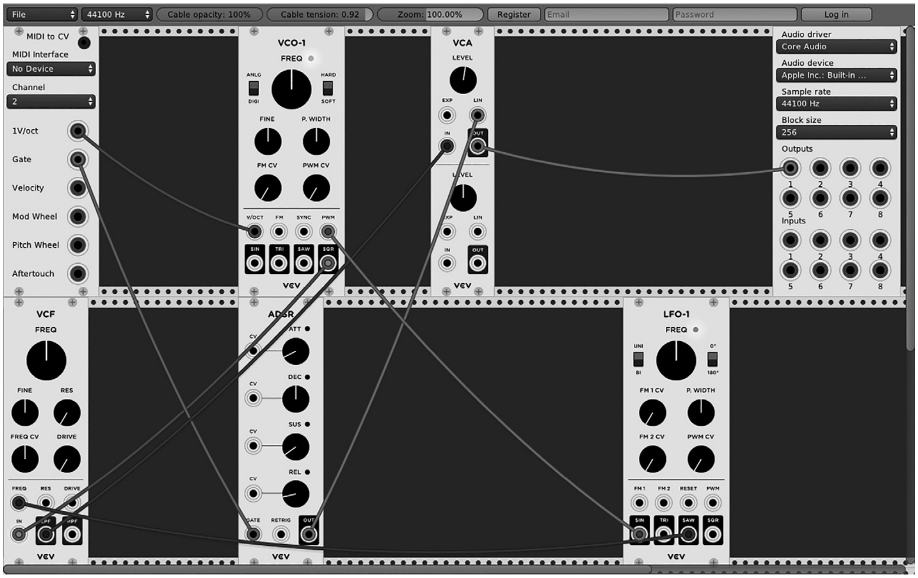

FIGURE 8.11 Adding further modulation

We've now created a basic synthesizer patch. It's a single pulse wave oscillator, with pulse-width modulation, a filter, and a modulated low-frequency filter. This, however, is only the tip of the iceberg. From this basic synthesis 'patch,' we can experiment further.

For example, the LFO could connect to the FM input of the VCO to produce FM, or you could take a different waveform from the oscillator (such as the triangle wave) and feed that signal into the oscillators FM input.

Alternatively, we could take the LFO's triangle wave output and connect it to the CV input of the attack portion of the envelope. This way the rise and fall of the triangle would modulate the surge of the attack time. Or, we could use the square wave to retrigger the entire envelope, by doing so, every time the square wave reaches the high state, the envelope would restart again.

Of course, all of the modulations will move at the same rate because we're using the same LFO. However, we could employ a secondary LFO. And if we wanted more voices, we could introduce a second oscillator, or if we wanted to control the action of the filter with an envelope, we could add another envelope.

Instead, we could output the square wave of the LFO into a clock divider – this would multiply the clock into a series of subdivisions, and these different divisions could be used to retrigger other modules, or if the LFO frequency is fast enough, we could even use it as a sound source.

There is no limit to the connection options: oscillators can be used as modulators and modulators can be used as oscillators. Oscillators can modulate themselves or other functions, or specific stages of envelopes can be used to trigger

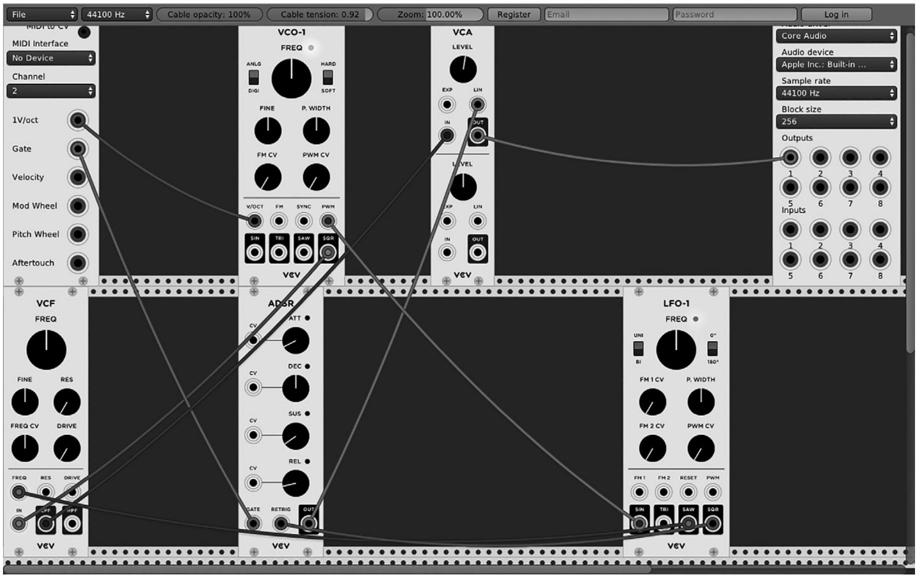

FIGURE 8.12
Restarting the envelope

and time different events. For example, the EOR (**E**nd **O**f **R**ise) of an attack stage could trigger an LFO start while the EOF (**E**nd **O**f **F**all) of a decay stage could trigger another event.

There are many modules available for the Eurorack system, covering everything from wavetables through to sequencers, creative modulation options, granular synthesizers, logic modules, and samplers, through to spring reverbs (with real springs that can be plucked), delay units, and microphone input modules.

Modular is a system that can (and will) grow and expand with your needs. It offers a quick hands-on approach for synthesis that permits you to create new sounds that even if another artist had exactly the same set-up as you, she would struggle to replicate them.

The only disadvantage is that modular doesn't enable you to save presets, and it is challenging to recreate sounds you've previously created unless you take photographs of the set-up and it is not too tricky to decipher. It is, however, an ever-growing and evolving scene that while not yet fully embraced by all artists, is working its way through all genres of dance music.

CHAPTER 9

The theory of sound design

To hear a sound is to see a space.

Louis Kahn

An understanding of synthesizer architecture is only part of a much larger puzzle. We next need to know how to create interesting sounds with them and then how to process and mix these sounds. Of all these, sound design is perhaps the most difficult to master. Not only will the sounds you design dictate the genre, but they will also determine the overall quality of the music. In fact, poor sound design is *the* main contributing factor that separates an amateur production from a professional.

Almost every synthesizer is supplied with any number of sound presets ranging from 30 to over 800. These are designed to show the extent and capabilities of the synthesizer but, unfortunately, many novice (and, perhaps even more unfortunately, many professional) producers rely solely on these. While this approach provides immediate results, it removes the exploration and eureka moments, and often results in bland music suitable only for elevators.

Great music is born from an artist's passion. This energy somehow manifests itself in the music and elicits an emotional response in listeners. If 300 artists before you have used the same synthesizer preset as you, why should anybody care? Besides, it's hard to find the fun in spending hours surfing preset after preset in countless synths in an attempt to find something that fits with your current project. Just as a guitarist must learn how to play the guitar before he can produce something unique, so should you learn how to design your own custom sounds.

Sound design can be divided into three distinct categories: audio design, synthesis design, and processing/effects chain order. Although all three are inextricably linked, the most difficult to master is synthesis. The first step towards mastering this art is to set time aside to experiment and to learn the characteristics of your synthesizer. While it is tempting to dive into creating music, the time taken to learn the characteristics of any particular synthesizer is ultimately rewarding.

Learning the characteristics of a synthesizer can be accomplished in a matter of days and can begin with simple exercises. The first should be to familiarize yourself with the oscillators and how they sound. On many subtractive analogue synthesizers, this isn't a difficult challenge because they feature a limited number of waveforms.

With wavetable synthesis such as *Xfer Serum*, *Native Instruments' Massive* or *Vengeance Avenger*, this can prove a tad more challenging. These feature anywhere from 30 to over 400 wavetables and auditioning each can become tedious. It should therefore be divided into separate, simple exercises.

On the first day, it would be advisable to remain with 20 wavetables (or the oscillators in an analogue subtractive synth) and attempt to create interesting sounds with just these limited options. Then on the second day, experiment with a further 20 wavetables and so forth, until all of the options have been exhausted. Of course, this will take time but the better you understand the character of the synthesizer, the more competent you will become on it.

SOUND DESIGN APPROACH

In order to program great sounds, you must first have some idea of the direction and style of the tone you want to create. Without this, programming degrades into guesswork without direction. If your ears are not yet trained to listen critically to music, knowing where to start is perplexing because you have little perspective to work from. If this is the case, I suggest you select a piece of music you are familiar with, pick out an instrument, and attempt to describe its sonic texture. This exercise will aid you in synthesis design and also improve your critical listening skills.

Many leads and basses in current dance music consist of layered instruments: three or four different instruments sounding together to create a rich timbre. So when starting, simplify the procedure by only taking note of how the sound behaves over time dynamically and texturally. For the timbre, you can experiment with mixing the oscillators or wavetables together with modulation to produce interesting sounds.

The first step is to conceptualize the character and momentum of the instrument through careful listening. You can ask the following questions:

- Does the timbre start immediately?
- How does it evolve in amplitude over time?
- Does it sound plucked, struck, blown, or bowed?

- What happens to its pitch when it's sounded?
- Does it continue to ring after the notes have been sounded?
- How bright is the sound?
- How much bass presence does the sound have?
- Does it sound hollow, rounded, gritty, or bright?
- Does the brightness change over time?
- Is there any modulation present?
- What does the modulation do to the sound?

Table 9.0 can help with further help with conceptualizing sounds.

Table 9.0

Generalization	Technical term	Synthesizer parameters
Type of sound	Harmonic content	Oscillator's waveforms/ wave shapes
Brightness	Amplitude of harmonics	Filter cut-off and resonance
Timbre changes over time	Dynamic filtering	Filter's envelope or/and LFO modulation
Volume changes over time	Dynamic amplitude	Amplifier envelope
Pitch	Frequency	Oscillator's pitch
The sound has a cyclic variation	LFO modulation	LFO waveform, depth, and rate
Tremolo (cyclic variation in volume)	Amplitude modulation	LFO augments the amplifier
Vibrato (cyclic variation in pitch)	Pitch modulation	LFO augments the pitch
Sound is percussive	Transient	Fast attack and decay on the amplifier
Sound starts immediately or fades in	Attack time	Lengthens or shortens the attack and decay stages
Sound stops immediately or fades out	Release time	Lengthens or shortens the release of the amplifier
The sound gradually grows 'richer' in harmonics	Filter automation	LFO augments the filter cut-off of ADSR modulation on the filter

In order to help with the identification and recognition of sounds, we'll reproduce a timbre that you can hear on the additional resources pack available to download from http://www.dancemusicproduction.com.
DMM resource: Example 9.0 Mp3 – The reference sound.

Listening to the reference material, there are some assessments that we can form. The first is the dynamic behavior of the timbre. The initial transient of a note is by far the most critical aspect of any tone. The first few moments of the attack stage provide the listener with a significant amount of information and have a substantial influence on the timbre's characteristics.

We perceive tones with an immediate attack to be louder than those with a slower attack. Equally, we perceive any short transient style sounds to be quieter than sustained sounds with the same amplitude and frequency content. Therefore, if a timbre appears 'slack' when played, something as simple as reducing the attack and release stage can make it appear more defined.

The transient behavior of the different melodies will strongly affect the energy and flow of the music. Short transient notes make music appear faster while longer notes slow the pace of the music down. This is not to suggest that if you wish to produce a fast track that all instruments should exhibit a short transient behavior.

Electronic dance music relies on the contrasting behavior of the timbres and we should not only think in term of tone but also time. With a fast attack and slow release on some instruments and a slow attack and rapid release on others, it will result in a mix that exhibits contrasting rhythms. For instance, a quick 1/16th pattern bass line can appear faster by introducing a slow evolving pad into the background. In many cases, music that exhibits a fast-paced groove will employ a fast transient on most instruments while slower paced genres will use longer, drawn-out notes to pull the groove back.

An amplifiers decay and sustain can add further time-based characteristics to a tone. Decay is perhaps best viewed as part of the creation of the transient as this controls the time it takes for the timbre to reduce in amplitude before it encounters the sustain. As discussed previously, sustain can effectively be considered a gain parameter that defines the amplitude of a note for as long as the note is held. So, if sustain is set to maximum, there can be no decay because there is nothing to decay towards. Similarly, if the sustain is set to minimum, then the release parameter would be redundant. Here, the decay would act as the release parameter and so there is no gain for the release to act on.

With short transient style sounds and motifs, both amplifier decay and sustain have little influence on the timbre. Rather, these parameters are suited towards the creation of long, developing timbres such as pads and strings. Here, the decay and sustain are employed with a long attack stage to produce a timbre that rises and falls gently in volume while modulated or augmented further with filters and modulation.

Once the overall shape of the amplifier is modified, the filter envelopes and filter cut-off/resonance can be used to alter the tonal content of the timbre. It is beneficial to determine the envelope of the filter first, and then afterwards adjust the cut-off and resonance of the filter. This is because the envelope will define the action of the cut-off.

A filters envelope is the same as an amplifier envelope, consisting of an ADSR action. But rather than modify the volume, this envelope modifies the action of the filter. If the attack is immediate, the filter will act on the sound on key-press. If the attack is longer, then the filter will sweep open up into the tone. More important, however, is the filters decay because this determines how quickly the filter falls to the sustain.

In the context of a timbre, filter decay contributes to the *pluck* of a sound and is a character-defining parameter. The modulation of this decay parameter whether via automation or via a secondary envelope, is responsible for the Progressive House pluck leads. And is equally accountable for the squelch and plucked basses alongside the many dance pluck style timbres throughout almost every genre.

With progressive pad and string sounds, the filters attack and decay are set longer than the amplifiers attack and decay rates. This is because, as the amplifier envelope begins the sound, the filter slowly sweeps in, resulting in harmonic movement throughout the sustain portion of the amplifier's envelope.

A filter envelopes sustain and release behave in the same manner as the amplifiers, but the envelopes sustain will determine the cut-off amount of the filter.

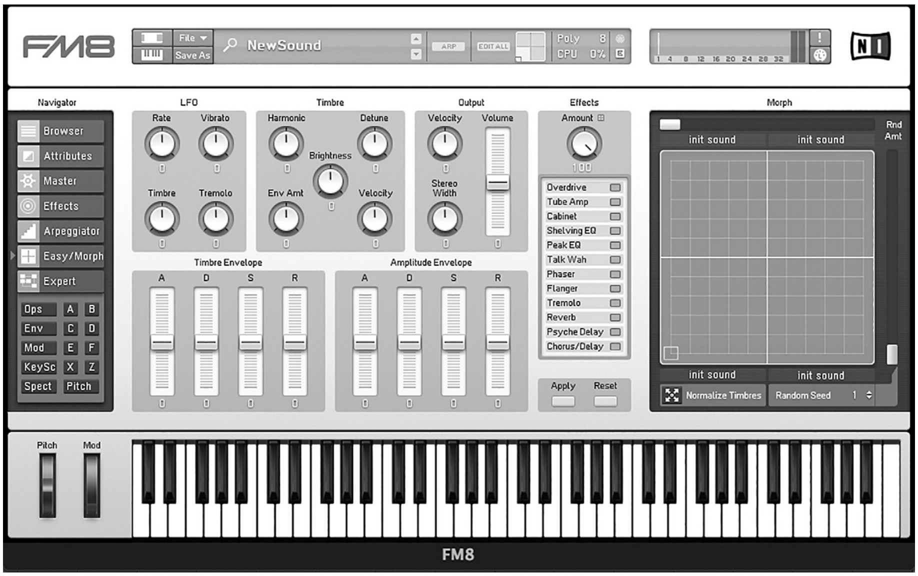

FIGURE 9.0
The basic modification parameters in FM8

If the filter sustain parameter is set to maximum, the filter will open to its cut-off value and remain there. This means that the filter's decay envelope will not affect the sound as there is nothing to decay towards. Alternatively, if the sustain is set to minimum, then the decay will act as the release parameter as there is no sustain to release from. Understanding the basic concepts of how envelopes will interact with one another is essential for sound design and programming.

During the modification of the filter envelope, it is often necessary to adjust both cut-off and resonance. This is because the current cut-off value will determine the filter envelope's value. In other words, the attack parameter of a filter's envelope will gradually increase from closed to the filter's current setting, but it is unable to exceed it. Therefore, if a low-pass filter were adjusted to 10 kHz (about halfway), the attack parameter would gradually open the filter from closed (0 Hz) and sweep up to the maximum of 10 kHz in a period determined by the attack parameter.

Many timbres will employ a low-pass filter with a 12 dB transition as this produces the most instantly musical results, but a 24 dB transition used on other tones can introduce contrast. An excellent example of this is to duplicate a channel and synthesizer and employ a 12 dB and 24 dB on the two consecutive channels. By doing so, the two transitions interact creating a more complex tone that warps and shifts in harmonic content.

Experimenting with other filter types such as bandpass and high-pass can create *thinner* timbres for layering. Employing a low-pass filter on a tone and a bandpass or high-pass filter on a secondary layered tone can result in many bright overtones. These overtones will mix with the original timbre to produce a sound rich in harmonic content.

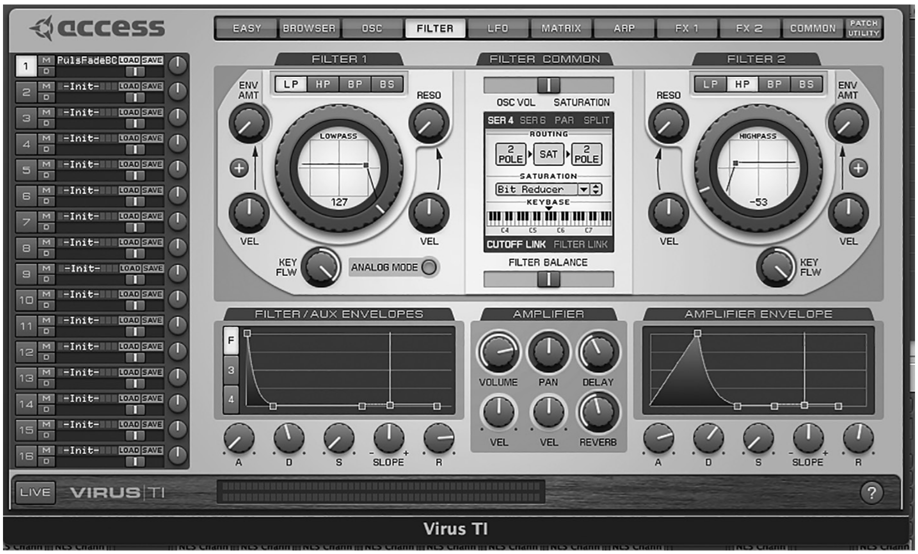

FIGURE 9.1
The filter and amplifier section from the Virus Ti

Further modifications can be obtained with filter key tracking. This makes it possible to change the way in which a filter behaves when playing higher or lower notes. Its purpose is well suited for electronic music sound design because it can be used to add subtle differences to the different pitches in a melody.

If filter key tracking is applied to a timbre by a positive amount, the filter will open further as the pitch increases. This is useful in the creation of moving chords (especially 1/5ths and 1/7ths) but also plays a role in the production of Dubstep and Complextro/Electro-style timbres.

As these genres rely heavily on the manipulation of pitch bend through the course of a note, if the pitch is increased, the filter also opens creating the atypical 'growl.' Similarly, employing filter key tracking on a low-pass filtered bass with a high resonance can produce a bass that brightens in texture as pitch increases. Alternatively, using negative tracking will produce bright bass notes at lower registers and darker tones at higher pitches.

MODULATION

If we are forced to pinpoint one defining element to any professional timbre, it must be its time-based development. Although the filter envelope can be employed to create harmonic movement throughout the timbre, its action is limited. Once the filter envelope is exhausted, there is no more movement to the sound and therefore it is the synthesizer's modulation possibilities that truly define its potential. All synthesizers feature a modulation matrix. The more complex this matrix is, the more interesting are the sounds that can be created.

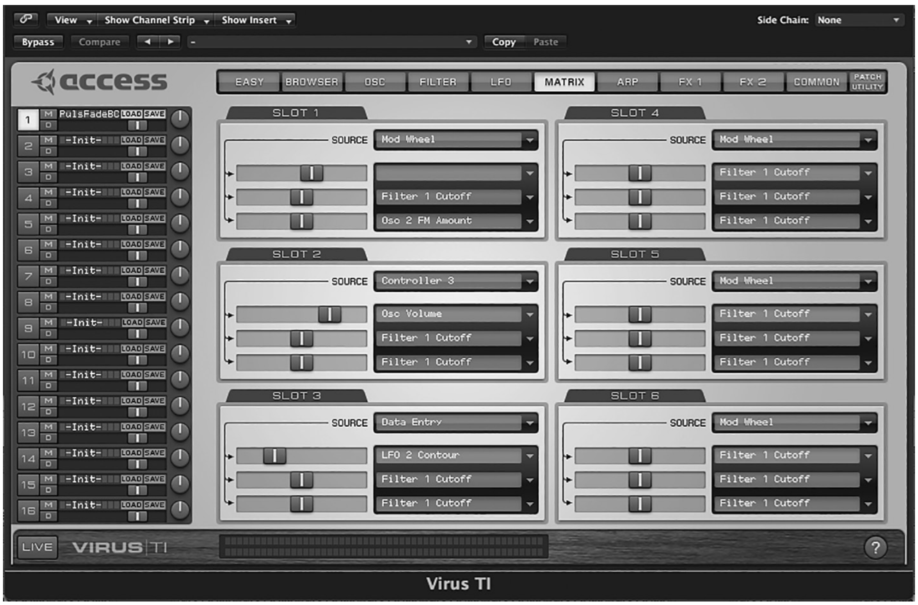

FIGURE 9.2
The modulation matrix from the Virus Ti

The matrix is similar to a patch bay and permits you to specify what sources will affect certain destinations. In many synthesizers, the most common modulation source is an LFO. The most common destinations for the LFO to modulate are oscillator frequency, filter cut-off, and resonance.

By employing a sawtooth LFO to modulate a filter's cut-off, the harmonic content will rise sharply but fall slowly. The LFO's rate will govern the speed at which this takes place. As the tempo of dance music is of paramount importance, it is common to sync this form of modulation to the DAW's tempo such that the tonal movement is synchronous with the tempo of the song.

This application is minimalistic, however, and, to program professional timbres, it is common to modulate multiple destinations from multiple sources. It is not possible to list all of the available modulation options available as they vary between models. Experimentation and an examination of your favorite patches will provide the real key to successful programming. But familiar sources and destinations are shown in Table 9.1.

In the end, no publication can ever replace your willingness to spend as much time as necessary experimenting with a synthesizer. You should also take time to examine any number of presets to gain an understanding of how the modulation affects the overall character of a timbre. This understanding is fundamental to sound design and like playing any instrument, without practice, you will never succeed at it.

Table 9.1

Source	Destination	Result
LFO	Filter frequency	Cyclic modification of the harmonics
LFO	Resonance	Cyclic modification of the harmonics
LFO	Oscillator pitch	Cyclic modification of the pitch
Envelope	Oscillator pitch	Introduces a pitch pluck at the beginning of a sound
Velocity	Filter frequency/ resonance/oscillator pitch	The harder a note is struck, the more the harmonics or pitch change
Mod wheel	Filter frequency/ resonance/oscillator pitch/oscillator mix	Movement of the mod wheel will introduce harmonic or pitch changes

APPROACHING SYNTHESIS

Provided you are familiar with the synthesizer, there are two ways synthesis can be approached. You can use careful analysis to reconstruct a sound envisaged for the music, or find a patch close to the initial idea and tweak the available parameters to sculpt it more favorably to your mix. Both of these options are viable options for sound design, but to take either approach, you must consider the variables.

First and foremost, unless you are 'scavenging sound' – i.e., you are looking to build a sound to inspire you creatively – it is common to have a melody or motif that requires a timbre. A common mistake is to adjust parameters while striking random notes on a controller keyboard to audition the changes. While this does provide some aural clues to the character of the timbre, it does not provide them with any *context* of music.

The envelope generators, LFOs, filters, and effects of a synthesizer will all heavily influence a motif, not just the timbre. A motif constructed of 1/16th notes would appear very different if the amplifiers release were altered. If it were shortened, the motif would become shorter and more defined, perhaps permitting longer delay effects or reverb settings. But, if the same parameter were lengthened, then the notes would flow together, and delay, or reverb, would only create problems within the mix.

Alternatively, lengthening the amplifier's attack parameter would severely affect the timing of the notes, moving them in – or out – of time with the rest of the music. Likewise, low-frequency oscillators may restart their cycle on patch auditioning, but may not do so through a motif as the note length changes. And if filter key follow is employed, the action of the filter will change according to the current pitch.

Whether modifying a preset or programming from a blank canvas it is crucial to take note of available frequency ranges and carefully consider the application of stereo and effects. Indeed, when it comes to introducing new tones into a mix, you must take the frequency range of all the other instruments into account.

Each sound in an arrangement will occupy a specific portion of the frequency range. If you were to program each of these individually without any regard to the other instruments that play alongside, you may end up with a host of harmonically complex timbres. These all sound great in isolation, but they will create a cluttered mess when they are playing alongside one another. So, whenever approaching synthesis, it is prudent to prioritize the instruments, giving those that are genre defining the most detail and room while simplifying any further parts.

For example, in a genre such as uplifting trance, the primary focus of the track is the lead instrument and its melody. When approaching this genre, the lead is treated as a priority instrument. This means that the lead is programmed to be as harmonically rich as necessary. But, the instruments that surround this such as chords or pads are designed to sit behind the lead. If they were equally as rich in harmonics, they would compete with the lead for frequency room within the mix.

Without taking this thoughtful frequency approach and instead believing it can be 'fixed in the mix' with EQ, you're only setting yourself up for failure. Mix EQ should be considered a chisel for fine-tuning and detailing, not a jackhammer for reorganization and repair.

Hearing, and *understanding*, the frequency range available to you is a result of ear training and experience. Both come with practice, but for beginners, it is wise to employ a spectral analyzer on each channel and also on the main mix bus. Many audio workstations provide spectral analyzers as a free plug-in, and using these, it is possible to visually see how much of the frequency range each instrument is consuming.

Many sounds consist of numerous overtones that do not contribute to the body of the timbre. This must be taken into account when viewing the frequencies in a spectral analyzer. For instance, a bass timbre and melody can easily consume frequencies from 40 Hz through to 12,000 Hz, but, in many cases, the character of the bass may only contribute a small percentage of this. Thus, provided the bass is not the driving element of the music, you could employ an EQ to shelve off everything below 80 Hz and everything above 1000 Hz.

Although this would result in the bass losing its character in isolation, when performed with all other instrumentation it will not be noticeable. Moreover, this 'progressive mixing' approach not only creates additional bandwidth for further instrument design, it also makes the final mix easier.

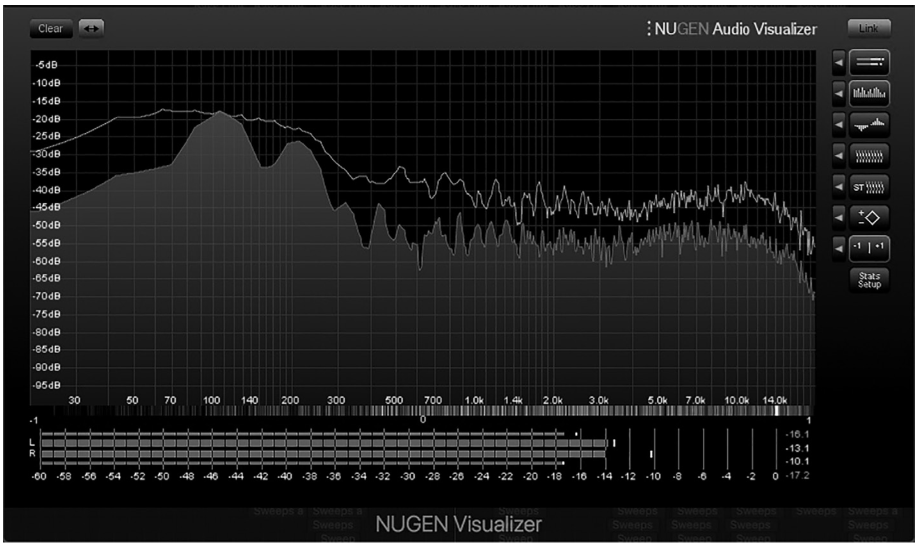

FIGURE 9.3
Spectrum analyzer on the mix bus

Similar consideration should be given to the application of any effects *during* programming. Almost all synthesizer patches are designed to sound their best in isolation, but when these are combined to the final mix, the effects tails, delays, and chorus all blend to produce a cluttered and indistinct result. As much of the sonic impact of electronic dance music is a result of contrast, you should avoid washing every timbre in numerous effects.

The defining sounds of any particular genre will benefit significantly from effects, and a lead instrument will have a more significant impact if many of the other instruments are 'dry' in comparison. If all the instruments are wet with effects, this impact is significantly lessened, and the contrast is lost.

MONO PATCHES

Many sounds should be in mono rather than stereo. Most plug-in and hardware instruments will exaggerate the stereo spread to make individual tones appear impressive. This effect is created through the use of layering two different waveforms or oscillators spread to the left and right speaker.

While this makes tones sound great in isolation, they will accumulate across the stereo field and reduce the available soundstage. Therefore, you should look towards programming sounds in mono unless it is a priority instrument that would benefit from being in stereo. Many instruments may not offer a mono compatibility and if this is the case, you can always mono the channel in the DAW's mixer.

Of particular note, while both kick and bass form an essential part of all dance music, the low energy produced by both should be mono and sit dead center so that both monitor speakers can share the energy. If the bass plays a more significant role in the music, then it often consists of two layered sounds. The first layer containing the low-end information remains in mono while the second layer – containing no low frequencies – fills the stereo image.

While customizing synthesizer presets to fit into a mix can be a rewarding experience, it is nonetheless a *basic* approach. And one that is only suitable provided you can find the right tone to start with. This often requires you to embark on a mind-numbing journey through countless presets, a search that will always tend to extinguish any creative impulses.

However, if a considerable amount of time is spent modifying and examining presets, modulations and effects, then the core of design from a blank canvas is little more than an understanding and experimentation of the oscillator section of the synthesizer. The oscillators and the various augmentations available in the mix section were discussed in detail in an earlier chapter, and, once this basic knowledge is affirmed, the rest is down to listening and experimenting with the available oscillators. I cannot stress enough that, to be competent with any synthesizer, it is vital to listen to each oscillator and the results of experimentation from mixing and detuning oscillators, along with the application of oscillator sync, frequency modulation, and ring modulation.

CHAPTER 10

Samplers

I think it's innovative and creative to stay away from flat-out sampling somebody else's record. To me, that doesn't show your creative side unless you take a little piece and add to it, almost like spice on a chicken.

Missy Elliot

The introduction of the sampler in the 1980s created one of the most significant technological and creative shifts in the history of music. It gave artists the opportunity to record, edit, and playback audio in ways never before seen outside a professional studio and is ultimately responsible for the emergence of new genres such as Drum & Bass and Jungle.

Since then, the sampler has been overshadowed by the DAW. It is now possible to record, cut, slice, move, stretch, time-compress, and edit audio directly on the computer, so what could once only be performed with a sampler has become commonplace. But while the DAW has overshadowed hardware sampling, sample playback plug-ins remain just as vital for the production of music.

These are not samplers in the strictest sense of the word because they're not capable of recording; they're only capable of playback. Nevertheless, there are countless multisampled instruments on the market for them to use. These can range from samples of classic electronic instruments through to libraries consisting of entire orchestras with each instrument sampled multiple times for expression. Plus, with the current trend of sampling vocals or synths and playing them out of range (Major Lazer 'Lean On' being the most prolific use) an understanding of samplers and some of their creative uses is as useful today as it's always been.

The original hardware samplers are the digital equivalent of an analog tape recorder. Rather than recording the audio signal onto a magnetic tape, sounds were recorded digitally into **R**andom **A**ccess **M**emory (RAM). After the audio signal had been recorded, it can be manipulated with a series of editing parameters.

FIGURE 10.0
Native Instruments' Kontakt, a popular third-party software 'sampler'

This is where the **S**ample **P**layback **S**oftware (SPS) has taken over. Prerecorded audio is imported and manipulated with parameters similar to synthesizers. This includes amplitude and filter envelopes, LFOs, and various other modulation options. Also, you can play a single pitch sample across the sampler at different pitches. For this, the sampler artificially increases or decreases the original sample frequency with the note you're playing.

The downside to this approach is that if the sampling frequency is increased or decreased by a significant amount, it no longer sounds anything like the source. If you import a piano sample at the note C3 and play this at the same pitch from the sampler, the note will play the note as you expect. If you play this same sample at C4, however, the sampler must increase the frequency of the original sound by 12 semitones (from 130.81 Hz to 523.25 Hz), and it will sound nothing like a piano.

Even though this may seem to limit a sampler's practicality, this pitch degradation has become responsible for many classic sounds in dance music. A favorite technique among House artists is to sample an instrument playing a chord. When played at the original note through the sampler, the chord will repeat as you would expect but as you change the pitch, the chord changes in strange yet exciting ways. This is known as *chord-planing* and features on countless dance records.

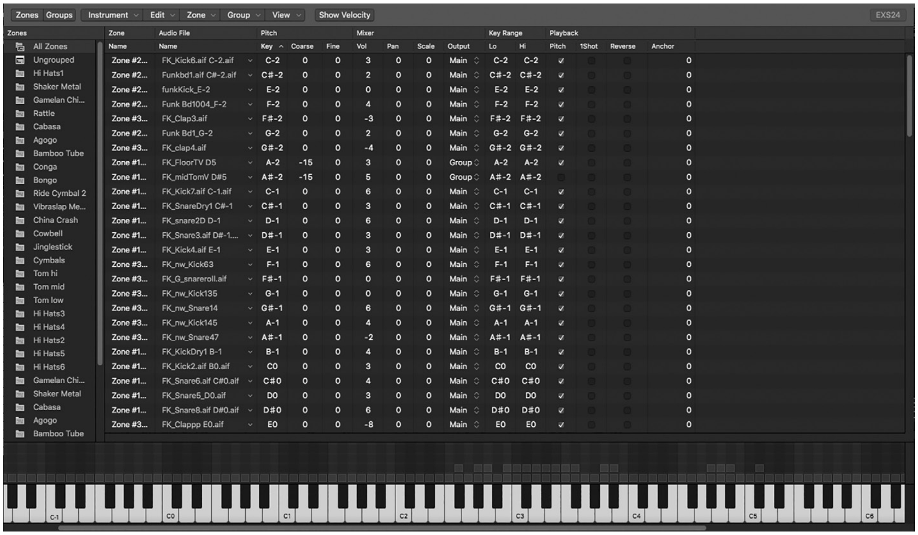

FIGURE 10.1
A multisampled instrument in Logics EXS

If we want to recreate a real instrument within a sampler, then we have no choice but to record an instrument at every few intervals. Most samplers can only reproduce an acceptable instrument three, four, or five notes from the original root note so if a replica of an instrument is required; it must be sampled at every few intervals. For example, a piano could be sampled at C0, E0, G0, and B0, then C1, E1, G1, B1, and so forth until the entire range of the piano is recorded. This technique is known as 'multisampling.'

With the relatively low price of both RAM and HDD, almost all instruments today are sampled at every single note of the instrument, each at various velocities. And if it's a real instrument, it will often be recorded with multiple microphone positions too. While this approach produces incredibly realistic reproductions of the original instrument, it also requires a significant amount of RAM and a large (and often dedicated) HDD.

HARDWARE SAMPLERS

While SPS has become the primary format for those who need to reproduce real instruments, for everything else, dance music artists will often look towards the older hardware samplers. This is because many of the older hardware units impart a specific sonic character that is highly sought after. Part of this appeal is due to the limited bit rates of the era, since many could only record in 8-bit or 12-bit. While some suggest you can achieve the same result using a bit-rate reducer in a DAW, it discounts the sampler's converters, asynchronous clocking on playback, and the high-frequency distortion from the anti-aliasing filters. Consequently, samplers such as the Akai S950 and the Ensoniq ASR-10 are in high demand and the prices on the secondhand market have sky rocketed.

> DMM resource: The resource files contain a sample of a drum loop. The first is the drum loop in the DAW with 12-bit reduction applied. The second is the same drum loop run through my Akai S950. The difference between the two is apparent.

Using these older samplers, memory is still a concern. Because they hold sounds in their onboard RAM, the maximum sampling time is limited by the amount of available memory. At full audio bandwidth, one minute of mono recording will use approximately 5 megabytes (MB) of RAM. If you're sampling a keyboard instrument in mono, this could equate to 80Mb of memory, and you need to double this if you want it in stereo. Consequently, various techniques have been adopted to make the most of the available memory.

The first is to loop the sample. As the overall length of a sample determines the amount of RAM that is required, reducing the sample's length means more samples will fit into the available memory. As many sounds have a unique attack and decay period but the sustain element remains relatively consistent, the sustain portion can be continually looped for as long as the key is held. Once it is released, the sample can then play the release portion. This means that only a short burst of the sustain period must be sampled helping to conserve memory.

This requires skill and patience, however, because the sustain period of any sound is rarely static. Many will exhibit slight tQMural changes in the harmonic structure. Therefore, if too small a portion of this tQMural development is looped, the results sound unnatural. Conversely, if too long a piece is looped, the decay or some of the release period may also be captured, and this would seem strange too. Also, any loop points must start and end at the same phase and level during the waveform's cycle. If not, the difference in phase or gain could result in an audible click as the waveform reaches the end of its looped section and returns to the beginning.

Some samplers work around this by automatically locating the nearest zero crossing to the position you choose. While this increases the likelihood that you can achieve a smoother crossover, if the waveform's level is different at the two loop points there will still be a glitch. But these glitches can often be avoided with crossfading.

With a crossfade, it is possible to fade out the end of the looped section and overlap it with a fade in at the start of the loop. This results in a smooth crossover between the two looping points, reducing the possibility of glitches. Although this goes some way to resolve the problems, it is not always the ideal solution. If the start and end of the looped points are at different frequencies, there will be an apparent change in the overall timbre during the crossfade. Unfortunately, there is no quick fix for avoiding these pitfalls and success can only be accredited to patience, experimentation and experience.

SAMPLE CDS

With the development and growth of the internet, the term sample CD has become a misnomer. During the reign of hardware samplers, sample content was delivered on a compact disk that could be installed into a hardware sampler. Today, all sample content is provided via the internet digitally to the purchaser, so there are no CDs involved. A better description would be *sample content*.

The majority of sample content today is delivered in WAV and AIFF. Audio workstations and many hardware platforms recognize both these, and both can be 'dragged and dropped' directly into the workstation. Or they can be dragged into a compatible software sampler where they can be key mapped and edited further.

If the sample consists of only one note, and you want to play it across the keyboard, it will need to be key ranged. And, if you're going to sustain the note for longer than its natural duration, it will also need to be crossfaded. This can be time consuming, taking a good hour or so depending on the sampler's interface.

Because of this, some content that consists of multisampled instruments may be supplied in a format suitable for a specific sampler. For this, the audio samples are stored alongside a particular format of data that contains information about sustain loops, key, and velocity ranges. With this, you only need to copy the folder to the computer and then use the relevant sampler to open the file. The sampler will then load and key map everything automatically. Many of these formats are proprietary, however, and will only open in the correct model of sampler.

FIGURE 10.2
Logic Pro's EXtreme sampler

Many SPS are also compatible with older hardware sampler standards such as AKAI and E-MU. These were the most popular formats available during the 20-year reign of hardware sampling. Consequently, there is a vast library of AKAI and E-MU samples that can be purchased for very little on auction sites such as eBay. Note, however, that importing an AKAI format CD into a software sampler may not always perform as expected and some software samplers will interpret the data differently. This often results in differences in the timbre and tQMure of the sample but in more severe cases the key mapping may not come out quite as expected, the sustain samples may not loop correctly, or the sampler may just crash altogether ...

If the sample content consists of loops rather than multisampled instruments, then the formats may comprise ACID, APPLE, WAV, and REX. Both ACID and APPLE loops are audio files that contain additional tempo information so that when inserted into specific workstations, the loops will time stretch or compress to suit the current working tempo. Moreover, if the speed of the project is adjusted, the samples will change to suit the new tempo.

A similar principle applies with REX and REX2 files. This is a proprietary format from Propellerhead software. Although this format is no longer as popular, it does still feature on a lot of sample content. With this format, musical loops (most typically drum loops) have been analyzed for their transients and treated to any number of 'slices' to cut the loop into individual rhythmic components. By doing so, the tempo or pitch of each can be freely adjusted without affecting the timbre of the content.

FIGURE 10.3
The author's AKAI S5000

Many DAWs feature this style of automated transient detection and loop slicing as a standard, although different DAWs will use different terminology. Apple's Logic Pro refers to this as *flex time editing* while Cubase terms it *vari-audio* or *audio warp*.

BASIC SAMPLING PRACTICES

Many of the standard sample contents available consist of loops or single hits. These can be used 'as is' but usually you will want to map the sounds across a sampler for easier access. This process requires some knowledge of key mapping.

For mapping, the 'root' key must first be configured in the sampler. This is the key that the sampler will use as a reference point for pitching the following notes up or down along the length of the keyboard. For instance, bass samples are typically set up with the root key at C1 or C0. If this note is played, the sample will play back at its original frequency. The sample is then stretched across the keyboard's range as much as required. In most software or hardware samplers, the root key is set in the key range or key zone page along with the lowest and highest possible notes that will be available for that particular sample.

With this information, the sampler spreads the root note across the defined range, and the bass can be played within the specified range of notes. To gain the best results from the subsequent pitch adjustments on each key, it is preferable that the sampler's key range is set to three or four intervals above and below the root note. Further than this and the sound may degrade, but this isn't necessarily such a bad thing with dance music. Some classic tracks have used wildly out of range samples as the bass or lead.

When setting a key zone for bass sounds, it may be possible to set the range much lower as the fundamental frequency determines the pitch and the lower the pitch, the more difficult it is to perceive pitch. Thus, for bass samples, it may be possible to set the lowest key of the key zone to 12 semitones (an octave) below the root without introducing any unwanted artifacts. In fact, pretty much anything sounds okay if it's pitched down low enough.

Most competent samplers allow some key zones to be set across the keyboard. This makes it possible to have, a bass sample occupying F0 to F1 and a lead sample occupying F2 to F3. If organized in this way, both the bass and lead can be played simultaneously from an attached controller keyboard. In taking this approach, you do need to confirm that each key zone can access different parameters of the sampler's modulation engine. If this is not possible, settings applied to the bass will also apply to the lead.

After the samples are arranged into key zones, you can configure how the sampler will behave with note velocity. This is useful for recreating the movement within sounds. It can add expression to melodies and prevents the static feel that is often experienced with a series of sampled sounds. A standard application for this is to map velocity to low-pass filter so that the harder the key is depressed, the more the filter opens.

'Velocity crossfading' and 'switching,' if available, is also worth experimenting with. Switching employs velocity values to determine which sound should play. The two samples are imported into the same key range and striking the key harder (or more softly) results in switching between the two different samples. Velocity crossfading employs this same principle but morphs the two samples together creating a (hopefully) seamless blend rather than an immediate switch between one and the other.

SAMPLE LOOPS

Alongside single synthesizer hits, the majority of sample content features a somewhat ubiquitous number of preprogrammed or prerecorded drum loops to suit specific genres. These can be dragged directly into the DAW's arrange page, then sliced, diced, rearranged, and played alongside the music. However, a more creative avenue is available if these loops are imported into a sampler and either triggered or converted via MIDI.

Many samplers will measure the transients in a loop and place slices at each transient. These individual slices are mapped across the note range so that each note plays one of the transient samples from the loop. This approach lets you completely redesign the drum loop by moving notes around in the piano roll editor and is a favorite technique in dance music.

If you choose to import a loop into a sampler, the loop will start from the beginning every time a key is depressed. If a key on a controller keyboard is tapped repeatedly the loop will continually restart, producing a stuttering effect. This technique can be used to significant impact to create breakdowns in dance music.

Alternatively, the sampler can be set to 'one-shot trigger mode,' so that a quick tap on a controller keyboard plays the sample in its entirety even if the key is released before the sample has finished playback. This is useful if you want the original sample to play through to its usual conclusion while triggering the same sample again to play over the top.

TIME COMPRESSION

So far, we've assumed that a loop or musical phrase that has been sampled is at the same tempo as the rest of the mix, but this is rarely the case. Accordingly, all DAWs, samplers, and step sequencers will provide some form of time-stretching functions so the tempo of the phrase or loop can be adjusted to fit with the music.

This can be useful, but the quality of the results is proportionate to how far they are pushed. While it should be possible to adjust the tempo of a loop by 25BPM without introducing any digital artifacts, adjustments above this may introduce digital noise or frequency artifacts that compromise the audio.

This can be a desired side effect, such as in the production of Drum & Bass but, in most cases, you'll want to avoid it. Therefore, it is advisable to place the loop directly into the DAW and employ its time-variant slicing features to slice the loop into constituent parts. Once sliced, the loop can be adjusted in tempo, exported from the workstation, and re-imported back into the sampler.

This form of time-variant beat slicing is only useful on transient material such as drum loop since the automated process scans the audio for transients. If the audio is a vocal loop or similar, it is less likely to feature transients, and therefore the only option is to cut the audio into separate words physically, and sometimes even syllables, and then time-stretch each to complete the loop.

Of course, not all loops are from sample content, and they may be taken from other records. For obvious reasons, I cannot condone stealing samples from another artist, but it would be naive to suggest that it doesn't occur. The Hip-Hop movement and many House records have their foundations deeply ingrained in this form of sampling.

Although there is plenty of sample content dedicated to both these genres, they are avoided because everyone else has access to the same. Instead, producers search through the more obscure or most unlikely resources, a technique known as *crate digging*. For example, Roger Sanchez's massive house anthem 'Another Chance' was formed entirely around a sample from the beginning of Toto's 'I Won't Hold You Back' and Eric Prydz 'Call on Me' started with a sample Steve Winwood's 'Valerie.'*

* 'Valerie' was only sampled in the first version. On hearing the track, Steve Winwood rerecorded the vocals for the record.

With many dance records, the more obscure the original record is the better, as it's unlikely any other producers will have access to the same records. Plus, if required, you'll be able to acquire copyright permission easily. These records are sourced in the majority of secondhand and charity shops.

Beyond sampling other records, it can be equally advantageous to sample some of your own synthesizer's. For example, sampling a mono synthesizer and layering that sample across some keys in the sampler can be used to create a unison effect. This is a favorite technique among those using modular synthesis since these are mono sources. Alternatively, if employing a hardware plug-in instrument such as the Virus Ti, the units DSP can be saved for other tasks by multisampling the instrument and applying it in a software sampler.

In fact, multisampling the synthesizer at every key would recreate the synthesizer in the sampler. And in some instances, a sampler may offer more parameters than are available on the synthesizer enabling you to, say, synchronize the LFO to the tempo, use various LFOs, or access numerous different filter types. Sampling from films can also open many creative avenues. Christopher Bertke, aka *Pogo*, has made a career using nothing more than small samples taken from movies and sequencing them together. His 'Upular' remix consists of samples taken only from Disney/Pixar's *UP* and is currently reaching towards 10 million views on YouTube.

Alternatively, with some creative thought, a recording of a ping pong ball being thrown at a garage door can become a tom drum, hitting a rolled-up newspaper against a wall can make an excellent house drum kick, and the hissing from a compressed air freshener can produce great hi-hats.

While many musicians rely on the sampler to record a drum loop (which can be accomplished with any DAW), they are not using them to their full QMent and experimentation is the real magic to any sampler. Therefore, what follows are some general ideas to get you started in experimental sampling.

Creative sampling

Although cheap microphones are worth avoiding if you plan to record vocals, they do have other creative uses. Connecting one to an audio interface (or step sequencer) and striking the top of the mic with a newspaper of your hand can be used as the starting point of kick drums or scratching the head of the microphone can be used as the starting point to Guiros.

You should also listen out for sounds in the real world to sample and contort. Todd Terry acquired snare samples by bouncing a golf ball off a wall, and Mark Moore used an aerosol as an open hi-hat sound. Many Techno and Tech House artists regularly carry a portable digital recorder around with them to record natural ambiances to use in their music.

Hitting a plastic bin with a wet newspaper can be used as a full slurring kick drum and scraping a key down the strings of a piano or guitar can be used as the basis for whirring string effects (it worked for Dr. Who's TARDIS anyway). Once sampled these sounds can be pitched up or down, or affected as you see fit. Even subtle pitch changes can produce useful results. For example, pitching up a sampled snare by just a few semitones results in the snare used on Drum & Bass, while pitching it down gives you the snare typical of Lo-Fi.

Sample reversing

This is probably the most immediate effect to try but, while it is simple to implement, it can produce excellent results. The most straightforward use is to sample a cymbal hit and reverse it in the sampler. If played at a much lower frequency, it can be used as an uplift sound effect in an arrangement, while if performed higher it can be employed as an incidental sound effect. More creative options appear when you consider that reversing any sound with a fast attack but long decay or release will create a sound with long attack and an immediate release.

For example, a guitar pluck can be reversed and mixed with a lead that isn't. The two attack stages will meet if they're placed together, and these can be crossfaded together. Alternatively, the attack of the guitar could be removed so that the timbre begins with a reverse guitar that moves into the acute attack phase of a lead sound or pad. Taking another tack, if a tone is recorded in stereo, the left channel could be reversed while the right channel remains as it is to produce a mix of the two timbres. You could then sum these to mono and pitch them up and down the key range.

Pitch shifting

A popular method used by dance producers is to pitch shift a vocal phrase, pad or lead sound by 1/5th as this can often create impressive harmonies. If pads or leads are shifted further than this and mixed in with the original, it can introduce a pleasant phasing effect.

Perceptual encoding

Any perceptual encoding devices (such as mini-disk) can be used to compress and mangle loops further. Fundamentally, these work by analyzing the incoming data and removing anything that the device deems irrelevant. In other words, data representing sounds that are considered to be inaudible in the presence of the other elements are, and this can sometimes be beneficial on loops.

Physical flanging

Most flanger effects are designed to have a low noise floor that isn't much use if you need a dirty flanging effect for use in some genres of dance. This can, however, be created if you own an old analog cassette recorder. If you record the sound to cassette and applying a small amount of pressure on the drive spool the sound will begin to flange in a dirty uncontrollable manner, this can then be rerecorded into the workstation and sampler to be played at different pitches.

Transient slicing

Although also possible within a DAW's arrange page, a favorite technique for House tracks is to design a pad with a slow resonant filter sweep and then sample the results. Once in a sampler, the slow attack phase of the pad is cut off so that the sound begins suddenly and sharply. As the transient is the most crucial part of the timbre, this creates an interesting side effect that can be particularly striking when placed in a mix.

Creative use of samplers comes from a mix of experience, experimentation, and serendipity. What's more, all of the preceding examples are just that – examples to start you on the path towards creative sampling and you should be willing to push the envelope (no pun intended) much further. The more time you set aside to experiment, the more you'll learn and the better results you'll achieve.

SAMPLES AND CLEARANCE

Ever since the introduction of sampling, single hits, melodies, vocal hooks, drum loops, basses, and even entire sections of music from countless records have been sampled, manipulated, and otherwise mangled in the name of art. Hits by James Brown, Chicago, Donna Summer, Chic, Chaka Khan, Sylvester, Lolita Holloway, Locksmith, and Toto, among innumerable others have come under the dance musicians' sample knife and been remodeled for the dance floor.

Although in the earlier years of dance artists managed to get away with releasing records without clearing the samples, this was because it was a new trend and neither the original artists nor record companies were aware of a way that they could prevent it. This changed in the early 1990s when the sampling of original records proved it was more than just a fad. Today, companies know exactly what to do if they hear that one of their artists' hits has been sampled and in most cases, they come down particularly hard on those responsible. So, if you plan to release a record that contains a sample of another artist's motif, drum, vocals, or entire verse/chorus without first applying for clearance, you may well end up with a lawsuit.

To help clear up some of the myths that still circulate – I spoke to John Mitchell, a music solicitor who has successfully cleared samples for many other artists and me.

What is copyright?

Copyright exists in numerous forms, and if I were to explain the entire copyright law, it would probably use up most of your publication. To summarize, it belongs to the creator and is protected from the moment of conception, exists for the lifetime of the creator and another 70 years after their death. Once out of copyright the work becomes 'public domain,' and any new versions of that work can be copyrighted again.

So it would be okay to sample Beethoven or another Classical composer?

In theory, yes, but most probably not. The question arises as to where did you sample the recording from because it certainly wouldn't have been from the original composer. Although copyright belongs to the creator, the performance is also protected via copyright. When a company releases a compilation of Classical records, they will have been rerecorded, and the company will own the copyright to that particular performance. If you sample it, you're breaching the performance copyright.

What if you've transcribed it and recorded your own performance?

This is a legal dispute that is far too complex to discuss here since it depends on the original composer and the piece of music. While the original copyright may have expired, a company may have purchased the copyright and own the transcript. My advice is to check first before you attempt to sample or transcribe anything.

So what can you get away with when sampling copyright music or speech?

Nothing. There was a rumor circulating that if the sample is less than 30 seconds in length, you don't need any clearance, but this isn't true. If the sample is only a second in length and its copyright protected you are in breach if you commercially release your music containing it.

What if the sample were heavily disguised with effects and EQ?

You do stand a better chance of getting away with it, but that's not to say you will. Many artists spend a surmountable amount of time creating their sounds and melodies, and it isn't too difficult to spot them if they're used on other tracks. It isn't worth the risk!

Does this same copyright law apply to sampling snippets of vocals from TV, radio, DVD, etc.?

Yes, sampling vocal snippets or music from films breaches many copyrights, including the script writer (the creator) and the actor who performed the words (the performer).

What action can be taken if clearance hasn't been obtained?

It all depends on the artist and the record company who own the recording. If the record is selling well, then they may approach you and work a deal, but be forewarned that the record company and artist have the upper hand when negotiating monies. Alternatively, they can take out an injunction to have all copies of the record destroyed. They also have the right to sue the samplist.

Does it make any difference how many pressings are being produced for retail with the uncleared samples on them?

No, if just one record were released commercially containing an illegal sample the original artist has the right to prevent any more pressings being released and also has the rights to sue.

What if the record were released to DJs only?

It makes no difference. The recording is aired to a public audience.

How much does it cost to get sample clearance?

It depends on many factors. How well known the original artist is, the sampled record's previous chart history, how much of the sample you've used on your track and how many pressings you're planning to make. Some artists are all too willing to give sample clearance because it earns them royalties while other will request ridiculous sums of money.

How do you clear a sample?

Due to the popularity of sampling from other records, companies are appearing every week who will work on your behalf to clear a sample. However, my advice would be to talk to the Mechanical Copyright Protection Society. These guys handle the licensing for the recording of musical works and started a Sample Clearance Department in 1994. Although they will not personally handle the clearance itself – for that you need someone like myself – they can certainly put you in touch with the artist or record company involved.

CHAPTER 11

Compressors

We have become the tools of our tools.

Henry David Thoreau

Compressors are one of the most abused processors in the dance musician's toolbox. Originally designed to control peaks in audio, their misuse now defines *part* of the sound of electronic dance music.

Compressors offer a way to control musical dynamics. This is the ratio between loud and soft sections in the same piece of music. If you're recording a vocal track that requires the talent to whisper in the verse but shout in the chorus, this sudden shift in dynamics makes it difficult to select a single acceptable recording level.

If you set the recording levels to capture the quiet sections when the performance increases in volume, it will result in distortion. However, if the recording levels are set only to capture the loud sections, then when the talent whispers there will be a low dynamic ratio between the vocals and the noise generated by the equipment. This means the noise could be evident and ruin the quality of the recording.

Before compressors, the solution was to *ride the gain*. You would sit by the gain control on the recording device and adjust it whenever the source volume changed. This is how the early radio engineers prevented clipping the broadcast antennas.

Gain riding isn't the perfect solution, however, because it's a sheer pain to do. Nobody wants to spend their time hovering over a recording device on constant alert. In addition, you also need some warning of upcoming changes in gain. Either that or you need lightning fast reflexes, which are not something engineers are well known for.

Jim Lawrence, a radio engineer, presented the answer. He came up with a solution called a 'leveling amplifier.' The **L**evelling **A**mplifier **1** (LA1) could automatically control the dynamics of a performance by employing a photoresistor that monitored the incoming signal. As the signal level increases, the photoresistor grows brighter. This produces an increased impedance that reduces the gain of the signal at the output. As the input level decreases, the photoresistor dims, and the impedance lowers, resulting in an increase of gain level at the output.

Jim Lawrence went on to form the *Teletronix* Company to mass produce the now highly coveted LA-2A leveling amplifier that – internally anyway – works on the same principle as the LA1.

By inserting a leveling amplifier (compressor) between the signal source and the recording device, when the gain exceeds a certain threshold, it is reduced. This prevents clipping of the signal.

But controlling dynamics in this way presents further opportunities. For example, by managing the dynamics in a recording, it makes mixing easier. If the amplitude of a recording is inconsistent, the fluctuations in volume result in the quieter sections disappearing behind other instruments. But, if you increase the volume of the soft parts, when the recording becomes louder it could become too prominent in the mix. While this could be dealt with today with the application of *mix automation*, a more straightforward solution is to just even out the dynamics.

By compressing the signal, any part of the signal that exceeds the user-defined threshold is reduced in gain. However, elements that do not exceed the threshold remain untouched. This means that the dynamic range between the loudest and quietest sections is reduced, and audio maintains a more even volume throughout. This form of dynamic restriction is shown in Figure 11.1.

FIGURE 11.0
The first commercial studio compressor – the Teletronix LA-2A leveling amplifier

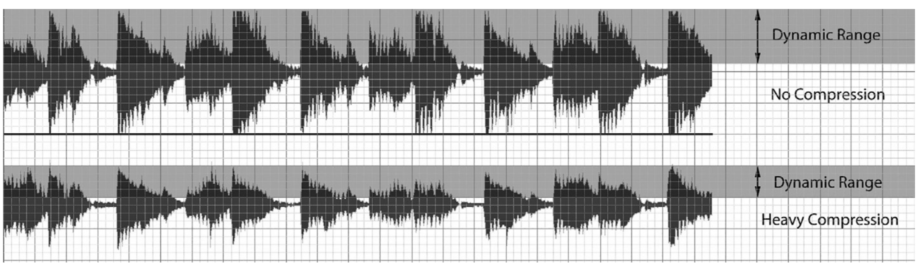

FIGURE 11.1
Before and after compression

The effect of this compression is twofold. It will level out any volume fluctuations in the signal to create an average signal level. This allows you to further increase the amplitude without fear of spurious audio spikes clipping and distorting. But, also, when the dynamic range of a sound is reduced, we perceive it to be louder. Human hearing has developed so that sustained sounds appear louder than transients, even if the transient happens to be at much higher amplitude than the continuous tone. This is why a TV program used to seem quieter than the advertisements that follow it. The commercials had a more restricted dynamic range than the TV program so we would perceive them as louder.

In addition to leveling dynamics of an entire recording, a compressor can massage the overall dynamic character and response of sound. Alongside making audio appear louder, it can also saturate *tones*, so they exhibit power and attitude. Of particular note, it was the use of a compressor's side-chain that defined the sound we associate with dance music today, i.e., a kick that appears to punch a hole in the rest of the mix every time it strikes.

Before we go further, though, we should first have an understanding of compression. It is essential to understand how each parameter on the compressor can affect a sounds dynamic envelope. To do this, we can examine how a compressor behaves on an audio signal.

Figure 11.2 shows a block diagram of the volume of an audio signal. It runs at a constant gain before increasing in gain and then decreasing again. In this example, the sine wave has theoretically been recorded on a 16-bit device and therefore offers a –96 dB dynamic range. We cannot increase the gain of the loudest part of the sine wave any further because, if we do, it will increase beyond its dynamic range and distort.

Let us compress it. The first parameter on a compressor is the threshold. This determines whereabouts in the signal's amplitude range the compressor will act. If the sine wave breaches the threshold, the compressor will introduce gain reduction. The threshold parameter is calibrated in decibels and is adjusted so that the average signal level lies *just below* the threshold. By doing so, if there are any sudden increases in gain, the signal exceeds the threshold and gain reduction takes place. For this particular example, we'll set the threshold to –32 dB. For ease of understanding, it is best to consider this as creating two independent dynamic ranges in the audio. There is the dynamic range of the compressor that is –32 dB and the dynamic range leftover of –64 dB (32 dB + 64 dB = 96 dB). This behavior is shown in Figure 11.3.

FIGURE 11.2
A block diagram of an audio signal

FIGURE 11.3
Application of compressor's threshold

FIGURE 11.4
8:1 ratio compression

When the sine wave exceeds the threshold, the compressor jumps into action. How much compression is applied to the signal is determined by the ratio parameter. The ratio is the difference in signal level between the audio exceeding the threshold and the signal level that leaves the compressor. If the ratio parameter is set to 4:1, for every 4 dB that the signal exceeds the threshold, it is reduced to only 1 dB at the output of the compressor.

To put this into practice in the current example: if you were to set an unusually high ratio of 8:1, for every 8 dBs the sine wave exceeds the threshold, the compressor reduces it to just 1 dB. As the sine wave breaches the threshold of the compressor by 32 dB, the compressor will reduce every 8 dBs in this compressors dynamic range to only 1dB.

It is possible to calculate the amount of gain reduction applied with simple math. We only need to divide eight into 32 dB. As there are 4 x 8s in 32, we could speculate that the compressor will 'squash' the 32 dB of the sine wave that exceeds the threshold into just 4 dB. As Figure 11.4 shows, this modifies the dynamic envelope beyond its original shape.

The result from this form of severe compression can be either beneficial or detrimental. Here the sine wave signal that exceeded the threshold has been compressed, and its original peak has been massively reduced in gain. This reduction in the overall dynamic range brings it closer in amplitude to the rest of the sine wave. Consequently, the total gain of this wave could now be increased further. This excessive application can be employed for sound design. For example, the kick in dance music has grown to be one of the dominant elements in the mix and requires plenty of body to create the punch. If a kick is lacking in body, by applying this style of compression, the dynamic range between the transient and body is reduced. This increases gain in the kick's body resulting in a thicker sounding kick drum.

Similarly, using the same form of heavy compression on a snare drum produces a stronger, forward snare. The transient of the snare exceeds the threshold resulting in the compressor clamping down on it. This reduces the dynamic range between the transient and tail and results in a snare typical of some styles, e.g., House, Techno, and Tech House.

While extreme compression offers some sound design benefits, it must be used with caution. An excessive gain reduction will reduce high-frequency content while also severely modifying the dynamic envelope. This can be most noticeable on real-world instruments and vocals because we instinctively know how these sound. Being overzealous with compression on these produces strange results that are not pleasant to the ear.

ATTACK TIME

To make dynamic restriction less noticeable to the listener, compressors feature an attack parameter. This parameter is configured in milliseconds determines how slowly the gain reduction is applied once the threshold is breached.

This parameter is often mistaken for controlling how long the compressor *waits* before compression is applied, but *compression will* ***always*** *begin as soon as the threshold is breached*. Instead, the attack parameter indicates the time it will take for the compressor to reach its maximum gain reduction.

The attack can seriously affect the tonal content and character. Assuming the threshold remains at –32 dB and the ratio set to 8:1, if the attack parameter is set to 100ms, the compression will slowly increase gain reduction over a period of 100ms. This results in the compression bypassing the transient and compressing only the tail.

As shown in Figure 11.5, the attack can seriously transform dynamic envelopes, and it does not permit you to transparently apply heavy compression. While it will allow the initial transient through unmolested, maintaining some transparency and high-frequency content, the constant increase towards a heavy gain reduction results in a redesign of the dynamic envelope.

If you require more transparent results, a higher threshold and lower ratio can be applied. Or since it is the relationship between threshold and ratio that defines the amount of gain reduction, a lower threshold and higher ratio. Both should be combined with a fast to medium attack time. This permits the transient to pass through the compressor mostly unmolested, so the high-frequency content remains intact, after which, gain reduction would increase to bring the remainder of the signal under control.

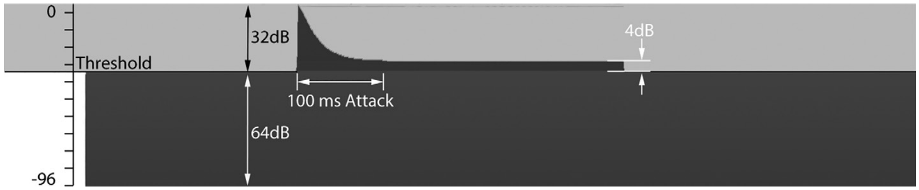

FIGURE 11.5
The effects of attack parameter

KNEE

If we require high gain reduction but want to maintain some transparency, we can use a soft knee. Many compressors permit you to alternate between soft and hard knee operation.

The knee defines how the compressor reacts as the signal approaches the threshold setting. On a hard knee, as soon as the threshold is breached, gain reduction is applied. In a soft knee application, the compressor monitors the signal and gain reduction is implemented gradually as the signal approaches the threshold.

With a soft knee, gain reduction begins when the signal is within 3 to 14 dB of the current threshold (compressor dependent). As the signal grows ever closer to the threshold, gain reduction is gradually increased until the threshold is exceeded whereby full gain reduction is applied.

Soft knees make the compressor's action less evident and therefore are suitable for use on real instruments and vocals where you may need to apply substantial gain reduction but want the processing to remain mostly transparent. The action of different knee settings is shown in Figure 11.6.

Compressors may also offer the option to switch between RMS (**R**oot **M**ean **S**quare) and peak mode. Since any compressor with a hard knee will control transients as they breach the threshold, it is still possible for short transient sounds such as hi-hats, kick drums, snares, and toms to breach the threshold unnoticed. This is because the transient is so short that by the time the breach is detected, it has already been missed.

To avoid this, you can employ the peak mode detection on particularly transient sounds. With this engaged the compressor becomes sensitive to short peaks and will apply gain reduction the minute signals draw close to the threshold rather than wait for a threshold breach. By doing so, any transient peaks can be controlled before they overshoot the threshold.

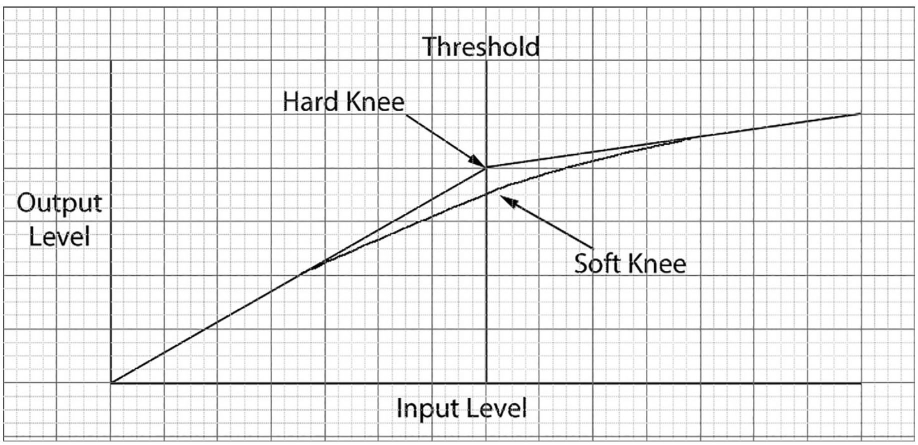

FIGURE 11.6
The action of the soft and hard knee

If you want transparency, however, peak mode should only be engaged when working with short transient hits. If it is applied to other instruments with a more sustained signal, the compressor will clamp down too quickly resulting in a loss of high-frequency detail.

For these instruments, RMS is more sensible. Here, the compressor will only control signals with an average signal level rather than short peaks. Notably, peak and RMS may not appear on software plug-ins as many employ a 'look ahead' system to monitor the audio channel's signals before they even reach the compressor. This renders peak and RMS detection circuitry redundant.

RELEASE TIME

To further maintain transparent compression, control is required to prevent sudden gain changes as the compressor relaxes. With the release parameter, you can define the speed that the compressor relaxes from gain reduction.

Contrary to many articles written about compression, the release parameter does *not* activate when the signal drops below the threshold. Indeed, many compressors do not monitor the threshold, and, instead, the release activates when the output level begins to fall below the input level. This means that both attack and release can occur *multiple times during* gain reduction and therefore both combined can strongly affect the dynamic envelope and tonal content of any sounds if not set carefully.

Figure 11.7 shows the effects the release setting can have on the dynamic envelope. With a release time of 1000ms, the sound spends longer in the compression cycle than if no release is used.

Many plug-in compressors employ auto-release functions. Here, the compressor continually monitors the signal and applies different release times depending on the program material. These provide the best solution when you don't want the compression to be evident.

HOLD

Because the release parameter can activate during the compression cycle, there will be occasions where the release will activate too soon. This is the case when applying compression to low-frequency waveforms. As these are particularly long, the positive and negative states of the waveform can trigger the compressor

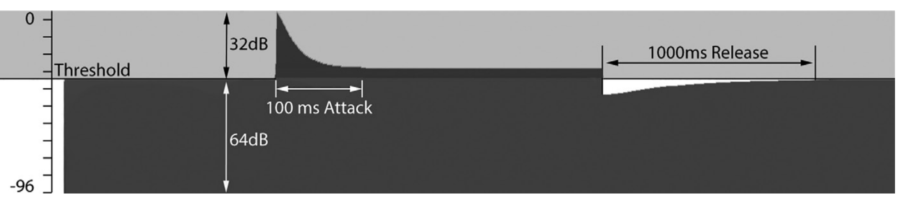

FIGURE 11.7 With high release settings, the timbre remains in the compression cycle for longer

resulting in constant gain changes throughout the waveforms cycle. To prevent this, compressors feature a hold parameter. Again, configured in milliseconds it allows you to determine a finite period that the compressor should wait before initiating the release phase.

MAKE-UP GAIN

As compression reduces the dynamics of a signal, it will reduce the amplitude of the signal. Therefore, after compressing, it is necessary to increase the gain to bring the signal back up to its original amplitude. Caution must be applied when increasing this gain, however. It should be adjusted so that the compressor's output is at the same level as its input because louder invariably sounds better.

This is because as volume increases both the low and high frequencies become prominent resulting in a feeling of increased power from the bass and more clarity from the highs. So, even if the compressor has been configured poorly if the output of the compressor is louder than the uncompressed signal, it will still sound better. Some compressors will attempt to apply an auto gain to match the input to the output, but these can be hit and miss. Instead, it is better to set the output level by 'ear,' and not to solely rely on signal levels or meters.

PRACTICAL COMPRESSION

The first use of compression is for its original application; to control the dynamics during the recording stages. Or sometimes to level out the average signal level to prevent the audio disappearing behind other instruments in a mix. In both these instances, transparent compression is the aim, but every compressor will impart its character regardless of how carefully they are set up and so the choice of the compressor is the first hurdle to overcome.

First, we must choose between solid state and valve topology. In the past 25 years, *analog* and *valve* (tubes) have become the buzzwords of electronic dance music production. Of the compressor plug-ins released over the past 10 years, most have been either emulation of older analog or retro valve gear or at the very least emulate the 'sound' of valves and analog. This shows the apparent trend towards using valve style compression and is due to the introduction of second-order harmonic distortion to a signal.

FIGURE 11.8
The Empirical Lab's FATSO

Second-order harmonic distortion – in the original hardware – was a result of the random movement of electrons in the valve design that occurred at exactly twice the frequency of the compressed signal. The result of this coloration is, somewhat subjectively, a pleasant warmth or character. As the amount of compression or amplitude is increased, the distortion increases respectively resulting in a warmer tone the more heavily compression is applied.

Solid-state circuitry doesn't behave in this same manner. By replacing the valves with transistors, the second harmonic distortion is removed, and the compressor maintains a transparent sound. Solid state *will* still color the sound but not always in a (subjectively) pleasant way. Also, if the solid-state circuitry is driven too hard, it results in audible distortion artifacts.

This is not to suggest you should use only valve equipment. Countless producers can't even hear the effects of second-order harmonic distortion. The result contributes to approximately 0.2% of the signal content and so few producers today spend any time training their ears. But, also, excellent production is a careful mix of all types of gear and techniques. Some audio will sound better with valve, and some will sound better with solid state.

Solid-state does react much more quickly when compared to valve and produces a cleaner, more defined sound. If your ears aren't attuned, this clean response can be beneficial during recording stages with some valve compression applied afterward. There is no 'undo' in real-life recording situations.

In addition to valves and solid state, the second deciding factor is the detection system, because this too will contribute towards sonic color. Indeed, the detection differences and variations between compressors are the reasons why many professionals have a collection of different styles in their plug-in folder or hardware racks.

The different styles of compression can roughly be categorized into variable MU, **F**ield **E**ffec**t** (FET), optical, VCA, and digital.

Variable MU

The variable MU was the first ever mass-produced compressor available on the market. These compressors use valves for the gain control circuitry and do not feature a ratio control. Instead, the ratio is increased in proportion to the amount of incoming signal that exceeds the threshold. That is, the more the level overshoots the threshold, the more the ratio increases.

While these compressors offer attack and release stages, they're not suited towards material with fast transients even with their fastest attack settings. Also, due to the valve design, the valves run out of dynamic range relatively quickly so it's unusual to acquire more than 15–20 dB of gain reduction before the compressor runs out of energy.

MU compressors are renowned for their distinctive smooth character and are best used on vocals, guitars, and, on some occasions, a mix bus. While it can

FIGURE 11.9
The Fairchild 670 Vari-Mu

work well on a mix to add a form of consistency and smoothness, if the music is bass heavy, you must employ a side-chain to prevent the compressor acting on the bass. If not, the compression cycle will slur the transient and muffle the bass. This is also not a fast-acting compressor so it doesn't work well on drums or heavily transient material.

Both UAD and Waves produce perhaps the best software emulations of MU compressors with the acclaimed Fairchild 670, and, in hardware, the most infamous is the Vari-MU by Manley.

FET

FET compressors use a field effect transistor to vary the gain. These were the first transistors to emulate the actions of valves and provide fast attack and release stages. This makes the FETs particularly suited for use on transient material such as bass, drums, and vocals.

FETs (and in particular the 1176 models) distort readily, but this distortion is one of the defining features of the compressor, often described as rich and warm. Many engineers will place the 1176 on a mix bus or in a processor chain without applying any compression at all. It is solely just to capture the sound of the output transformer. Similar to MU, FET suffers from a limited dynamic

FIGURE 11.10
The Urei 1176LN FET compressor

FIGURE 11.11
The LA2A compressor

range, but they sound better the harder they are driven due to the distortion characteristic they impart.

UAD and Waves produce software emulations of FET compressors with the 1176LN compressors, and Softtube produces the FET compressor. In hardware, reproduction versions of the first FETs, such as the UREI 1176LN Peak Limiter and the LA Audio Classic II are a worthwhile alternative.

Optical

Optical compressors employ a photoresistor that reacts to the incoming audio by glowing brighter or dimmer depending on the intensity of the incoming signal. A phototransistor tracks the level of illumination from the bulb and adjusts the gain. Since the phototransistor must monitor the light bulb before it takes any action some latency is created in the compressor's response, so the more heavily the compression is applied, the longer the envelope times tend to be. Consequently, most optical compressors utilize soft knee compression.

This soft knee response aids to create a more natural sounding and transparent compression but does mean these compressors are not ideal for compressing transient material such as drums. The best use of an optical compressor is with vocals, synthesizers, pianos, and a mix bus (although it should be side-chained for this use).

Plug-in giants UAD's LA2A and the Joe Meek SC2 Pro Tools plug-in are popular choices for Opto compressors, but one of the most successful is the Softube CL1B compressor.

VCA

Voltage Controlled Amplifier compressors are the fastest available. They offer the highest amounts of gain reduction coupled with the highest level of transparency. This makes the VCA the best compressor for general duties where clarity is of paramount importance. However, they do distort aggressively if pushed hard and the lesser quality VCA units remove high-frequency detail during the compression cycle.

These are sometimes used to pump a mix because of the fast reaction times of the envelopes. Many plug-in manufacturers produce VCA compressors and emulations. UAD produce the infamous Drawmer dBx 160 (the first commercially available VCA compressor) while Brainworx produces the VSC-2 quad discrete compressor.

Possibly the most recognized hardware VCA compressors are the SSL and the Empirical Labs Stereo Distressor. The last is a digitally controlled analog compressor that allows you to switch between VCA, solid-state and op-amps. Two versions of the Distressor are available to date: the standard version and the British version. Of the two, the British version produces a much more natural, warm tone and is the preferred choice for many producers.

Digital compressors

Computer-based digital is the generic name to encompass software compressors that do not emulate any other form of compression. Since it is possible to emulate classic compressors in the digital realm, many developers prefer to release compressors based on valves, etc., since these are guaranteed to shift more units.

FIGURE 11.12
The DBX 160 compressor

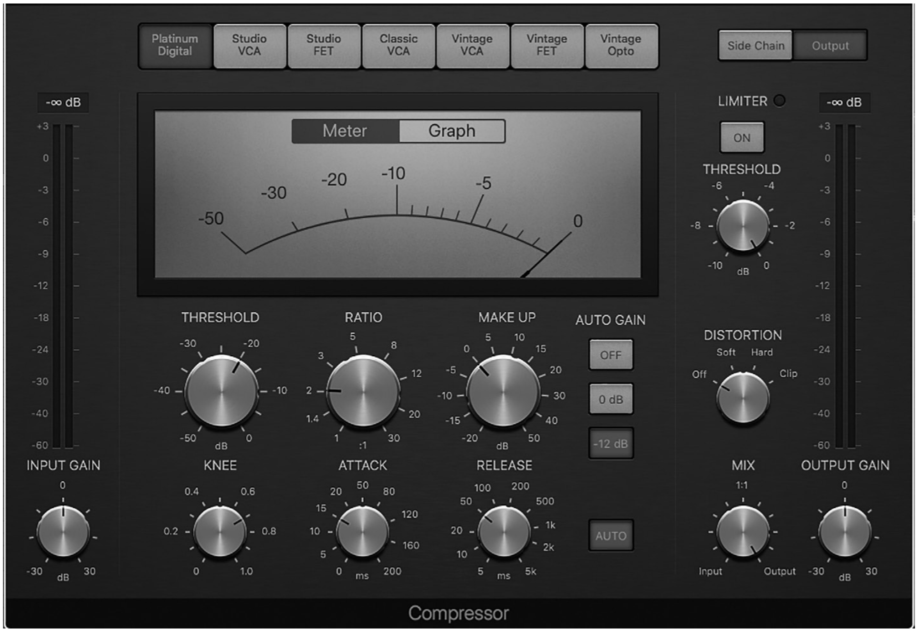

FIGURE 11.13
Logic Pro X digital compressor

Nonetheless, while these compressors do not feature any coloration to attract producers, they are the most precise and transparent. They will often employ a look ahead function to monitor the audio channel before it reaches the compressor and this allows them to predict and apply compression without any transient creeping through the threshold. Although transparency is not always a selling point for compression, it does have many uses on vocals, drums, guitars, and synthesizers.

SETTING UP

Once you've made an *informed* choice on the compressor for the job at hand, it needs setting up. Here, we must ask ourselves what the purpose of the compression is. Is it to control dynamics transparently? Is it to add a specific coloration? Or is it to redesign the dynamic envelope?

For the typical compression duties such as recording or dynamic limiting for mixing, the purpose is to program the compressor to maintain transparency. For this, start playback of the audio through the compressor and set the ratio at 4:1.

With the audio playing, gradually reduce the threshold parameter until the loudest parts of the signal read between −8 and −10 dB on the gain reduction meter. Set the attack parameter to 0ms or the fastest possible option and the release should be set around 500ms. Using these as preliminary settings, you can then adjust them further to suit any particular sound.

The higher the dynamics of the instrument that you're compressing, the higher the ratio and lower the threshold should be. This will help to keep any massively varying dynamics under control, but it is essential to listen carefully to the results as you adjust these parameters and, in particular, pay attention to the initial transient of the audio. Here, you should be listening for loss of high-frequency detail and how much the compressor molests the transient.

Begin to slowly increase the attack while listening and watching the gain reduction meter. The attack should be set so that the transient isn't dulled by the action of the compressor while also ensuring that not too much of the signal is creeping past the compressor unaffected. During this adjustment, the compressor should often be bypassed to compare the compressed and uncompressed version.

The release should be set so that the action of the compressor isn't immediately noticeable but also so that the compressor has time to recover when the signal has left the compression cycle. This is best accomplished by reducing the release until you can hear the action of the compressor and then begin to increase to the point just beyond being noticeable. On many compressors, you may note that as the release is lengthened, the compressed audio loses high-frequency content. If the release is shortened, the high-frequency detail increases. It is essential to listen to these effects so that you can set the release correctly.

It is also essential to ensure that the release is set short enough so that gain reduction stops in silent passages. You can do this by checking that the gain reduction meter drops to 0 dB during any passages that you do not want in the compression cycle. For example, if you were compressing a hi-hat so that the gain reduction meter reads –8 dB on each hat strike, the release should be set so that the gain reduction meters falls to 0 dB between each hat.

For example, if the gain reduction were to drop to say, –2 dB between each hit rather than 0 dB, it's safe to assume that it would only require 6 dB of gain reduction rather than 8 dB. Alternatively, the release parameter should be reduced. If not, the compressor will permanently apply 2 dB of gain reduction throughout since it's unable to recover before the next hat.

When a compressor is unable to recover from its cycle, it will distort the transients as well as modify the dynamic envelope. This means you should also take into account insert effects chains. If a compressor is being applied for dynamic control rather than sound design, placing a compressor *after* a delay effect would result in the delays compressed and therefore recovery time may not be possible at all.

Finally, by ear, you should increase the gain of the compressed audio signal, so it's the same level as the signal entering the compressor. The best way to accomplish this is to consistently bypass the compressor to hear the signal before and after compression and adjust the make-up gain so that they appear to be at the same audio level.

EVIDENT COMPRESSION

While the aim of compression is to remain transparent, deliberately making a compressor's action apparent forms the cornerstone for the sound of dance music. As touched on previously, compression can be used as a sound design tool to modify the dynamic envelope of individual tones. For example, by employing heavy compression (a low threshold and high ratio) on an isolated snare drum, the snare's attack avoids compression, but the decay is squashed. This decreases the transient to body dynamic ratio producing a stronger 'thwack'-style timbre familiar in many genres.

Similarly, employing a similar approach on bass or lead timbres can result in a more powerful, energetic sound or, if applied heavily, it can result in the body and release becoming louder than the attack. This produces a sound that appears to 'suck' upwards as it progresses and is a frequent approach in some genres.

The most popular use, however, is the abuse of side-chains. Many hardware and software compressors feature a side-chain function. With these, you can insert a secondary signal from a hardware device or another audio channel within the DAW that can control how the compressor behaves on a signal entering at its primary inputs.

An example of side-chain compression is during a radio show. When a radio DJ speaks, the background music lowers in volume and when they stop talking the volume of music increases again.

This is accomplished by feeding the music signal input into the primary inputs of the compressor or, in the case of a DAW, placing a plug-in compressor on the track containing music. A microphone is connected to the side-chain input, or in software, a secondary audio channel can be configured to act as a side-chain input to the compressor. With this configuration, the threshold of the compressor is controlled by the side-chain signal, so whenever the side chain signal is present the compressor activates.

This application has many practical uses during mixing by 'ducking' instruments to allow others to sit over the top more easily. In some mixes, a lead guitar or instrument will often occupy the same frequencies as the human voice. This results in a mix that can appear cluttered making it difficult to distinguish the voice or hear the words. Placing a compressor on the guitar track and using the vocals as a side-chain signal can avoid this. By setting the compressors ratio, attack and release parameters appropriately, every time the vocals are present, the compressor will activate and lower the gain of the guitar.

This same technique has become requisite in dance music production, although rather than vocals acting on the primary signal, a kick drum is used. This effect referred to as *gain pumping* is accomplished by running the entire mix through a compressor and inserting a kick into the side-chain inputs. By doing so, every time a kick occurs the mix ducks and produces the effect of a kick drum punching holes in the mix.

This is accomplished in digital audio workstations by first duplicating the kick drum channel to produce two channels. The second channel is set to no output (so it provides no sound at the mixers outputs), and a compressor is placed across the entire mix – this can be a bounced mix on a single channel or all the current mix channels sent to a single bus. The second kick channel with no output is then used as the side-chain insert.

If you don't have a mix available, the resource contains two tracks – a complete mix track and a kick drum track that you can use to practice on:

1. Set your workstations tempo to 134BPM and insert both audio files from the resource package onto two channels in your digital audio workstation.
2. Insert a compressor with a side-chain option on the full mix track and set the ratio of the compressor to 4:1 with a fast attack and release.
3. Use the kick track as a sidechain for the compressor.
4. Begin playback and then slowly reduce the threshold. As you do so, the mix will begin to pump with every kick.
5. Finally, experiment with different attack, release, and threshold settings until it produces the effect you prefer.

DMM resource: Contains a video file example of pumping a mix.

This effect is not only used to pump an entire mix and is used in many genres to pump bass lines, melodic leads, and rhythmic pads. Employing the same technique but placing the compressor on a sustained bass note can produce a pumping bass.

DMM resource: Contains a video file example of pumping a bass.

This technique is also used in genres that use deep bass elements that play consecutively with the kick drum. A deep kick and bass playing together can merge creating a muddled sound as the frequencies mash together. By pumping the bass with the kick, this can be avoided. With a compressor inserted onto the bass track and a kick as the side-chain signal, every time the kick occurs the bass will drop in volume, creating free space for the kick.

Some producers choose not to use the same kick as the track and prefer to employ a secondary kick to side-chain the compressor. This second kick is set to no output (so it cannot be heard) and functions solely as a sidechain trigger. David Guetta is well known for employing this technique.

Rather than employ a compressor and side-chain, a favorite technique for many is use software plug-ins to emulate the effect. Many software programmers have released plug-ins to produce the chain effect but with more control. Software such as Xfer's LFO Tool and CableGuys Volume Shaper can both create pumping effects. These do not require a side-chain input; you can just draw the curve you want to act as the side-chain.

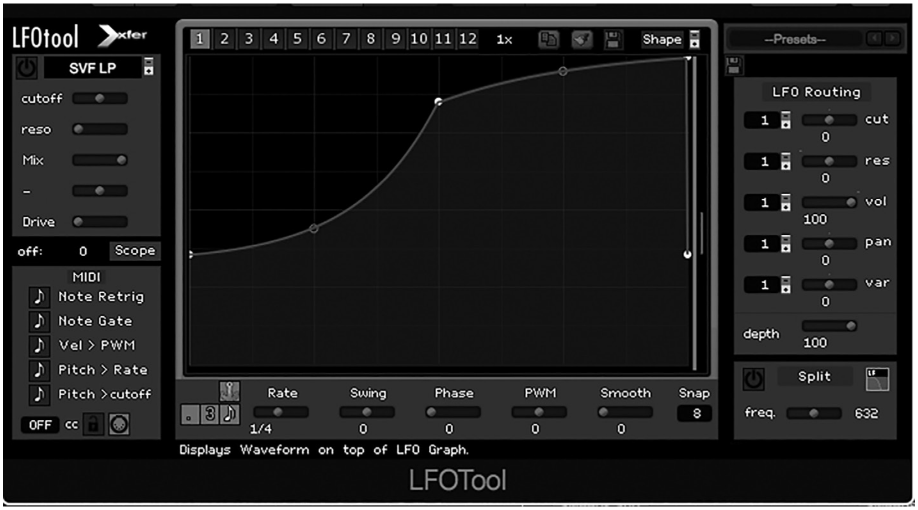

FIGURE 11.14
Xfer LFO tool showing the side-chain action

Some genres such as Downtempo, Ambient House, Jazz House, and Trip-Hop don't require such a heavy-handed pumping method. Here it's generally accepted to gain pump the mix without side-chaining a kick. This requires a more careful and considered approach, however, because it's possible to destroy the excursion and end up with a flat sound.

Figure 11.15 shows a typical ambient mix in the waveform editor of an audio workstation. It's clear from looking at the waveform that the highest energy – the loudest part of the loop – is derived from the kick drum. If a compressor is inserted across this particular mix and the threshold of the compressor is set just below the peak level of the loudest part, each consecutive kick will activate the compressor. If the ratio, attack, and release are then applied *carefully*, it is possible to create a gentler form of pumping that can appear more cohesive to the mix.

A good starting point for this is a ratio of 3:1, mixed with a fast attack and release time. Slowly reduce the threshold until it's only activating on each kick in the example mix. An easy way to accomplish this is to watch the gain reduction meter on the compressor and ensure it only illuminates every time a kick occurs.

Finally, experiment with the release time of the compressor and note how as the release is lengthened, the pumping appears to be less dramatic. Similarly, note that as the release time is shortened, the gain pumping becomes more evident. This is a result of the compressor activating on the kick before rapidly releasing when a fall in gain reduction is detected. The rapid change in volume produces a similar effect as side-chain pumping but is less obvious and produces a gentler feel to the mix.

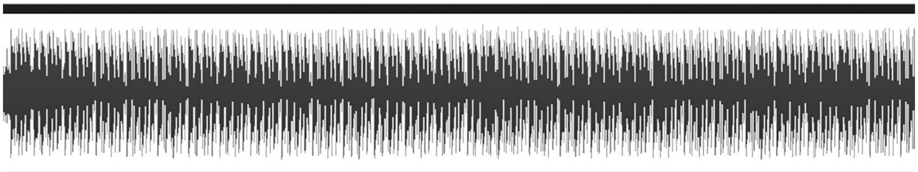

FIGURE 11.15
The waveform of a mix

The key element to this effect lies with the timing of the release parameter, and this depends on the tempo of the drum loop. It must obviously be short enough for the compressor to recover before the next kick but also long enough for the effect to sound natural, so experimentation and careful listening are crucial.

CHAPTER 12
Further processors

Like all tools, modern technology has produced some wonderful moments in music, and also some horrors.

Hugh Hopper

Although compression has earned its place as the processor for attaining the 'sound' of electronic dance music today, it's not the only useful processor. Indeed, further processors play a role in both the sound design process and the requisite feel of the music. This chapter concentrates on the behaviors of the different processors that are widely used in the design and production of dance music.

LIMITERS

A limiter is a dynamic processor comparable to the action of a compressor. But rather than compress by a specified ratio when the threshold is exceeded, a limiter prevents the signal from exceeding the threshold. Therefore, no matter how loud the input signal is, the limiter will compress the peaks so that the signal cannot exceed the set threshold. This action is termed *brick wall* limiting. Notably, some software limiters may permit a slight increase in the level above the threshold. The amount of limiting overshoot allowed depends on the unit, but this form of soft limiting very rarely exceeds a few decibels. It is only employed to permit the odd transient to overshoot. This allows for a more natural or open sound from the limiter and is the preferred choice of some engineers.

Because a limiter's action is restricted to controlling peaks that may result in distortion, limiters only require three controls: an input level, a threshold, and an output gain. The input defines the overall signal level entering the limiter. The threshold, like a compressor, establishes the level at which the limiter will begin attenuation of the signal. Signals that exceed the threshold are immediately compressed to prevent clipping or distortion of the outputs. The final parameter is an output control to increase the average signal level.

As discussed in previous chapters, because we determine a sustained timbre to appear louder than a transient, by employing a limiter, it is possible to increase the perception of loudness in a mix. However, as with compression, a limiter can also severely modify the dynamic envelope, and therefore it must be used with caution. Judicious application of limiting with little regard to the dynamics can result in a timbre or mix that appears louder and thus more appealing, despite destroying the natural energy and presence.

To maintain a natural, uncolored sound, some limiters feature additional functions such as a release parameter. With this, it is possible to set the time it takes for the limiter to recover after limiting. As with compression envelopes, this must be applied cautiously to ensure that the limiter has enough time to recover before the next threshold is breached. If the limiter cannot recover in time, it may distort the transients that follow and modify the dynamic envelope.

As the purpose of any limiter is to prevent transients from breaching the threshold, there is technically no requirement for an attack parameter because as soon as a breach is detected, it should be limited. However, many software plug-ins have the advantage of being able to *look ahead* and predict the upcoming signal, so they employ an attack parameter.

The attack is user definable but will be very short, or it may employ a soft or hard setting. This operates in a similar respect to the knees on a compressor. With a hard attack, the limiter will act as soon as a peak is close to overshooting while a soft attack has a smoother curve with 10 or 20 milliseconds. This approach reduces the likelihood of clicks or snaps that result from the limiter acting too quickly on the incoming signal. These look ahead limiters are sometimes called ultramaximizers.

FIGURE 12.0
An ultramaximizer

Although some producers employ a limiter on instruments and parallel sub-mixes, typically their primary purpose is to restrict the dynamics of a mix. By reducing the dynamic ratio of a piece of music, the average signal level can be increased, and thus it can be made louder. This has contributed to what is termed the 'loudness war,' as natural dynamics suffer in the effort to produce the loudest mix. While the argument rages, you must nevertheless exercise caution when limiting to ensure there is a careful balance between maximum loudness and the kick's excursion.

With electronic music, the kick drum provides the central rhythmical element of the music. It is the most crucial element of any dance music record and must exhibit the energy to punch a listener in the chest. This punch is in direct relation to the amount that the kick physically moves the speaker's cone. The more the speaker cone moves, the greater the punch of the kick becomes. This speaker cone *excursion* is related to the size of the kick's peak with the rest of the music's waveform.

If the difference between the *peak* of the kick and the main body (known as the **R**oot **M**ean **S**quare, or RMS, of the music) is reduced through severe limiting, the kick's excursion is reduced. This is because as the dynamic range is restricted, the RMS will move the cone by the same amount. The effect of this substantial dynamic restriction is a mix that, although loud, lacks the energetic punch of the kick. The impact of this application is shown in Figure 12.1 and can be heard on the book's resource pack.

DMM resource: The effects of severe dynamic restriction.

On average for most electronic music, approximately 3 to 6 dB is a reasonable amount of limiting without the kick excursion being too heavily restricted. However, the exact figure depends on the mix. If the mix is already heavily compressed it would be wise to avoid any more than 3 dB at the limiting stage, otherwise, the dynamic ratio may be too low.

FIGURE 12.1
A mix before and after heavy application of limiting – note how as the RMS increases, the peaks decrease

NOISE GATES

A noise gate is a dynamic processor that attenuates signals *below* a given threshold. Its purpose is to remove any low-level noise that may be present during what should be a silent passage.

For example, during a recording session with a vocalist (often termed 'talent' in the industry), it is common practice to leave the recording device running throughout the length of the track. This is to capture all the nuances of a single performance. With this approach, there will also be times where the talent rests for a moment between bars of the song.

During these silent passages, it's possible for the microphone to pick up extraneous noises such as the talent tapping her foot, resting, breathing heavily, clearing her throat, or adjusting her stance. With a noise gate and a low threshold set below the talent's performance level, every time she stops singing, the signal will fall below the threshold, and the noise gate will activate. When activated, the gate will close off any signals resulting in silence.

While in theory, a noise gate could just consist of a threshold parameter, it is more complicated in practice. We must consider that not all sounds start and stop abruptly. Some instruments and vocals will progressively increase in gain. Violins, strings, and pads, for example, will gradually increase in gain over a finite period. If a noise gate featured only a threshold parameter, these sounds would be suddenly audible once they exceed the threshold. Similarly, if they also were to reduce in gain gradually, they would be abruptly silenced as they fall below the threshold again.

To counteract this, noise gates feature attack and release parameters. Similar to a compressor's envelope, these are configured in milliseconds and, allow you to determine the attack and release times of the gate's action. The attack parameter specifies how quickly the gate opens after the threshold breach while the release parameter determines how fast the gate will close after the input signal has fallen below the threshold.

Like compression, both must be configured carefully because an incorrect adjustment could result in both envelopes modifying the dynamic envelope of the signal. For example, if the gate's attack period is slower than the incoming signal, the noise gate will remodel the signal's attack stage. Similarly, if the release were too short, the dynamic envelope would be further molested. The possible dynamic modifications of incorrect adjustments are shown in Figure 12.2.

Another consideration is that not all signals will remain at a constant amplitude throughout their period. When the talent holds a note, for example, it will result in a varying degree of amplitude. If this occurs close to the threshold, the signal may fluctuate above and below the threshold. With a short attack and release, the result is a constant activation and deactivation of the gate producing an effect known as *chatter*. Alternatively, with a longer attack and release, it may create severe volume fluctuations throughout the performance.

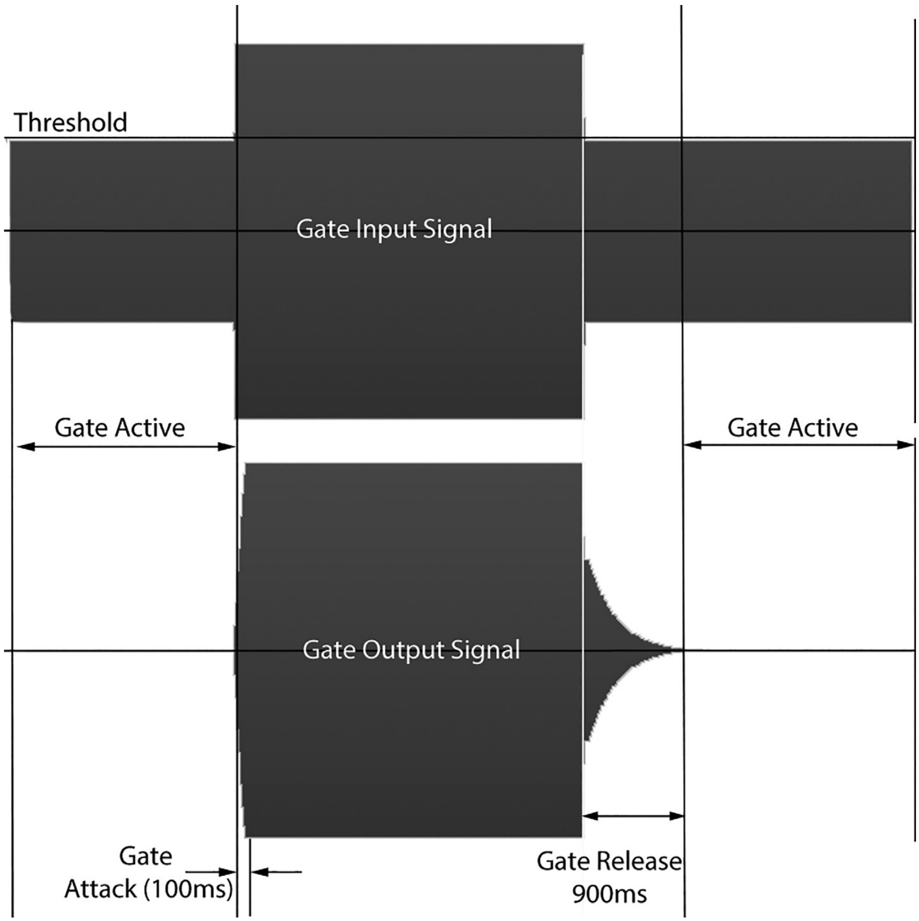

FIGURE 12.2
A noise gate modifying the dynamic envelope

To prevent this, gates feature an automatic or user definable hold time. Using this, you can determine a specific amount of time that the gate will pause before activating its release after a signal has fallen below the threshold. This way, if a signal falls below the threshold for only a short period before returning above it, the hold parameter will prevent the gate from chattering.

This hold function is often confused with a similar process termed *hysteresis*, but the two have a substantial difference. Whereas hold will force the gate to wait for a predefined amount of time before closing, hysteresis adjusts the threshold's tolerance independently for both opening and closing the gate.

With hysteresis, if the threshold of the gate were set at –12 dB, the audio signal must breach this before the gate opens. However, the same signal must fall a few extra decibels below –12 dB before the gate will close again. This action produces results similar to a hold function but produces more natural sounding results than hold and therefore is the preferred choice for real-world instruments and vocals.

In addition to hold or hysteresis, many gates feature a range control. This controls the amount of attenuation for signals that fall below the threshold. In many instances, a gate will close completely when a signal falls below its threshold for total silence. However, there are occasions whereby you may wish to keep some background noise, for example, when recording drums or acoustic guitars. Drum strikes will resonate, and acoustic guitars exhibit fret squeal as the performer's fingers move up and down the fret.

We expect to hear these effects, and if they are removed via a gate, it will sound unnatural. A similar effect can also occur with talent breathing. Unfortunately for engineers, talent needs to take a large intake of breath before performing a verse or chorus. If a noise gate removes this breath entirely, it may appear forced and unnatural.

The range is calibrated in decibels and allows you to define how much any signals that fall below the threshold of the gate are attenuated. The further this range is increased; the more signal gain is reduced until at its maximum setting the gate will silence altogether. This parameter is configured so that any signals that do fall below the threshold are only just audible when placed within the mix to provide a more realistic nature to the sound.

Many gates, and in particular software emulations, feature side-chain inputs. These function in the same manner as a compressors side-chain. These *key* inputs allow you to input a workstation's audio channel into the gate to control the threshold. This action has various creative uses.

The most common is to place the noise gate on an audio track with a sustained timbre, such as a pad or string section and then program a hi-hat rhythm on another track. This hi-hat rhythm is used as the key input of the gate so that each time a hi-hat occurs, the gate opens and the sustained pad timbre is heard. Alternatively, if the workstation permits, you could input a virtual instrument into the gate's key inputs so that each time a note is struck on the instrument, the gate opens allowing the pad sound through.

Naturally, both these actions supersede the threshold of the gate, but the attack, release, range, hold, hysteresis, and range controls remain available permitting control over the reaction of the gate on the audio signal.

TRANSIENT DESIGNERS

Transient designers are yet another form of dynamic processor that will modify a sounds dynamic envelope. This deliberate modification of a dynamic envelope is a significant sound design tool for the creation of electronic music. Consequently, a transient designer should be considered one of the fundamental tools to own.

Many transient designers feature only two controls: attack and sustain. With these, you can modify the attack and sustain characteristics of a prerecorded audio file similar to a synthesizer. As we determine a significant amount

FIGURE 12.3 The Oxford Envolution transient designer

of information through a sound's attack stage, adjusting this can change the appearance of a timbre.

For example, by reducing the attack stage, it's possible to move it further back in a mix while increasing the attack will bring it to the forefront. They also prove to be indispensable with drum loops since modification of a kick, snare, or a hi-hat's attack and release stage can affect the groove of the loop.

EQ

EQ is principally a frequency-specific volume control that permits you to attenuate or increase the gain of a specific band of frequencies. This application can be in the context of an individual tone or more often in the context of a complete mix to provide each instrument its space to breathe in and be heard clearly. As basic as this premise is, EQ is a problematic processor to master, and despite the relatively small amount of parameters offered, the theory behind its application is involved.

The most commonly employed EQs are parametric and paragraphic. Both these perform the same function but display information differently. A parametric EQ shows only parameters while paragraphic shows both settings and a graphic display of the EQ applied.

Regardless, both consist of three main parameters over frequency, bandwidth, and gain. The frequency pot is used to select the center frequency that you want to either enhance or remove. The bandwidth pot determines how much of the frequency range either side of this central frequency should be affected, and the gain is then used to increase or attenuate the gain of these selected frequencies.

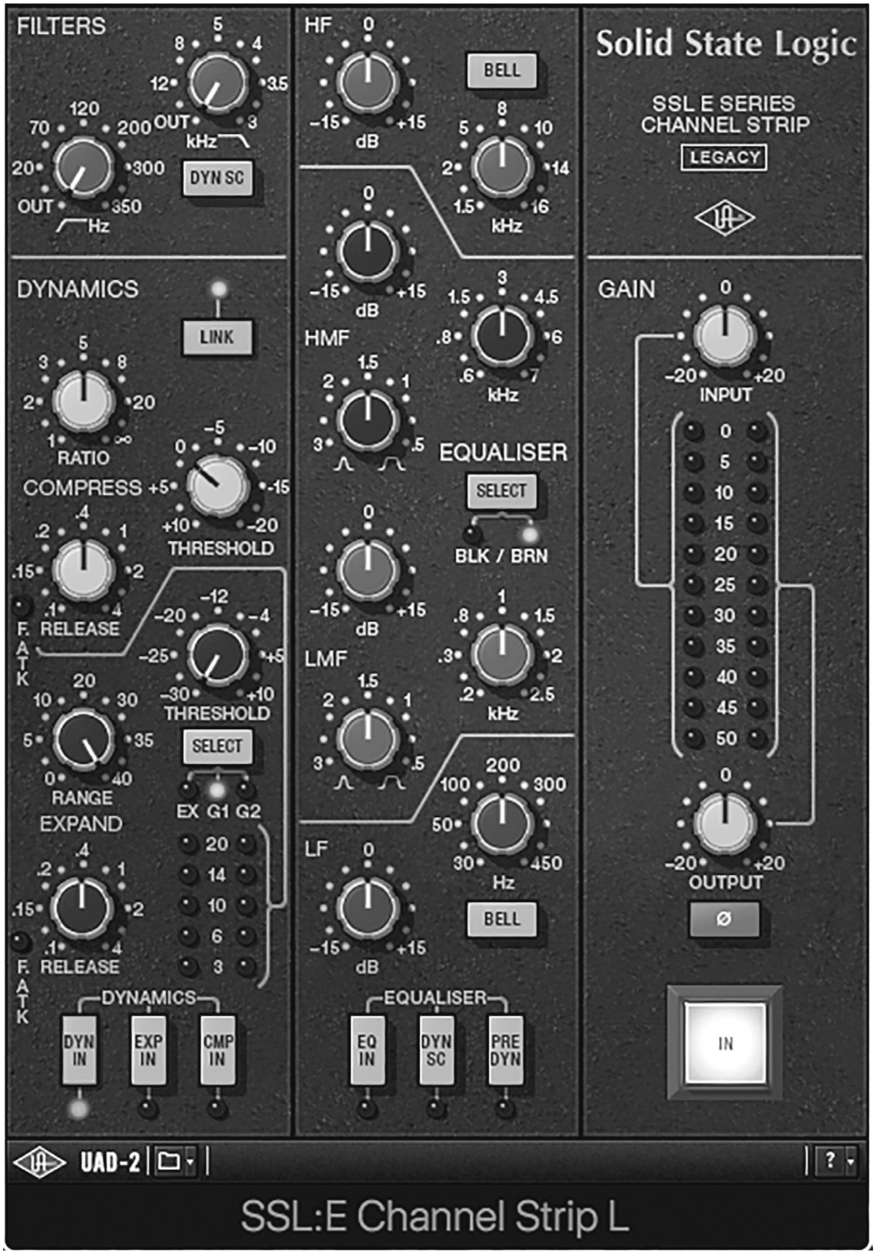

FIGURE 12.4
A parametric EQ

FIGURE 12.5
A paragraphic EQ

For example, if you wanted to EQ a timbre that contained frequencies from 600 Hz to 9 kHz and the frequencies between 1 and 7 kHz required a cut, using the frequency parameter, you would home in on the center frequency (4 kHz) and then set the bandwidth broad enough to affect 3 kHz either side of the center frequency. By then reducing the gain of this center frequency, the frequencies 3 kHz either side would be attenuated too. The size of the bandwidth is known as the Q (for quality) and the smaller the Q number, the larger the 'width' of the bandwidth.

As simple as this premise may appear it's a little more complicated in theory since Q is not constant and will change depending on the amount of boost or cut applied. For example, if an unusually large boost or cut is applied the bandwidth will begin particularly wide and become progressively finer as it carves further into the frequency range. This action is shown in Figure 12.6.

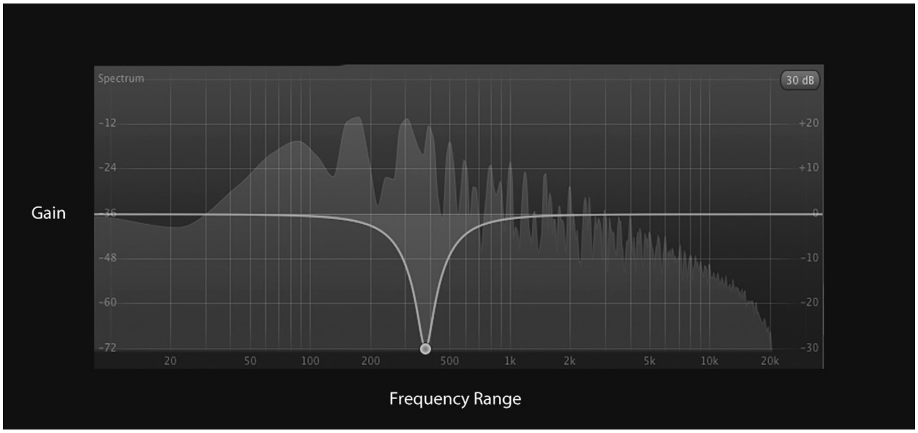

FIGURE 12.6
The action of non-constant Q

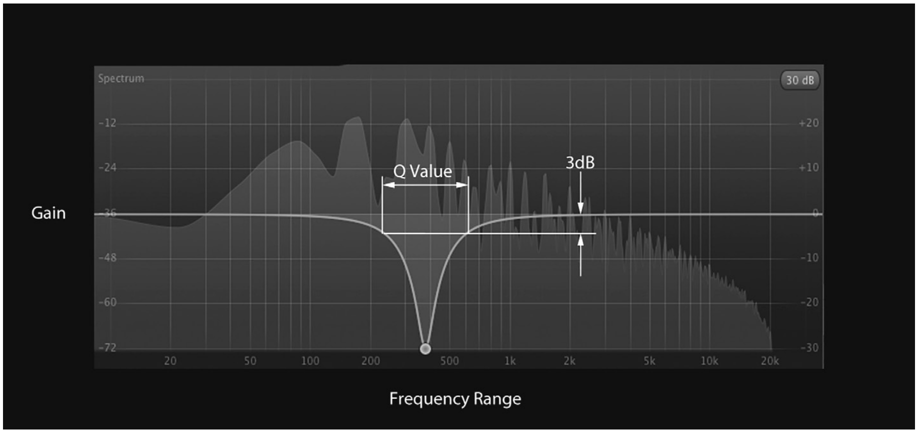

FIGURE 12.7
The measurement of Q

This non-constant Q action makes exact measurements challenging and therefore the actual value of the Q is determined by measuring 3 dB into the boost or cut applied. This is shown in Figure 12.7.

While taking a measurement 3 dB into a cut or boost will provide a reliable method for determining the current width value of the Q it is inadvisable to refer to a Q value as a specific range of frequencies. Instead, it should be related to in octave values since octaves are exponential. This means a Q frequency value will be different depending on the octave it is being applied too.

As discussed in a previous chapter on acoustic science, Pythagoras discovered that the relationship of pleasing harmonics was strongly related to mathematics and that exponential increases in frequencies were responsible for harmonics that the human ear found pleasing. From this, he also determined that doubling a frequency was equal to moving up by an octave in the musical scale.

This means that if we consider that A1 on a musical scale is 110 Hz, by doubling that frequency, it will produce the tone A2 occurring a 220 Hz. Moreover, if the frequency of A2 were increased to 440 Hz, it provides a tone at A3, and if that again were doubled to 880 Hz, it produces the tone A4. This exponential relationship of frequencies and the octave is shown in Figure 12.8.

With this in mind, if you chose to apply an EQ cut of 6 dB at 500 Hz, it would occur between A3 (440 Hz) and A4 (880 Hz) on the musical scale. If the Q value here were to carve out 232 Hz between these two octaves, it would leave 208 Hz unaffected in this particular octave. Thus, a frequency cut with a Q of 232 Hz removes a significant proportion of frequencies in this octave.

(440 Hz between A3 and A4, removing 232 Hz from that = 440 Hz – 232 Hz = 208 Hz)

Compare this to if you were to apply that same Q of 232 Hz between A6 and A7 on the scale. Here, A6 is 1760 Hz, and A7 is 3250 Hz, a difference of 1760 Hz. If the cut were placed at 2 kHz with a Q of 232 Hz, it would leave 1528 Hz unaffected in this octave range. Comparatively, this is an insignificant amount – not even removing a quarter of the frequencies in this octave range – compared to over half the frequencies removed when applied between A3 and A4.

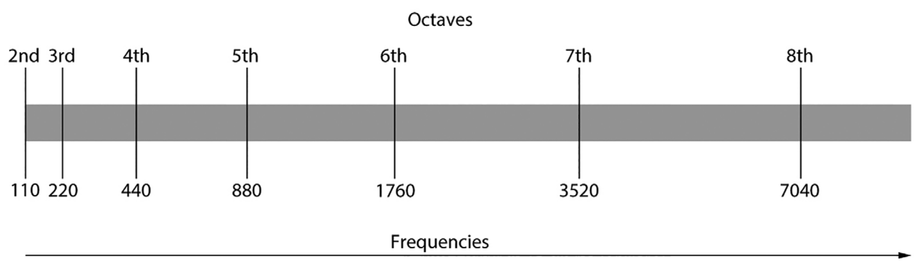

FIGURE 12.8
The exponential increase of frequencies related to octaves

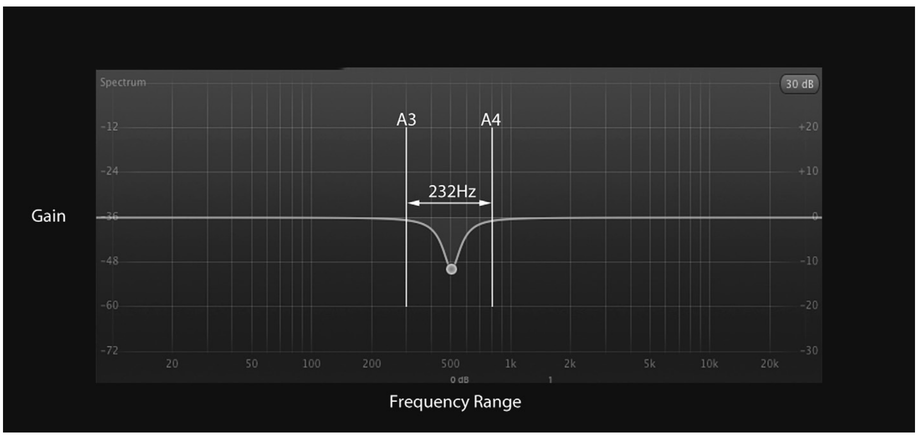

FIGURE 12.9
232 Hz cut applied between octaves A3 and A4

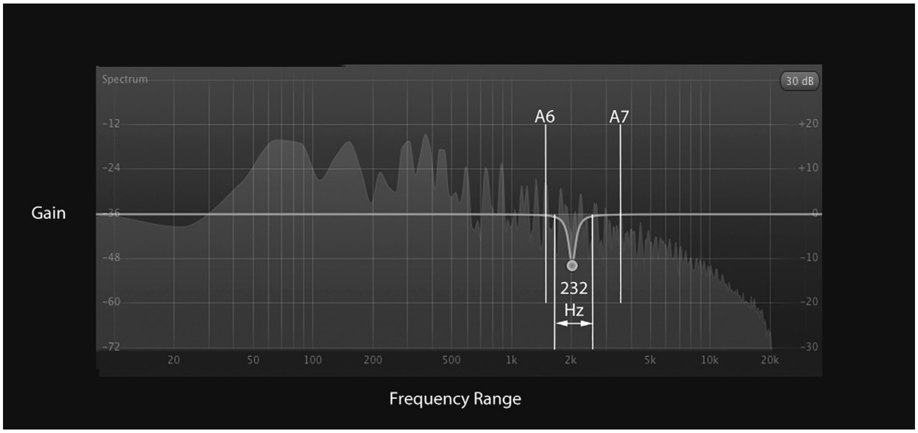

FIGURE 12.10
232 Hz cut applied between octaves A6 and A7

Consequently, it's easier to refer to Q regarding octave measurements rather than frequencies. By referring to a Q's value in octaves rather than frequencies, the Q will carve out an exponential number of frequencies depending on the octave. Therefore, we perceive it to affect the same amount of frequencies regardless of the octave. With this in mind, engineers and producers tend to rely on some basic Q values for production work. These are:

- Q of 0.7 = 2 octaves
- Q of 1 = 1 1/3 octaves
- Q of 1.4 = 1 octave
- Q of 2.8 = ½ octave.

Very generally speaking, a bandwidth of half of an octave or less is used for removing problematic frequencies while a width of a couple of octaves is suitable for shaping a large area of frequencies. For most work, however, and especially for those new to working with EQ mixing, a non-constant Q of 1.0 is perhaps the most suitable. This is often referred to by many engineers as the 'magic Q' since it can be used to solve most problems within mixing and sounds more natural to the ears.

It is, of course, possible to calculate your Q values using the following mathematical theorem of Q = (CF/(TF – BF)). Here, CF relates to the center frequency while TF and BF refer to the top and bottom frequencies to be carved out.

For example, if a central cut is placed at 1000 Hz and it resulted in affecting the outermost frequencies of 891 Hz and 1123 Hz, these would be the top and bottom frequencies.

In this situation, you would begin with the simple math of TF – BF (1123 Hz – 891 Hz = Q of 232 Hz). With the central frequency placed at 1000 Hz, it's a simple case of dividing 1000 Hz with the Q frequency of 232 Hz that produces a Q of 4.31 (1000 Hz/232 Hz = 4.31). This is approximately one quarter of an octave.

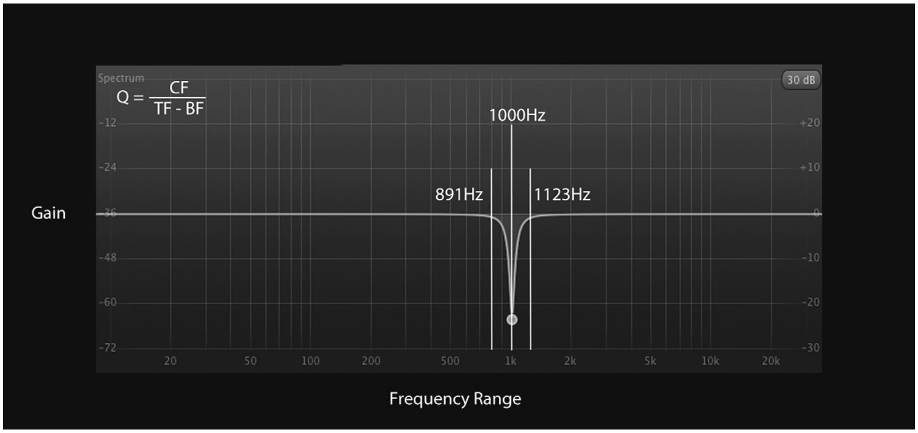

FIGURE 12.11
Frequencies affected by a cut

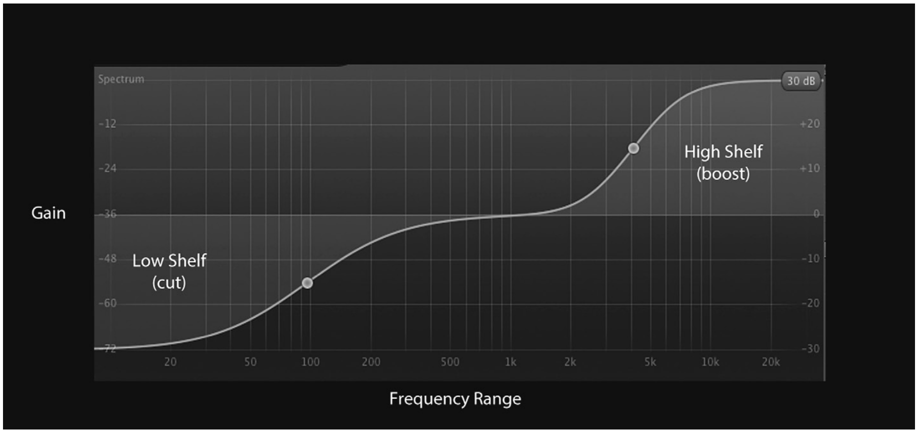

FIGURE 12.12
The action of shelving filters

In addition to the standard parametric features on a plug-in EQ unit, they will often also permit the use of shelving filters. These are similar to the low-pass and high-pass filters that adorn many synthesizers. As previously discussed, a low-pass filter will attenuate the higher frequencies that are above the cut-off point while a high-pass filter will attenuate all the lower frequencies that are below the cut-off point. Within EQ, a low-shelf filter can either cut or boost any frequencies that occur below the cut-off point. A high-shelf filter can cut or increase any frequencies that arise above the cut-off point.

The specifications of the plug-in will determine the amount of control offered over shelving filters, but typically they will provide two parameters: one to select the frequency (sometimes termed the knee frequency) and a second parameter to cut or boost the frequencies above or below the chosen frequency.

FIGURE 12.13
The API 560

Since many speaker systems are incapable of producing frequencies below 20 Hz, it is good practice to remove these frequencies by 'rolling off' all the frequencies below 20 Hz. Doing so, you can ensure that the speaker does not lose energy (and thus volume) by attempting to replicate frequencies beyond its frequency range.

Further EQ

Both parametric and shelving filters are the most commonly used EQ, but there's a secondary form of EQ available: the graphic band EQ.

Graphic band EQs, or more simply graphic EQs, are different to parametric EQs in that all the bands are fixed, and it is only possible to increase or attenuate these fixed bands. This means the more bands that are available the smaller the frequency bands that are affected will be.

Most graphic EQs employ a constant Q so that regardless of how much gain or attenuation is applied the Q remains constant throughout, provided they have a large number of bands to cover the frequency range; this makes graphic EQs particularly surgical. This is the reason why they have been the mainstay of professional mastering suites as it permits the engineer to contour the tonal content of a mix accurately.

> Not all graphic EQs are constant, and some of the more revered units will feature 10 or fewer bands but employ a non-constant Q. An impressive example of this style is the API-560, a unit that is one of the most respected and admired 10-band EQ systems. This acts similarly to parametric/paragraphic EQs, with the Q becoming progressively thinner the more the specific frequencies is attenuated or boosted.

Although these forms of EQ may seem particularly limited, their use should not be underestimated. The benefit of any reasonable graphic band EQ is that they are immediate and do not require you to home in on a specific frequency. This immediacy allows you to quickly experiment and concentrate on the 'sound' rather than the technical theory behind its use.

Above all, the potential audience does not *see* EQ, they only hear it. One of the most significant problems facing many producers today is the visual aspect. Working on 'visually enhanced' EQs makes you more likely to trust your eyes rather than your ears.

EXPANDERS

Expanders are compressors working in the opposite direction. Whereas a compressor reduces the dynamic range of an audio signal, an expander increases it. However, since expanders are the exact opposite of compressors, they feature the same parameters: a threshold, ratio, attack, and release time.

The threshold parameter on an expander determines which parts of the incoming signal should be processed. That is, any signals that breach the current threshold setting are increased in volume (i.e., expanded) compared to those that remain below that are left unmolested. This action increases the dynamic range of the signal.

The level of expansion applied is expressed via the ratio control. This performs in the same manner as the ratio parameter discussed in compression. It allows you to set the ratio of expansion compared to the level of the signal level breaching the threshold. Similarly, both the attack and release parameters operate in much the same way, permitting you to set the time it takes for the expander to respond to signals that exceed the threshold and the time it takes for the expander to stop processing when the signal begins to fall. Finally, an output gain can be used to increase or more usually attenuate the signal at the expander's output stage.

CHAPTER 13
Effects

Know what you're trying to do before you do it. Turning knobs at random isn't enlightening any more than throwing paint at a wall blindfolded will let you paint a nice picture.

Steve Albini

Alongside processors, an understanding of effects is equally important. For obvious reasons, what follows is by no means an exhaustive list, to do so would require a book by itself. Instead, I plan to discuss the most commonly employed effects within dance music to introduce the foundations of production.

REVERBERATION

Reverberation describes the natural reflections we hear in different environments. It provides essential auditory clues as to the space, furnishings, and general structure of the area that a sound exists in. As described in an earlier chapter, when any object vibrates the resulting changes in air pressure emanate out in all directions, but only a small proportion of this reaches the ears directly. The rest of the energy strikes nearby objects, floors, ceilings, and walls before then entering the ears.

These secondary reflected waves create a series of discrete echoes following the direct sound, and it's from this 'reverberant field' that we decipher information about our surroundings. This effect is so fundamental to our survival that even if blindfolded, you are still able to determine the approximate size and properties of an area from the sonic reflections alone.

This is a result of the sound absorption properties from different materials. Every surface features a distinct frequency response, and various materials absorb sonic energy at differing frequencies. A bare brick wall will reflect high-frequency energy more readily than a wall covered in pictures or wallpaper. Similarly, in a large hall, it would take longer for the reverberations to decay than in a smaller room. In fact, the further away from a sound source you are the more reverberation you will experience. And, if you stand far enough away from the reflective surfaces, it would result in a series of distinct echoes.

While compression could be considered an essential processor, reverberation could be regarded as the most important effect. We experience reverberation regularly in our everyday life, so when it's missing, it is unnatural and unsettling. Recorded vocals, synthesizers, and many audio samples do not exhibit natural reverberation and it's common practice to add it artificially.

Artificial reverberation is available in two formats, convolution and synthetic. Convolution is considered by many to be the better option as it relies on *impulses* (samples of sonic reflections) of real spaces. Since these impulses are based on sonic fingerprints of real-world environments, they reproduce realistic acoustic spaces. But this comes at the expense of detailed customization while maintaining realism.

Synthetic reverberation employs algorithmic calculations to determine how specific acoustic spaces would react to a sound source. The sonic quality and realism offered by these styles of reverberation unit vary significantly from model to model. They can range from versatile lifelike recreations through to what can best be described as the sonic reflection you may experience from a cardboard box.

Reverberation also behaves differently depending on the sound source, the listener's distance from the source, the wall coverings, and any furnishings. Moreover, as reverberation plays an integral part in the human survival instinct, we are accustomed to how it sounds, and therefore many reverberation units offer a bewildering array of adjustable parameters to emulate this real-world response.

What follows is a list of the parameters you may come across on quality reverberation units. But depending on the unit and the algorithms employed only some of these may be available.

Ratio (sometimes labeled as mix)

The ratio determines the relationship between direct sound and amount of reverberation. If this is increased, there will be more reverberation than direct sound. If decreased, there will be more direct sound than reverberation. This is the primary parameter for adjusting how close to the original sound source the listener is. The higher the ratio (or mix), the further away the sound source will appear to the listener.

Pre-delay time

After a sound occurs, the time separation between the direct sound and the first reflections reaching the ears is known as the pre-delay. This parameter will be configured in milliseconds and determines the time differential between the original sound and the arrival of the very first sonic reflections.

This contributes to the effect of distance from a reflective surface. If the listener were standing in the middle of an empty room, the first sonic reflections would take longer to reach her ears than if she were positioned next to a reflective surface such as a wall.

Alongside emulating this response, pre-delay is also used to prevent the reverberation field from smothering the transient of an instrument. By delaying the first reflections, an instruments transient can pass through the effect unmolested. This prevents early reflections from suppressing transients resulting in an incoherent sound.

Early reflections

Early reflections offer a way to control the sonic properties of the first few reflections received by the listener. Because sound will reflect from a multitude of surfaces, this results in subtle differences between each subsequent reflection. With this parameter, it is possible to determine the type and style of surface the initial sound has reflected from.

Diffusion

Diffusion is a measure of how far the early reflections spread across the stereo image. The amount of stereo width associated with reflections is determined by how far away from the sound source the listener may be.

If the initial source is a distance from the listener, the stereo width of the reverberation will dissipate as it travels through the air. If the sound source or reflective objects are closer to the listener, reverberations will exhibit less stereo spread.

This parameter is often overlooked by many who wash a sound in stereo reverberation to push a sound into the background only to question why it doesn't sound correct in context with the rest of the instruments.

Density

After the early reflections, the source signal will continue to reflect off different surfaces. The further reflections will culminate to create a reverberant field. The size of this field can be adjusted via the density parameter.

Using this, it is possible to vary the number of reflections and how fast they should repeat. By increasing, the reflections become denser giving the impression of a more extensive reverberant field with more complex surface reflections.

Reverberation decay time

Decay determines the period the reverberation will take to decay. In large buildings, the reflections take longer to decay into silence than they do in smaller rooms.

By increasing the decay, you can efficiently increase the size of the 'room.' The amount of time it takes for the reverb to decay to 60 dB *below* the level of the direct sound is known as the RT60, and therefore some reverberation units may label this as the RT60.

This parameter should be used with caution because large reverberation tails can quickly evolve into a convoluted mess of frequencies. If a long decay time is employed on closely spaced notes, the subsequent reflections from previous notes may still be decaying when the next note starts. If this pattern is repeated, it will be subjected to more and more reflections, smothering transients and creating a melody with poor intelligibility.

This is not to suggest large settings should be avoided. A favorite technique is to overwhelm a timbre with a long decay but follow this with a noise gate. With a high threshold on the gate, as soon as the tone stops, the gate activates preventing the reverberation tail from washing over the following transients. The perception of heavy reverberation (without a tail) can make a sound appear huge in a mix.

HF and LF damping

The further reflections travel, the further high frequencies are absorbed by the surrounding air. Soft furnishings will also absorb higher frequencies so by reducing the high-frequency content (and the decay time) it's possible to create the impression that the sound source is occurring in a small enclosed area with soft furnishings.

Alternatively, by increasing the decay time and filtering less of the natural high-frequency content, the source will appear further away. Or by increasing the lower frequency damping, it is possible to emulate an ample open space. For instance, a sound occurring in a large cavernous area would result in low-frequency reflections and less high frequencies.

The strategy of employing reverberation realistically is to envisage a room and furnishings and proceed to replicate this with the parameters offered. It is difficult, if not impossible, to accurately recreate the natural response of a room precisely but a ballpark reconstruction is often acceptable.

Note that not all instruments should be treated with the same amount or style of reverberation and careful consideration must also be given to any number of factors including sound design to acoustic positioning.

In electronic dance music, the kick drum should remain at the forefront of a mix, but the producer may want some House-style strings towards the rear. While this could be accomplished with a reduction in volume, it may result in the strings disappearing behind other instruments. In this situation, slightly heavier reverberation applied to the strings could fool the listeners into perceiving that the strings are further away than the kick drum due to the stronger reverberation field surrounding them.

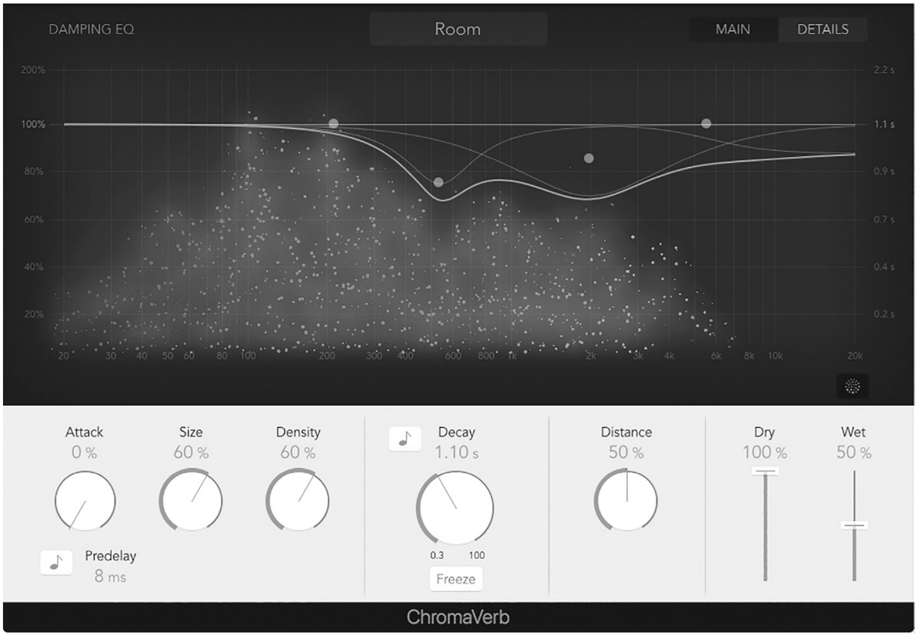

FIGURE 13.0
Logic Pro X Chromaverb

Similarly, the reverberant field resulting from any reverberation unit should not be considered the be all and end all. Further effects or processors following a reverberation unit can produce exciting or more realistic results. For example, it is not unusual to follow a reverberation with an EQ unit. This allows you to further sculpt a reverberation tail with EQ to ensure that the reverberation doesn't cloud the mix. Alternatively, reverberation may be applied to a channel and followed by a noise gate that has a keyed input from a secondary channel with rhythmic elements. This would produce a rhythmical reverberation. It's the experimentation and discovery of new techniques that remain central to any electronic music producer.

DIGITAL DELAY

Alongside reverberation, a digital delay is also one of the most critical and influential effects in the production of electronic dance music. It is not unusual for this, alongside reverberation, to find its way onto almost every channel in an arrangement.

Using these, it is possible to delay the incoming audio signal by a predetermined time. The delay time is adjusted in either milliseconds or via note values. The number of delays that are generated is termed the feedback. By increasing the feedback, it is possible to produce a multitude of repeats from a single sound.

A delay effect works by dividing the incoming signal into two channels and storing one of these channels for a short period before sending it to the output or by

FIGURE 13.1
The ultimate in digital delays – Sound Toys' Echoboy

feeding back through to the input. The storage and repeat cycle creates many discrete echoes with each subsequent repeat lower in amplitude than the previous. These echoes are often synchronized to the project tempo, matching both the resolution and grid of the current project and thus creating rhythmical effects.

While all delay units operate under this premise, many units offer far more parameters than adjustment of delay times. In many, you can delay the left and right channels individually and pan any subsequent delays to the left and right of the stereo image. Some units also offer pitch shifting of subsequent delays, employ filters to adjust the harmonic content of the delays, introduce distortion, apply groove adjustments, or offer an LFO as a modulation source.

The uses of these can be limitless within the production of electronic music. It's common practice to apply a delay to open and closed hi-hats to produce different rhythms or apply it to leads and basses to increase their groove. It's also used to generate its rhythmical effects, position sounds in large spaces, enhance the stereo image, adjust the placement of instruments, or, more simply, make a sound larger.

The last is known as *granular delay* and consists of setting delays to occur at less than 30ms. Settings at this speed produce what is known as the Haas effect. This is a psychoacoustic theory stating that any delay occurring within 40ms or less after the direct signal will not be perceived as a distinct event. The effect of delay times this short produces a timbre that appears thicker and wider. It is often employed on leads or basses to produce powerful driving grooves.

CHORUS

Chorus effects emulate the sound of two (or more) of the same instrument playing the same part simultaneously yet slightly out of time with one another. Since no two instrumentalists will play precisely in time, the result is a series of phase cancellations that can thicken the sound. This is analogous to two synthesizer waveforms slightly detuned and played simultaneously together; there will be a series of phase cancellations as the two frequencies move in and out of phase with one another.

A chorus unit achieves this effect by dividing the incoming signal into two channels. One channel is fed through an LFO modulated delay line before mixing back with the original signal at the output. By employing a very short delay of 40ms or less, the two channels blend into one at the output rather than resulting in a series of distinct echoes.

A chorus effect will typically offer three main parameters: intensity, rate, and mix. The intensity adjusts the modulation amount, as this is increased the magnitude of the effect is increased. The rate defines the frequency, or speed, of the LFO while the mix determines the balance between dry signal and the chorus effect. Some of the more capable chorus units may offer a selection of different LFO waveforms, although typically, a sine wave produces the most natural results.

It should be noted that because the modulation rate stays at a constant depth, frequency and the LFO waveform remains unchanging, it doesn't produce truly 'authentic' results. Because of this, it shouldn't be relied on to provide real-world vocal doubles or instrumental performances. Nonetheless, it has become a useful effect in its own right and is often employed to make oscillators and timbres appear thicker, broader, and more substantial.

FIGURE 13.2
A typical chorus effect

PHASERS AND FLANGERS

Phasers and flangers are similar effects but there are some subtle differences.

Flanging was introduced in the 1960s when engineers recorded and played back studio performances on two tape machines simultaneously so that, in the event one became damaged, they could switch to the second. It was discovered that if the engineer placed some pressure onto the 'flange' (the metal edge of the tape reel) for a moment, it resulted in a slowing down one of the tapes thus moving it out of sync resulting in the flanger effect. This was first employed on the 1967 song 'Itchycoo Park' by The Small Faces and, being a hit song, everyone quickly wanted to achieve the same effect.

Phasers are digital effects that divide the incoming signal into two copies and shift the phase of one before recombining it with the original signal again. This results in an effect similar to flanging but not as powerful.

With plug-in effects, both of these effects are produced by mixing the original incoming signal with a delayed version while also feeding some of the output back into the input. The difference between the two is that flangers use a time delay circuit to produce the effect while a phaser uses a phase shift circuit instead.

Both will employ an LFO to either modulate the phase (phaser) or the time (flanger). Both result in series of phase cancellations because the original and delayed signals are out of phase with one another. The result is that phasers

FIGURE 13.3
A flanger effect

FIGURE 13.4
A phaser effect

produce a series of notches in the audio file that are harmonically related (since they are associated with the phase of the original audio signal). Flangers, by way of contrast, exhibit an invariably different frequency because they employ a time delay circuit. Due to their similarities, both flangers and phasers will, however, share the same parameters. Both units also feature a rate parameter to control the speed of the LFO effect and feedback for the depth of the LFO modulation. Notably, some phasers will only use a sine wave as a modulation source, but flangers will offer many waveforms *and* delays. Of the two, flanging has become a staple in the production of electronic dance music.

DISTORTION

Distortion effects replicate the effects of analog and digital distortion. Typically, musicians are inclined to employ analog distortion units as these reproduce the sound of overdriven valves (tubes) and valve gear. Valves are highly regarded in many genres of music because of the second-order harmonic distortion they impart onto audio. The resulting sound is often termed as 'warm,' 'gooey,' or 'forward.'

Digital distortion is commonly a form of bit reduction. These give the impression that the number bits of an audio channel have been reduced significantly and produce a digital glitch style of distortion. This effect is often employed in dance music on drums to replicate the sound of early samplers and drum machines. These older hardware units were often 12-bit and had a very distinctive sound that is attributed to the early sound of dance music.

In the early distortion units, only two parameters were available, one for the amount of distortion and one to adjust the tone of the distortion (this was typically a filter). However, due to the subsequent importance of distortion effects in the production of dance music, these effects have developed further and now

FIGURE 13.5
Sound Toys' Decapitator distortion unit

offer everything from multiple bands of distortion through to emulations of distortion from overdriving classic analog hardware.

FILTERS

The simple filter is one of the most prominent effects of any modern studio producing electronic dance music. To create the atypical filter sweep that is present in just about every dance music record release in the past 25 years, you need a filter plug-in that can be inserted onto any (or all) audio tracks.

The action of a filter was discussed in detail in an earlier chapter, and a plug-in filter operates in the same manner. They commonly offer switchable low-pass, high-pass, bandpass and notch filters alongside resonance and sometimes distortion.

GLITCHES

'*Glitching*' is a technique that has been around for many years and is popular in genres such as *Complextro*, *Dubstep*, and *Future House*. This is a technique where audio is *sliced* into any number of smaller events and then randomly offset the events from one another to produce a glitch style effect.

Initially, this was a time-consuming process, so developers released plug-ins aimed at simplifying the process. Of particular note, *Sugar-Bytes Effectrix*, *Glitch*, *Vengeance Glitch Bitch* and *Cableguys Shaperbox* are effects units that divide incoming audio into individual steps of various lengths. You can then apply a multitude of effects to each. Typically, this involves distortion, modulation, shuffle, reverse, crusher, gates, delays, vinyl, and tape stop effects.

FIGURE 13.6
Moog multimode filter

FIGURE 13.7
Vengeance Glitch Bitch processor

There are many more effects available than discussed in this chapter. Everything from auto-panning, ring modulations, and pitch shifters are now possible as effects and developers are releasing new effects plug-ins on an almost daily basis. Many of these effects are based around those already discussed in this chapter and commonly consist of many effects chained together in various ways to produce different results.

And this is a fundamental production ethic within the production of electronic dance music. The arrangement and order of effects will provide different results. For example, a delay line employed after a reverberation will produce very different results to a reverberation placed after a delay line.

CHAPTER 14

The mixing desk

Technology presumes there's just one right way to do things and there never is.

Robert M. Pirsig

The purpose of any mixing desk, whether hardware or software representation within a DAW, is to combine any number of audio/instrument channels into a single stereo file. However, with a DAW's mixer offering limitless routing and bus options, the once simple mixer has developed into a sound design tool in its own right. This chapter will discuss the theory and application behind a mixing desk's internal bus structure along with processing and effects chains.

In a DAW, every time a channel is added to the project, an associated channel is created in the software mixer. Depending on the channel created, this respective mix channel may be stereo, mono, virtual instrument, MIDI, or bus. Except for the MIDI channel, all of these channels are designed to work with an audio signal and can access multiple *buses*.

A *bus* is a signal path that any number of signals can be routed to. Because a mixer's function is to combine all of the individual channels of a project into a single stereo file, every channel in a mixer will enter the mix bus. This is sometimes called the *main stereo bus* or the *2-bus* (because it consists of two buses, one for the left and one for the right channel). The mix bus carries all of the channels of the mixer, combining them into a left and right signal and routes them to the mixer's master fader. This master fader connects directly to the audio interfaces physical outputs.

DAW mixers feature a large number of buses permitting you to route signals in many different configurations. To understand this, we must first examine the layout and signal path of a typical audio channel.

Figure 14.1 shows the mix channel from Apple's Logic Pro X. The signal from a channel in the arrange page or from external hardware enters at the input stage of the mixing strip. It runs through the channel inserts (audio FX), into a series of auxiliary sends (sends), onto the channels pan (pan), and then into the mix fader before finally reaching the output. The output from this channel could be

FIGURE 14.0
The stereo mix bus

to the stereo mix bus or to external hardware (the output of which may enter a different mix channel). Although each DAW will display this layout differently, the path the signal takes will remain very similar.

An audio/instrument signal from the arrange page will first enter into the inserts of the mixing desk. Effects and processors can be inserted into a mix channels insert, but traditionally inserts are reserved for processors rather than effects. This is because processors are often required to process the entirety of an audio signal whereas an effect usually requires only a small percentage to be mixed with the audio signal.

By inserting a processor such as EQ into the mix, channels insert the signal will travel directly through the processor on route to the stereo mix bus. This means that the signal will move through one insert processor and then into the next and the next in a consecutive manner.

This means that if EQ were placed into the first insert and a compressor into the second, the EQ's *output* signal would enter the input of the compressor. In this configuration, if you were to adjust the EQ, it would modify the signal entering the compressor. This could result in a different set of frequencies triggering the compressor so you might have to adjust the compressor every time you made an EQ adjustment.

Alternatively, if a compressor were positioned *before* the EQ, the compressor could be used to control the dynamics of the signal entering the mix channel. But any EQ adjustments would not be subjected to dynamic control, and therefore any heavy-handed boosts may result in clipping the mixer. It's because of this continuous processing signal manner that careful consideration must be given in regards to the order of processors in a chain.

In many situations, decisions on the processor's order are based on a mix of creativity and thoughtful application. By placing EQ before the compressor, it

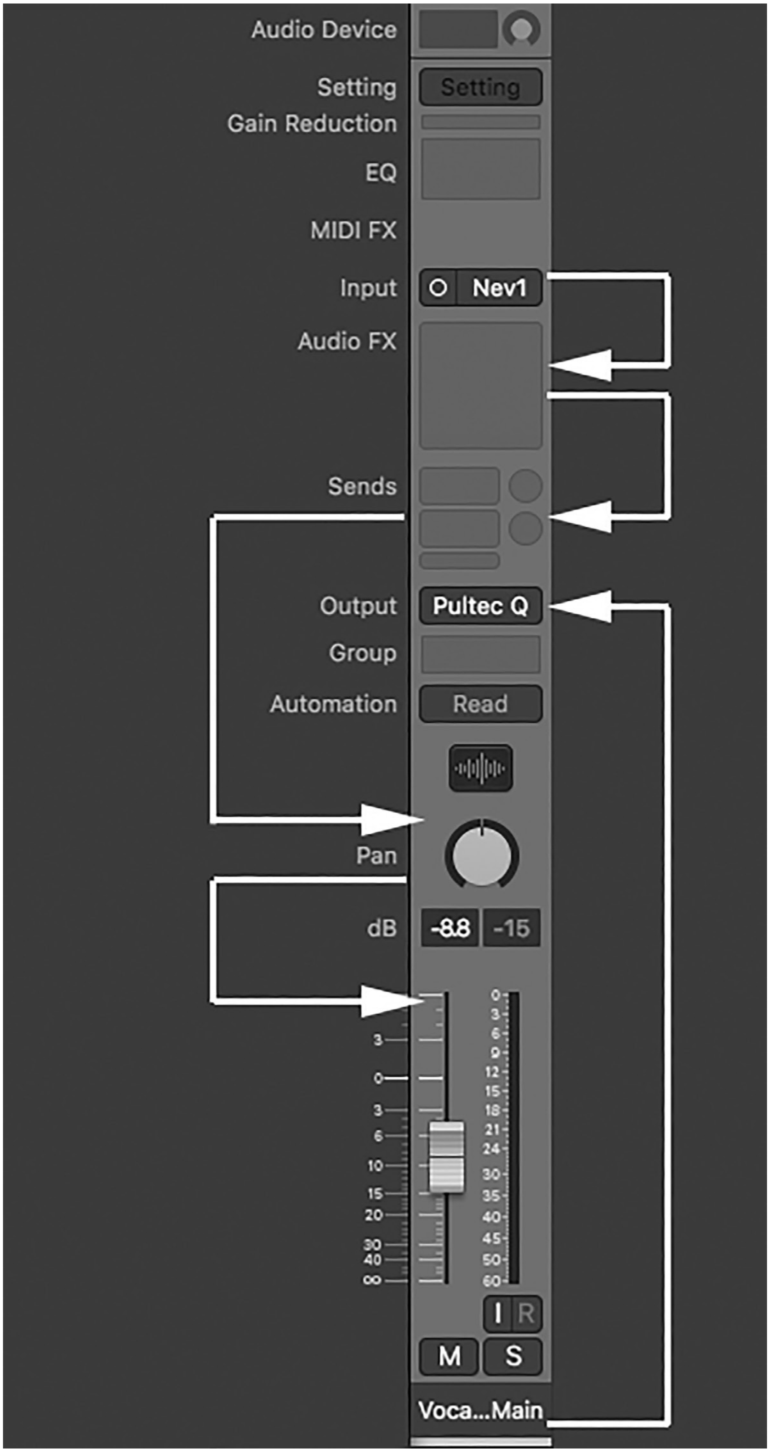

FIGURE 14.1
An audio track channel in Logic Pro

would be possible to create a frequency selective compressor. As the loudest frequencies breach the compressor's threshold, by employing EQ to increase quieter frequencies, it's possible to change the dynamic action of a signal.

PROCESSOR ORDERS

In a typical application, a noise gate is commonly the first processor inserted into a signal chain. This is because they can be used to remove any extraneous noise from the audio signal before further processing. If a compressor were placed before the gate, it would not only reduce the dynamic ratio between any noise and the signal you want to keep, but the reduced dynamic range would also make it difficult to program the gate.

Because a gate efficiently monitors the dynamic range by removing artifacts below a certain gain threshold, if the dynamics are reduced first it would be more problematic to program the gate to respond accurately. Consequently, the conventional application is a noise gate, followed by a compressor and then finally some EQ.

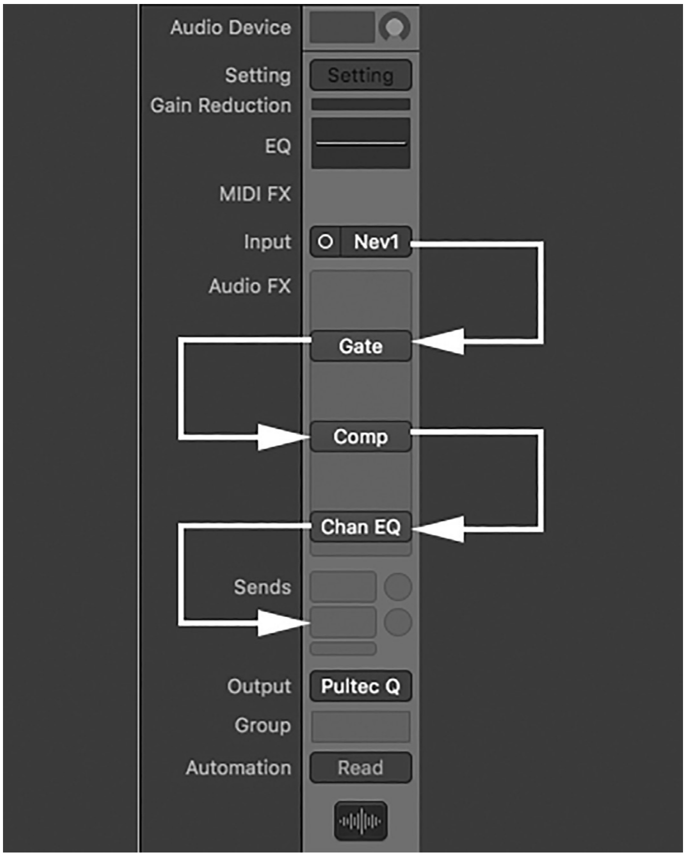

FIGURE 14.2
An effects chain (noise gate > compressor > EQ)

Further processing may be added in the form of effects. Although effects are not ideally suited to inserts, the first rule of music is that … there are no rules. Similarly to processing, however, the order of effects will determine a signals tonal modification. For example, if you were to place reverberation before distortion, the reverberation tails would be treated to distortion. But if reverberation were set afterward, the effect would not be as apparent.

Similarly, if a flanger were added to this set-up, the resulting sound would become more complicated. By placing the flanger after distortion, the flanger would comb filter the distorted signal producing a spaced-out phased effect. If the flanger were placed before distortion, the effect would vary the intensity of the distortion.

What's more, if the flanger were placed after distortion but before reverberation, the flange effect would contain some distorted frequencies but the reverberation tail would wash over the flanger diluting the effect but producing a reverberation that modulates as if it were controlled with an LFO.

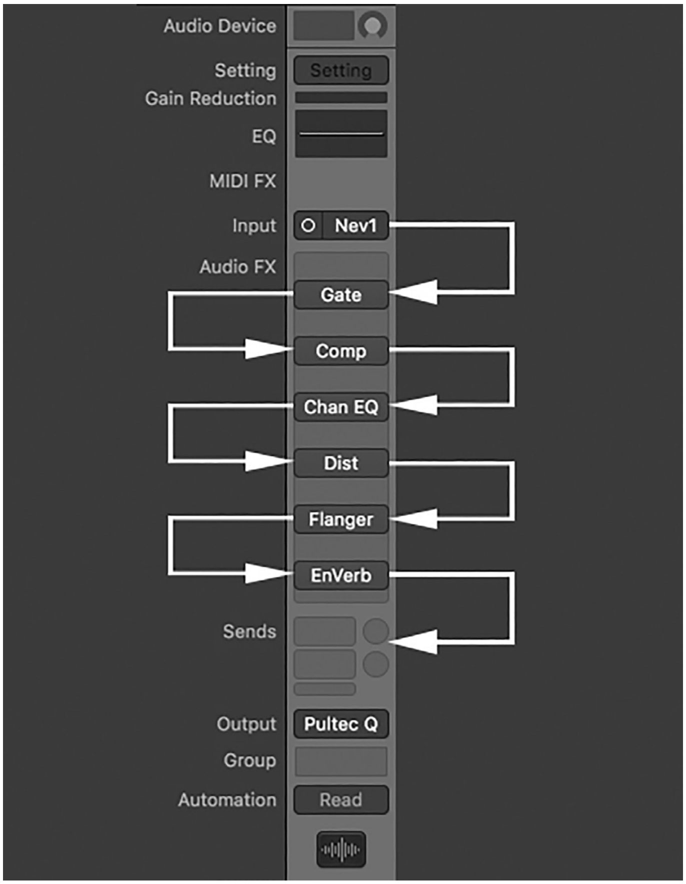

FIGURE 14.3
An effects chain (noise gate > compressor > EQ > distortion > flanger > reverb)

An alternative set-up could be to employ insert effects after the gate and compressor but *before* the EQ. By placing the EQ after insert effects, the EQ could be used to sculpt the tonal qualities produced by the effects.

For example, if the effect following compression were delay then distortion, the compressor would level out the dynamics creating a more noticeable distortion effect on the note decays. Additionally, since distortion will introduce more harmonics into the signal, they could then be modified with the EQ following the distortion.

Taking the above approach, EQ could be placed after each effect on an insert bus and then finally followed with another compressor. In this arrangement, the EQ could be employed to 'clean up' any annoying frequencies that are a result of an effect before it enters the next effect in the chain. By placing a compressor at

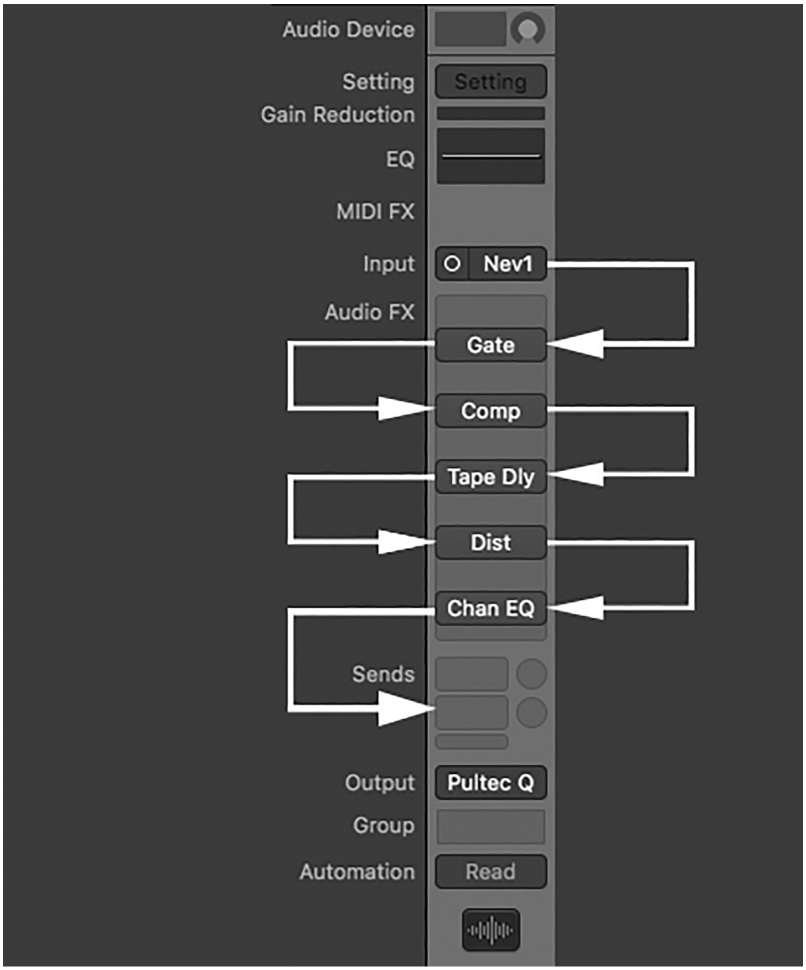

FIGURE 14.4
An effects chain (noise gate > compressor > delay > distortion > EQ)

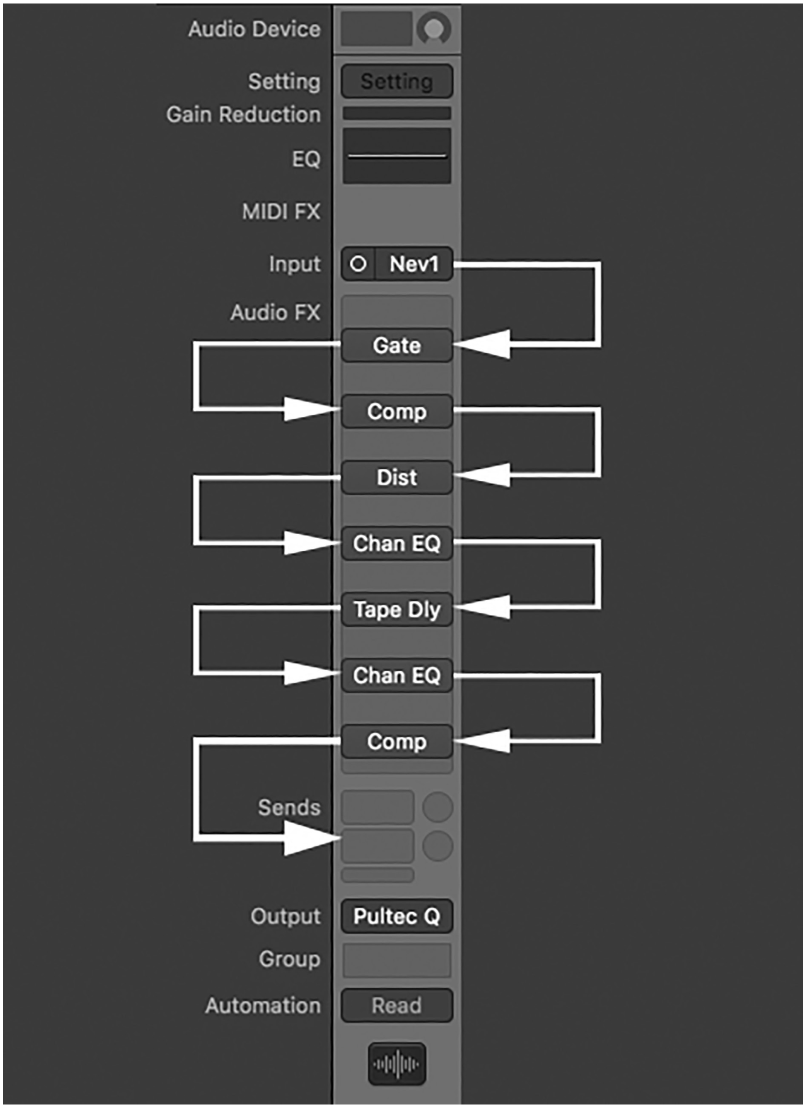

FIGURE 14.5
An effects chain (noise gate > compressor > distortion > EQ > delay > EQ > compressor)

the end of the chain, any peaking frequencies introduced by effects or EQ can be tamed with the compressor.

Different results can be attained by placing a gate after the effects, with a chain consisting of gate then compression followed by EQ, effects, and, finally, another gate. By placing the gate *after* the effects, the signal could be treated to a gate effect. Although there is a long list of possible uses for this, the most common technique is to apply a massive reverberation to a drum kick and then use the

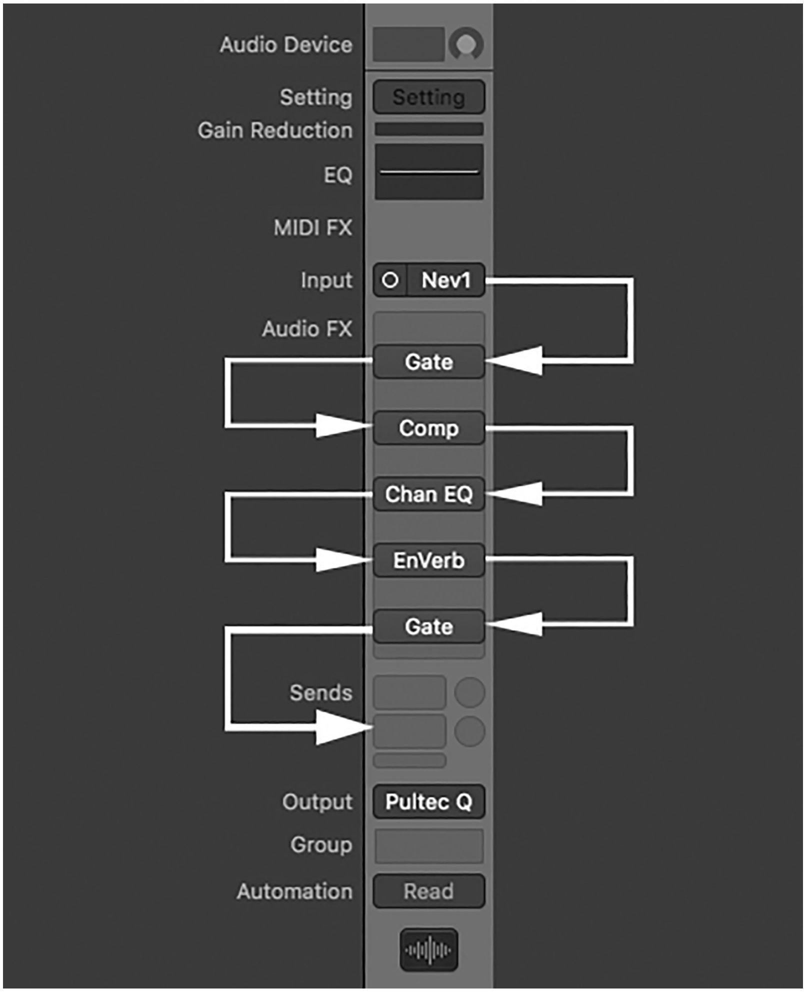

FIGURE 14.6
An effects chain (noise gate > compressor > EQ > reverb > noise gate)

following gate to remove the reverberations tail. This has the effect of thickening out the signal without the reverberation washing over the transients and blurring the image.

Although it is accepted that the compressor should come before effects, it can be equally useful to place it after. If chorus or flanger follows a filter effect with high resonance, it may result in clipping of the signal. By employing a compressor after these, it will control the dynamics before they're shaped tonally with the EQ. Placing compression after distortion will have little effect, however, since distortion effects naturally reduce the dynamic range and can even be used as a primitive compressor.

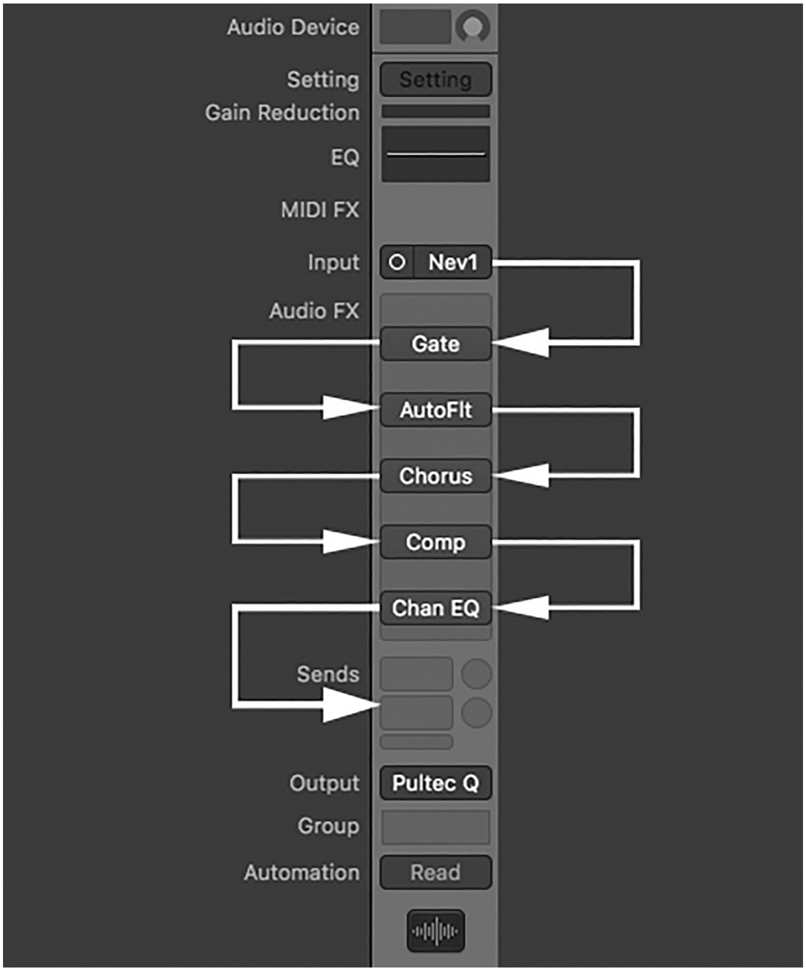

FIGURE 14.7
An effects chain (noise gate > filter > chorus > compressor > EQ)

AUXILIARY BUSES

The examples so far have consisted of effects *inserted* directly into the audio path, but in a more traditional application, they would be accessed through an auxiliary bus. This is because chaining effects through inserts result in the audio signal being processed entirely by the effect.

Many effects will feature a wet/dry parameter to control the relationship between the unprocessed dry signal and the wet effected signal, but they employ a single setting to accomplish this balance. This means that, as the wet level of the effect is increased, the dry level will decrease proportionally. Similarly, increasing the dry level will proportionally reduce the effects wet level. So, if you opted to thicken out a lead with reverberation, as the wetness factor is increased, the dry will decrease proportionally, which results in a wet reverberated sound. This can cause the transients to be washed over by reverberation, reducing the impact.

A preferable approach is to employ mix sends to access effects. Using sends, a percentage of a channels audio signal can be routed into an auxiliary bus and onto any number of auxiliary channels in the mixer.

As shown in Figure 14.8 auxiliary channels are the same as a standard audio channel featuring inserts, further sends, and a gain fader. The only difference is that they do not receive a signal from a channel in the arrangement and can only receive signals that are 'sent' to them via an auxiliary bus. In many DAWs, the moment you create a send bus in a channel, it will automatically create the subsequent auxiliary channel.

With the send parameter on the channel strip set at zero, the audio signal will ignore the send bus and move directly onto the gain fader. However, by increasing the send parameter, it is possible to determine how much of the channels signal is divided and routed to the auxiliary bus. The more the send parameter is increased, the more audio signal is also directed to the auxiliary bus.

The benefit of this approach is twofold. First, a channel's signal can be sent to an effects bus as well as continuing onto the channel's fader. This lets you maintain a dry level through the standard channel but also introduce some of the desired effects through the send bus channel. By adjusting the level of the direct signals channel *and* the auxiliary bus channels fader, it is possible to create a more crafted balance between the two.

What's more, every channel of the mix can have its signal sent down the *same* auxiliary bus. Therefore, every channel can access the same auxiliary bus. Frequent use for this is to employ a single reverberation on an auxiliary bus and then send numerous mix channels to this one bus. By doing so, every channel can access the same reverberation.

In this configuration, any effects inserted onto the auxiliary bus should have their wet/dry parameter configured to 100% wet. This is because you can adjust the amount of wet/dry balance using the audio channel's fader (dry sound) and the auxiliary channels fader (wet sound).

Auxiliary can function in either pre- or post-fader. In pre-fader, the signal is divided and delivered to the auxiliary bus *before* the channels gain fader. This means that if you were to reduce the direct sound fader, it would not reduce the amount of signal

FIGURE 14.8
An auxiliary channel in Logic Pro

FIGURE 14.9
Signal bus path

sent to the auxiliary bus. Therefore, if the channels direct sound fader were adjusted, the auxiliary channel may also need adjustment to balance the level between the two again. Pre-fader can, however, have creative applications. By inserting a reverberation on the auxiliary bus and sending pre-fader, you could reduce the direct sounds channel gain gradually resulting in a sound that gently fades away leaving only the reverberation behind.

For most applications, post-fader is the typical approach. Using this, reducing the direct channel's fader will also systematically lessen the auxiliary send, too, saving the need to continually adjust the auxiliary bus send level whenever the gain fader of the channel is changed.

Similar to the sequential arrangement of processors and effects on inserts, the same applies to an auxiliary bus. Because an auxiliary bus is a mixer channel that can only be accessed via sends, effects placed one after the other behave in the same manner as if inserted into a standard mixer channel. This means that the output of a previous effect will flow into the input of a following effect throughout the signal chain.

There are no rules that suggest effects should be used only on auxiliary tracks and processors used only as inserts. In fact, creative adaptation of mixing and matching both these approaches has resulted in some classic genre-defining effects. A favorite technique in dance music is parallel compression, whereby a compressor is placed on an auxiliary bus, and the direct channels signal is sent to the compressor via the auxiliary bus.

The direct channel is mixed in with the compressed signal on the auxiliary bus, and this produces a fatter sound because the compressor reduces the dynamics on the bus yet the original sound retains the peaks. Sometimes termed *New York compression*, this form of parallel compression can be employed on any instrument, but it is commonly applied to the rhythmical elements.

CHANNEL FADER AND SUBGROUPS

Once the signal passes the mixers send bus, it is routed to the fader for the channel. This gain fader does not amplify the channels signal but instead determines how much of the channel's signal is passed through to the stereo mix bus. This is why faders on many mixers are configured to 0 dB. They do not amplify any audio; they can only attenuate (i.e., weaken).

Aside from gain adjustments, you can also mute or solo the channel, adjust the panning of the channel across the left and right stereo spectrums, and route signals via subgroups rather than the stereo mix bus.

Subgroups

So far we have considered that every channel is routed directly into the main stereo mix bus where all sounds culminate at the stereo master fader. But it is possible to route any number of channels into a series of subgroups whereby the signals from the subgroups then enter the stereo mix bus.

A subgroup lets you send any number of channel outputs into a newly created channel. By doing so, the gain of all of these can be controlled universally with a single fader. A typical application for a subgroup is to route the kick, snare, claps, hi-hats and cymbal channels output all to the same subgroup. By adjusting each channel's individual gain – mixing down the drum elements – once they are routed to a single group track, the gain of the entire drum sub-mix can be adjusted with a single fader to suit the rest of the mix.

Because a subgroup channel shares all the functions of a standard audio channel, it is possible to apply any number of insert effects or access sends for the entire drum 'sub-mix.' For example, a compressor applied to the drum sub-mix could universally control the dynamics of all the drum elements. This would save on processing power within the host computer because there is no requirement to place a compressor on each drum channel. Alternatively, a compressor could be placed on every drum instrument *and* on the sub-mix to help 'gel' the instruments together.

A subgroup channel can be routed to the primary stereo bus of the mixer or can be further routed into a secondary subgroup. In fact, with most DAWs, you can have up to 128 buses that can be routed in any number of configurations. For example, a subgroup into subgroup into subgroup into subgroup, etc. Similarly, it is possible to send to an auxiliary bus and have that bus send to another auxiliary bus and then onto another bus and then into a subgroup, etc.

VCA

A similar principle to a subgroup is a VCA fader. These are fundamentally the same as a subgroup but only feature control over gain. There is no physical channel to insert effects or processor, and it is not possible to access sends. While this may appear unnecessary when compared to a subgroup, they can *spill* across a hardware controller.

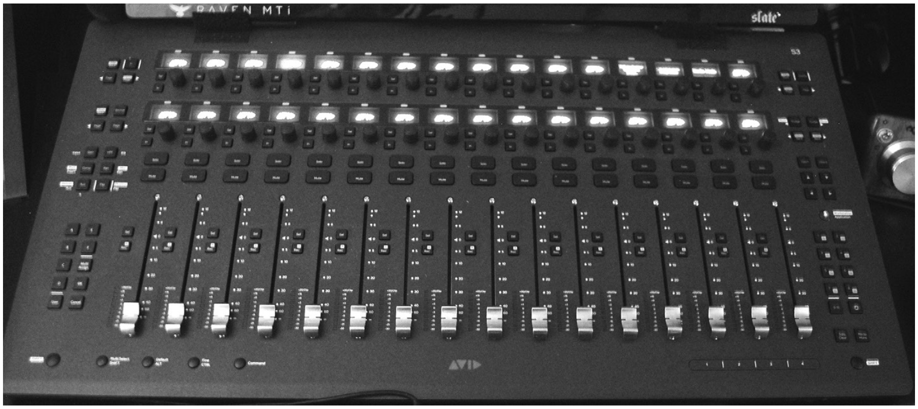

FIGURE 14.10
The author's Avid S3 controller

Many hardware controllers for hands-on mixing will feature a limited number of faders. This is typically limited to eight or 16 channels for reasons of cost or size. However, if you're mixing 30 channels of audio, you have to continually use *page* functions on the hardware to move through the different channels, and then you have to figure out which channel you're adjusting. To ease the workflow, you can group instruments into VCA channels.

For example, you may mix and then group all the drums into a single VCA, the bass instruments into another VCA, vocals, and harmonies into another and leads into another. By doing so, you can control the level of the drums, vocals, bass, and leads with just four faders, making mixing a more straightforward approach. However, if during the mix you hear a problem with the drum mix on the VCA, with the press of a single button you can *spill* all the drums across the hardware faders. This way, all of the individual drum channels open up on the interface so you can change their levels. And once done, you can then pack them back into the single VCA fader and continue mixing.

AUTOMATION

A final yet equally important part of software mixing is automation. With automation, every parameter, whether on an effect, processor or on the mixing desk itself, can be recorded and automated to move in real time during playback of the track. This style of automation has become a central element to the sound of many dance music records, allowing you to program everything from slow filter sweeps to sudden changes in effects.

Initially, mix automation was carried out by some engineers sitting by a mixing desk's faders and any connected effects. On playback, they would adjust the relevant parameters when they received the nod from the producer. The problem with the approach is that any parameter changes had to be performed flawlessly in one pass because the outputs of the desk were connected directly to a tape machine. In fact, this approach was the only option available back in the early 1990s. Producers and engineers used to sit by the filter and time the sweeps

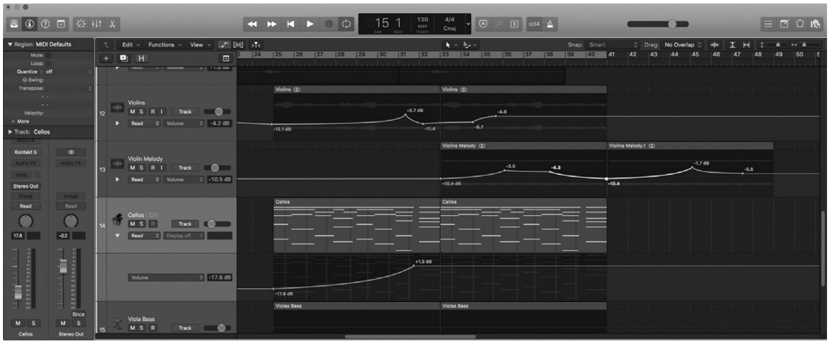

FIGURE 14.11
Mix automation

manually to fit with the music while recording the results directly to digital audio tape (DAT).

The development of the DAW has made this manual approach almost obsolete due to mixer automation. So much so that it isn't unusual for genres such as Techno or Tech House to have over 20 parameters changing simultaneously. And for the mixer and effects to jump to a whole new range of settings for different parts of the same track.

Mix automation is accomplished through the use of read, write, touch, and latch buttons located on the mixers channel. The creation of mix automation can be achieved in one of two ways: it can be drawn with a pencil tool, or the channel is set to *latch* mode and playback initiated. During playback, any parameters that are moved on that particular track, regardless of what they are, will be recorded live into the DAW.

The latter is the more common approach for many, since, after recording the automation, you can jump in and edit the automation with a pencil. This saves from having to search through a plethora of possible parameters in an attempt to find the parameter you want to edit.

Once automation has been recorded, this process can be repeated multiple times to record an unlimited number of automation parameters. After automation has been recorded and edited, the mixer is set into *read* mode. On playback, this replays all of the movements on the current track using the previously recorded automation data.

Although *latch* and *read* are the two most commonly used automation parameters, a further option, *touch*, can be used to edit any recorded automation data further. If *touch* mode is activated during playback, all currently recorded automated data will be read back as in read mode. However, if you adjust any parameter that was previously automated (such as a channel fader or effects parameter), the current movements will replace any automation data until the parameter is released again whereby the automation will continue along its originally recorded values. This makes touch mode useful for small live edits to previously recorded automation data.

CHAPTER 15

Hybrid studios

And now he's flanging the VU meters...

A&R, attempting to impress a client

While you can produce and mix a track with nothing more than a laptop and some carefully chosen software, in practice, this is rarely the case. Only a proportionately small number of professional artists produce their music within the confines of a computer. Many rely on a hybrid mix of both ***In The Box*** (ITB) and external analog synthesizers, processors, and effects.

Hardware instruments such as the Moog (pronounced *mogue*, like *vogue*), the Access Virus and various other synthesizers continue to sell in high numbers. Modular synthesis is on the uptake and processors such as the Empirical Lab's Distressor, Fatso, the Smart Research C2, API 525, Neve 1073, SSL G-Series Compressor, Tube-Tech CL1B, and Chandler TG-12413 are finding their way into many semi-professional studios. Moreover, the increasing popularity of hardware summing units such as the Dangerous 2 Bus, SSL Sigma, Neve 8816, and Neve Satellite show there is a continuing trend towards a mix of both software and hardware.

Indeed, many professionals reach a point where they understand it's a hybrid mix of both software and hardware that provides the most creative avenues and elevates their sonic potential. However, adding analog equipment consists of more than just plugging in some analog cables; it does require some knowledge of *gain structure*.

EARLY STUDIOS

Before audio workstations became the mainstay of a studio, producers relied on MIDI sequencers such as the Atari STE running Cubase or the Alesis MMT-8 hardware sequencer. These were connected via MIDI to various hardware synthesizers such as the Juno 106, TB303, and TR909, alongside multiple samplers such as the Roland S10, Akai S950, and Akai S1000.

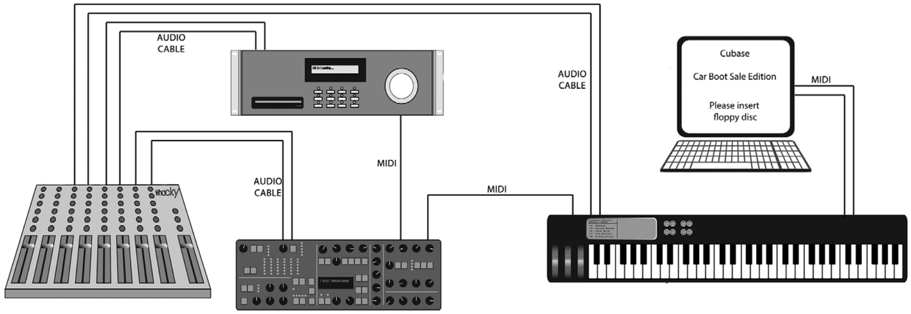

FIGURE 15.0
A typical serial configuration in a project studio (circa 1990s)

Finally, somewhere in the chain, there would be a few select processors such as a compressor or limiter, and usually a small number of guitar pedals. The audio outputs of all these devices were connected to one another in numerous configurations before they culminate into the mixing desk at the center of the studio.

With this set-up, some forethought must always be given to the signal gain. If the Juno 106 audio outputs were routed directly into a distortion guitar pedal and the output of the effects unit were routed into the S950 sampler and then into the mixing desks input, each device in the chain must have an appropriate gain. You must ensure that the output gain of each device in the chain is not too high, such that it overdrives the inputs of a receiving device, otherwise it produces distortion. But you must equally ensure that the output gain of each device is not too low, otherwise the noise generated by each device culminates through the chain and becomes noticeable.

To ensure a healthy ratio is maintained between the audio signal and unwanted noise, each device in the chain must have its gain finely managed. This fine-tuning of the output and input gain on each device is termed 'gain structure.' An appropriate gain structure ensures that you receive the best signal to noise ratio throughout the entire signal chain path.

Maintaining gain structure may appear inconsequential when working within a DAW. After all, there are no hardware devices to contend with and, therefore no noise. But, a common complaint among members of the dance music community is that music mixed entirely in a computer environment sounds 'flat' or lacks spatial resolution when compared to use of an analog console. This is because while the DAW may exhibit a higher dynamic range than hardware, many of the processors still emulate the behaviour of this equipment. This means that processors and effects can be overloaded, often without realizing, and this is one of the reasons for the lack of musical soundstage.

REFERENCE LEVELS

Maintaining a suitable gain structure in hardware and the DAW requires an understanding of signal levels and how meters are calibrated. Most audio devices feature some form of visual notification via a signal level meter that is calibrated in decibels. This informs you of the current outgoing or incoming signal levels (or, in some cases, both). For this to work, however, all the devices must share the same *reference level*. By doing so, the signal meters will all provide the same reading for the signal level regardless of who manufactured the device. If not, different equipment would give different interpretations.

For audio to travel through a cable, it must be converted into a low voltage. This voltage increases as the gain of the audio increases, and therefore we employ voltages as a reference for signal levels. Many years ago the reference level was milliwatts. From this, it was determined that if 1 milliwatt were traveling through the audio cable, the VU meter would read 0 dBm. This reference level signified that the audio signal was approximately 75 dB above the noise generated by the equipment.

If several devices were chained together and all used the reference level of dBm, you can adjust and maintain a suitable gain structure. With little more than watching the VU meters, you can ensure that the average signal level on each device remains at around 0 dBm. This results in an appropriate signal to noise ratio throughout all of the devices and produces the best sonic results.

As the years progressed, valves (tubes) appeared in studio equipment, but these require higher voltages. Milliwatts were no longer considered to be a suitable

FIGURE 15.1
A VU unit

reference level so engineers – using a convoluted mathematical theorem (shown below) – concluded that 0.775 volts should be the new reference level for audio equipment. This unique reference level became known as dBu.

(P = V2 / R (001W = V2 / 600 W (V2 = .001 W * 600 W
(V = sq (.001 W * 600 W)))

As more and more studio equipment began to rely on valves and the plates in the valves can be particularly noisy, engineers further boosted the voltage reference to 1.23 volts. This was to achieve the best signal to noise ratio between the equipment and meant that a device must receive 1.23 volts before it registers 0 VU, a reference level otherwise termed +4 dBu. This is the most common reference level in professional equipment today.

Since professional equipment employs a higher voltage, it is more expensive to manufacture, thus not all equipment adopts this standard. Other audio equipment that exhibits a lower noise floor such as hi-fi systems, CD players, many synthesizers, and other consumer-based studio equipment operate on a significantly lower voltage level of 0.316 volts. This equipment employs a reference level of 1 volt RMS (**R**oot **M**ean **S**quare – an overall average) and is termed as −10 dBV.

This creates an apparent signal level and meter difference between the two different formats. If −10 dBV equipment is connected directly to a +4 dBu input or vice versa, there is a significant voltage difference between the two, resulting in inaccurate readings on the signal meters. In fact, this difference is almost 4:1, resulting in an 11.8 dB difference between the two. In this situation, you can do little more than ignore the meters, listen carefully, and try to ensure the levels remain reasonable throughout the chain.

It does, however, introduce further problems with relation to the gain. While connecting devices with a +4 dBu output to a device with a −10 dBV input will result in a signal higher than anticipated, the gain can usually be reduced on the +4 dBu device to an acceptable level. But if a −10 dBV device (usually a synthesizer) is connected to a professional audio interface running at +4 dBu, the signal received at the audio interface is going to be 11.8 dB less than expected.

This requires the output of the synthesizer to be increased significantly to provide a healthy signal level. Some hardware synthesizers such as the Moog Voyager feature reliable output amplifiers with a low noise floor, and therefore output gain can be increased significantly without introducing low-level noise. However, many synthesizers don't have the available gain to provide an acceptable signal to a +4 dBu interface without increasing the noise floor.

In this situation, one workaround is to bridge the output of the synthesizer and input of the audio interface with a line amplifier. These are specialist amplifiers designed to increase a line level signal for input to a +4 dBu device. Alternatively, you could use a microphone pre-amplifier, many of these feature instrument inputs and can be used to increase the signal gain to a suitable level. If you use a quality, colored pre-amplifier for this, you can also add some interesting tonal character as the signal is amplified.

GAIN STRUCTURE IN AUDIO WORKSTATIONS

Of course, these reference levels do not exist within the digital domain of a DAW. There are no cables and, therefore, no voltages. There are, however, mathematical versions of this same procedure occurring. Whether recording channels through an audio interface or relying on samples and plug-in instruments, every channel is routed via the mix bus and summed within the DAW.

Many DAWs operate at 32- or 64-bit *floating point*. This increases the word length in the calculations and permits the DAW to access a significant amount of headroom to prevent overloading. Even with this extended headroom, however, the DAW has a limited way of representing the audio within a project. That is, sound within any digital device is rendered using a series of numbers, and these are not infinite: they end abruptly at 0dBFS. Termed *decibel full scale*, this is the DAWs version of +4 dBu, 0 dBm and 0 dBv levels. But unlike the analog domain, if a digital audio workstation is pushed beyond 0 dBFS, it results in unpleasant distortion while also reducing sound quality. This is because a DAW does not feature the same metering as analog devices and therefore the headroom is misinterpreted by many.

As previously discussed, signal meters in professional analog devices are commonly referenced in +4 dBu. That is, when 1.23 volts are traveling through a cable to the receiving device, it will register 0 VU, and this is considered to produce the most acceptable signal to noise ratio between two pieces of equipment. While applying processors, effects, and mixing, an engineer will continually maintain the gain structure with an average signal level of 0 VU for the body of the music. As further audio or processing is added to the mix, additional gain staging takes place. The engineer reduces the gain of individual channels, processors, effects, and the master fader to ensure that the average signal level remains around 0 VU.

However, 0 VU is *not* the uppermost limit of the devices in the analog domain. There is often another +18 to +20 dB of headroom *above* 0 VU on an analog desk. This is treated as additional headroom so that, on the odd occasion that the odd transient does slip by, it doesn't result in distortion. If this approach is compared to a digital audio sequencer, 0 dBu roughly translates to −18 dBFS, *not* 0 dBFS. It is not possible to convert a relative voltage to digital numeracy, and so it is only possible to determine results from the calibration of audio converters (the SSL Alpha MADI AX, in this case).

If you work in the DAW with the meters continually hovering around 0 dBFS, it is comparable to always working in the headroom of a mixing desk. Even on a high-end analog-mixing desk, mixing with levels so hot that they're 20 dB above 0 VU would produce a mix that sounds harsh and lacking in spatial resolution.

To further compound the issue, many plug-in effects and processors are modeled on hardware counterparts. They are designed to work at optimum at the equivalent 0 VU (−18 dBFS) level, not a signal that's being driven at 0 dBFS. By overdriving the plug-in effects, they are operating above their optimum programmed levels and can behave in unpredictable ways.

The common mistake for many producers is that they monitor the mix channel in the DAW post-fader. This displays the output level of the channel, *after* the fader. The problem with this approach is that you could be overdriving the channel *and* any effects or processors placed on it without knowing because you're monitoring the signal level *after* the mixer's fader.

For example, the output of many virtual synthesizers features a gain control. If you increase this, the signal from the synthesizer may enter the mixer channel at 0 dBFS (if it isn't already). If you then apply an EQ boost on the mix channel, the signal now exceeds 0 dBFS. You may follow this EQ with an analog-modeled compressor and an analog-modeled tube distortion unit, in this instance, the compressor and distortion are now receiving an overloaded signal.

Many professional analog-modeled units are designed to behave like their counterparts within a limited signal range. If this range is exceeded, they do not act in the same way, and it often compromises the tonal quality. However, you have no idea they're being overloaded because the DAW is only showing the signal level after the mix fader, not before. So in many situations, you should always monitor the signal pre-fader rather than post.

Gain structure is not just the domain of an analog hardware-based studio and has equal importance within the digital realm. By giving the math in the sequencers and the converters in the audio interface additional headroom, the bandwidth is not being driven hard, and this produces a more transparent, cleaner mix, with a stronger resolution.

CHAPTER 16

Fundamentals of rhythm

Music and rhythm find their way into secret places of the soul.

Plato

With the basics of audio engineering covered, we can now start to look at how to approach composing music, and for this, we should start with the fundamentals of rhythm.

Rhythm is the central focus of most genres of music but is particularly important in electronic dance music. However, it is also a concept that is more complex than it may first appear. For this chapter, we're going to examine the theory of tempo and time signature. But more importantly, we'll also investigate the various techniques that electronic music producers employ to create music that grooves.

TEMPO AND TIME SIGNATURES

Tempo and time signature are inseparably linked in music. Tempo is fundamental to all music and is a time-based measurement on the number of beats that occur every minute (aka **B**eats **P**er **M**inute – BPM). Whether this beat is perceptible as it is with the kick in electronic dance music or imperceptible – as it is with Classical – its purpose is to keep everyone in time.

Time signature informs musicians of how many beats occur per bar and the duration of each beat. Displayed similarly to a fraction with one number sitting atop another, the top number (the *numerator*) denotes how many beats occur within a measure (measure and bar are interchangeable terms). The lower number (the *denominator*) represents the size of note that is equal to a beat.

A bar can be viewed as a self-contained unit capable of containing one whole musical note (aka a *semibreve*). A semibreve always occurs for the length of the bar but if music consisted of nothing more than whole notes it wouldn't be particularly interesting, so semibreves can be divided into smaller notes. This process is identical to mathematical fractions; we can subdivide a whole note into numerous smaller elements but, collectively, they can only equal one whole.

For example, a semibreve could be divided in half so rather than play just one note for the duration of a bar, you could play two half notes (aka *minims*). Taking this a step further, one (or both) of these half notes could be subdivided again producing two quarter notes (aka *crotchets*). In this instance, a bar would now consist of two quarter notes and a half note.

Going further, crotchets could be subdivided further to produce two 1/8th notes (aka *quavers*). The bar would now consist of two eighth notes, one quarter note, and one half note. This style of subdivision could continue as far as required, employing a mix of different note sizes.

Alongside notes, we can introduce rests. This is the musical term for silence, but like notes, silence occurs for a finite period such as a quarter note, half note and so forth. The mix of notes and rests produces the rhythmical patterns that music relies on. Figure 16.0 shows this concept of the subdivision but in a more familiar setting for many electronic music producers.

Figure 16.0 is a DAW's piano roll editor set to the most common time signature of 4/4. With this signature, each bar is subdivided equally into four *beats*, and each beat is precisely one quarter note in length. These four subdivisions of the bar are displayed in the DAW's piano roll grid.

Just as it is possible to subdivide a note into any number of smaller divisions, it is equally possible to divide the DAW's grid into smaller divisions. These divisions are known as quantization values or pulses. When we subdivide the

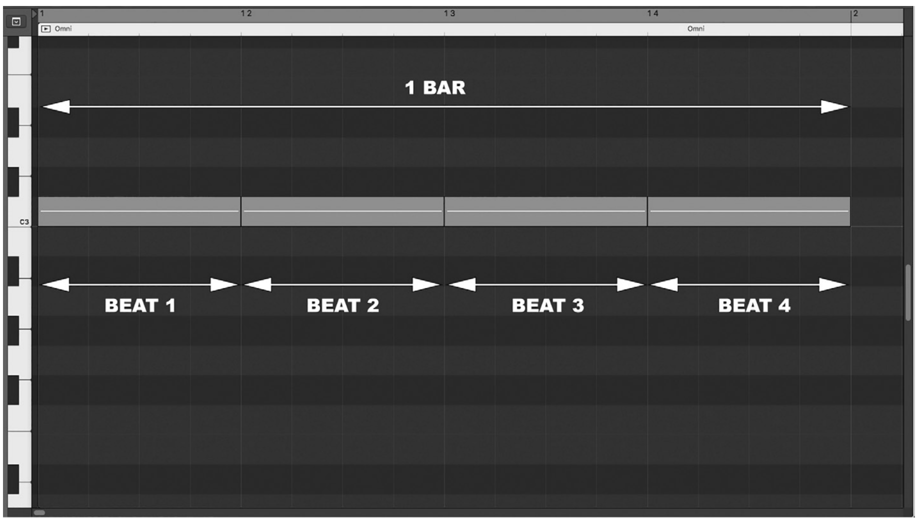

FIGURE 16.0
Logics piano roll set to 4/4 time signature

quantization grid in a DAW, it is not necessarily to determine the *size* of the notes, instead, it defines the *positions* in which we can place notes.

If the DAW's quantization grid were set to four pulses per bar, it would only be possible to position four notes in the bar. As pulses will always subdivide equally, it would mean one note at each quarter note pulse or beat of the bar.

If the DAW's quantization grid were increased to 16 pulses per bar, it would now be possible to place four notes *per beat* because four pulses x four beats = 16. If we were to subdivide this further, it is possible to create yet more pulses within the beat resulting in more positions to place musical notes.

The subdivision of a beat is known as *compounding the beat*. Figure 16.1 shows Logic Pro's piano roll and its resultant grid after compounding the beat a number of times.

This subdivision of beats is measured as the number of **P**ulses **P**er **Q**uarter **N**ote (PPQN). In the early 1990s, Roland concluded that a PPQN of 96 would be suitable for capturing human nuances, while a PPQN of 192 would be able to achieve the most delicate human 'feel.' Subsequently, most hardware-based step sequencers and drum machines of the time employed this resolution.

Most DAWs employ a PPQN of 960, which offers the possibility of subdividing a *beat* 960 times! At this range, it would produce a quantization grid so unnaturally small that you would require a microscope to view it. Consequently, many DAWs limit the grid to show a maximum of 64 evenly spaced pulses. Any movement further than this is accomplished by nudging a note backward and forwards to positions in between the displayed grid.

FIGURE 16.1
Compounding the beat

The process of compounding a beat can also work the same in reverse. Rather than subdivide a beat into two, it is possible to reduce the quantization so that two beats are amalgamated to produce the note equivalent (or positioning) of a half note. Here, there would only be two pulses available within the bar and therefore only two positions you could place a note within the bar. This process of turning two beats into one is known as *hyperbeat* because we're *crossing* over the beat. Both compounding and hyperbeat – by subdividing the quantization grid into larger or smaller pulses – lie at the very heart of producing electronic dance music.

RHYTHM IN DANCE MUSIC

Electronic dance music is based around a series of repeating rhythmical structures. But to prevent sending our listeners to sleep, we must employ some techniques to create more complex rhythmical structures. This is achieved by creating interlacing rhythmic textures of both chromatic (pitched) and non-chromatic (e.g., drum) instruments and forms the underlying structure of dance music. To make this effect, we divide both the beat and bars by equal and unequal amounts. This results in a series of asymmetrical pulses occurring in the quantize grid.

Let's begin by considering the structure of a house music drum rhythm. For the primary structure, we would compound each beat into four pulses. With a time signature of 4/4, this would result in 16 even pulses for the duration of the bar (four pulses per beat x four beats per bar = 16 pulses).

Figure 16.2 shows the quantization grid split into the 16 even pulses and the positioning of a typical basic house loop. The kick is positioned to occur on the *downbeats* (the first and third beat) and the *offbeats* (the second and fourth beats). On the DAW's grid, these positions occur at 1, 5, 9, and 13.

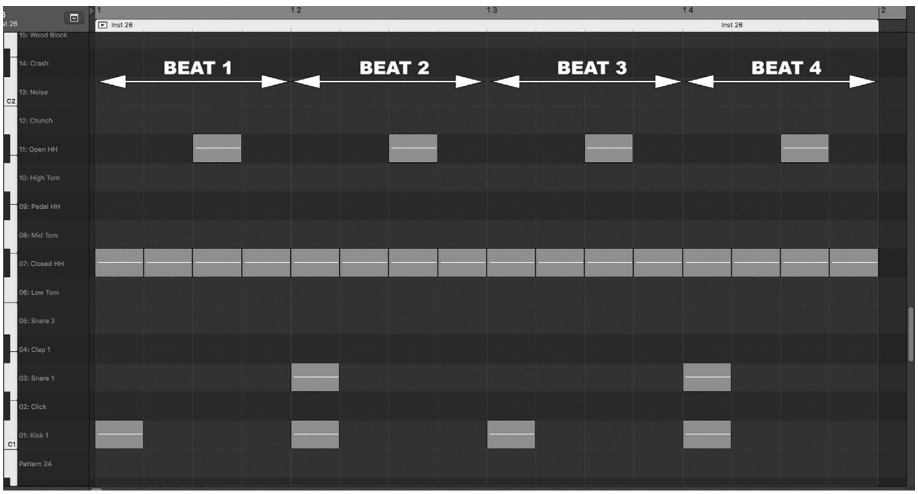

FIGURE 16.2
Basic positioning of the drums in EDM

A snare backs up the off beats at positions 5 and 13 while an open hi-hat is positioned on 3, 7, 10, and 15. Finally, a closed hat occurs on all of the 16 pulse positions. This produces an incredibly basic 'four to the floor' drum rhythm and is the fundamental structure to House, Tech House, Techno, Progressive, and most genres of dance music.

This alone would be unable to maintain sustained listening for an extended period. To permit sustainability for the track, further techniques must be introduced on top of this basic rhythm.

TRIPLETS

The most common technique is an effect known as *hemiola*. This is a triplet rhythm that, when played alongside the aforementioned even quantization grid, creates an effect known as *cross-meter modulation*. For this effect, we must change the DAW's quantization grid from an even value into an odd value.

Some DAWs aimed explicitly towards dance producers – such as Ableton Live – have a triplet grid that is accessed by right-clicking in the piano roll editor. Others such as Logic require the quantization value for the bar to be set at 1/12 rather than its default 1/16 value. With this quantization value, the length of the bar is subdivided unequally. Figure 16.3 shows the resulting grid of a triplet pattern.

With a triplet rhythm, there are three equal subdivisions per beat rather than four. Using this odd division of the bar, the pulses are offset, and it is now possible to place notes in positions that were not available when subdividing equally.

When notes are placed in these three positions the effect is a rhythm that is offset against the evenly distributed kick and snare. This produces a complex

FIGURE 16.3
Subdividing the bar into a triplet pattern

rhythmical interpolation that is fascinating to the ear. Consequently, this technique features on just about every dance music track in one respect or another.

DMM resource: An example of the triplet effect – here the triplet effect has been panned to the left so you can hear it.

Triplets can produce complex polyrhythmic combinations between different instruments. An example of this is known as 3:2 and is popular in genres such as Techno and Tech House. One rhythm of three notes plays over another rhythm composed of two notes.

This polyrhythmic technique can only be employed once at any one time. If various polyrhythms coincide then the music may appear confused or lacking in coherence. So, musicians will commonly develop on a single polyrhythm by introducing asymmetric phrasing.

ASYMMETRIC PHRASING

So far, we've considered subdividing the bar unequally to produce effects such as triplets. In this example, the individual notes do not sound concurrently between rhythms, but the length of the phrase remains constant.

In the basic House drum example discussed earlier, a 4/4 time signature resulted in four quarter notes per bar. A kick drum occurs on all four beats of the bar, and this produces an evenly spaced distribution of kicks. With hemiola introduced, it created an uneven subdivision of the bar where three notes occur to every four of the primary beat.

While this polyrhythmic technique adds interest to the rhythm, it is nevertheless unchanging. Due to the equal spacing of the kicks and the triplets in the bar, all the notes occur in the same position for the duration it's employed for. Listeners draw expectations that this pattern will continue and the rhythm begins to lose sustainability.

To prevent this, a standard technique is to maintain the triplet pattern over some bars and then reduce the triplet pattern from three notes to just one or two for some bars. This produces an asymmetrical phrase, a passage of music that is the same, yet different to those that preceded. The result is a *denial of expectation* in the listener providing further sustained listening.

DMM resource: An example of the asymmetrical triplet effect (panned left so you can hear the effect).

This effect is not limited to phrases and can also be employed in single bars by using notes of differing lengths. For example, if a 16 quantize grid subdivides

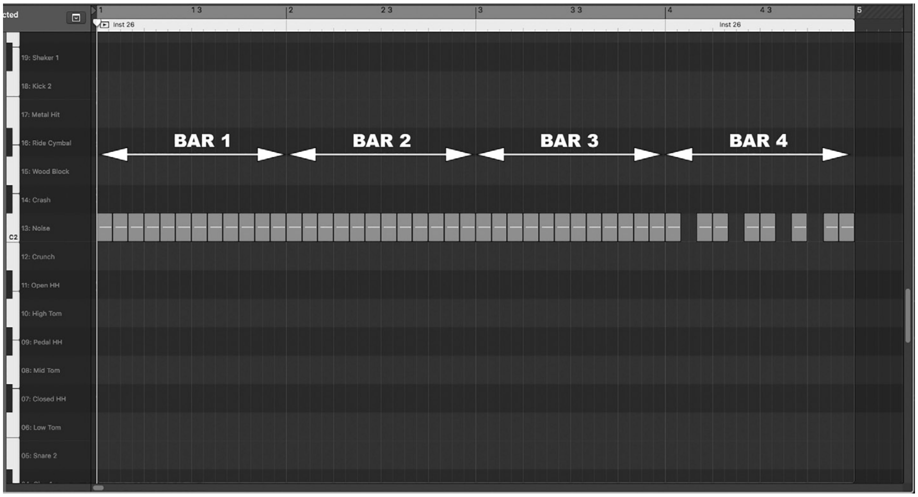

FIGURE 16.4
Asymmetrical subdivision over bars with triplet

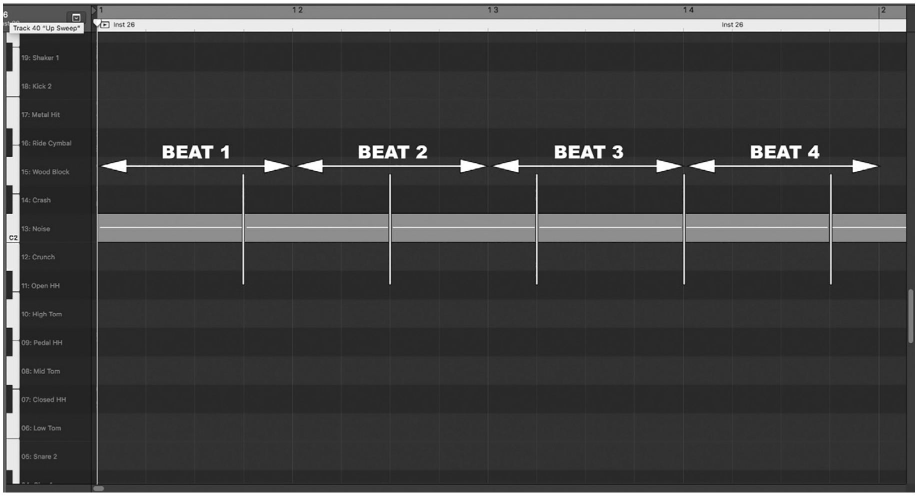

FIGURE 16.5
Asymmetrical subdivision of a bar

the bar, there is no requirement to place sounds symmetrically across this grid. Rather than equally divide the quantize grid and place sound on every second quantize event (2 + 2 + 2 + 2 + 2 + 2 + 2 + 2), sounds could be put on every third event. This would result in events of 3 + 3 + 3 + 3 + 4. Because of the asymmetrical subdivision of the grid, when played alongside a standard 4/4 kick drum, it introduces a skewed resolution to the melody. This style of approach is shown in Logic Pro's sequencer in Figure 16.5 and can be heard on the resource pack.

DMM resource: An example of the asymmetrical notes in the bar (panned left so you can hear the effect).

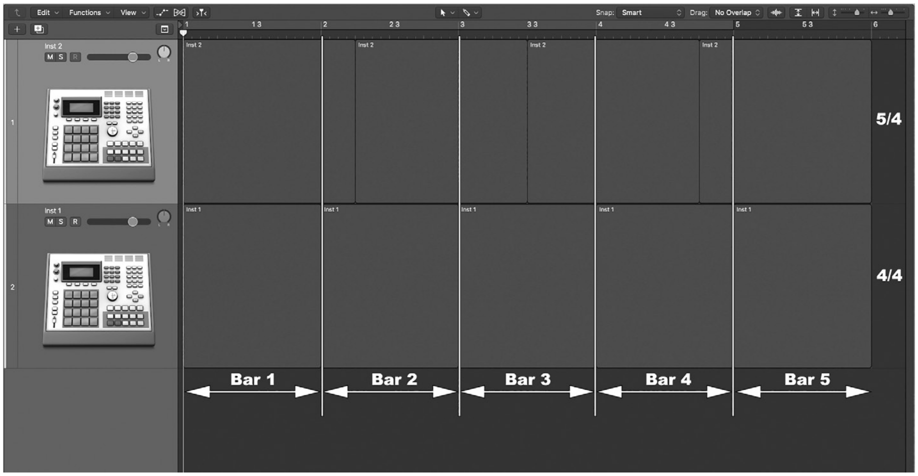

FIGURE 16.6
Polymeter

While these techniques aid in maintaining interest, to prolong repeated listening, they are combined with further techniques such as polymeter, syncopation, and swing.

Of these techniques, polymeter is perhaps the most evident and striking. It involves one or more simultaneous conflicting rhythms extending *beyond* a bar. This results in a rhythmical movement in and out of synchronization between different instrument channels.

This can be created by starting with a pattern or rhythm in a 4/4 time signature. Then, on the next DAW channel, employ a different time signature for the creation of the following rhythmical pattern. The alternative signature can be anything from 5/4 to 9/4. Using this metrical difference results in two rhythms of differing bar lengths.

For example, if 4/4 were used for the creation of one pattern and then 5/4 for a secondary pattern, the two patterns would be of a different length. They would move in and out of sync with one another over a specific number of bars. By employing this technique, the listener is offered two different rhythm signatures. This technique is shown in Figure 16.6.

SYNCOPATION

Further interest can be added to repeating patterns by employing syncopation. This is where the stress of the music occurs off the beat. All musical forms are based on patterns of strong and weak working together. In Classical music, it is the constant variation in these strong/weak dynamics that produces emotion and rhythm. Within electronic dance music, these dynamic variations are often accentuated by the rhythmical elements.

As touched on already, in 4/4 time signature the first and third beats of the bar are referred to as the *downbeats*. These will often receive the accent of the music because they are the likeliest place at which a musician would change chords or introduce a new instrument. Because of this, the notes that occur on these two beats are often louder than notes happening at any other position in the bar. In most forms of music, the first beat will receive the strongest accent while the third will receive a slightly weaker emphasis.

If this accent is switched and employed on a different note that occurs away from beats 3 and 4, it produces syncopation. By emphasizing notes that occur offbeat, using velocity commands, we can simulate the effect of some notes occurring more loudly than others.

Syncopation results in different levels of brightness and energy in the timbre. By changing this every few bars on rhythmical elements such as hi-hats or chromatic instruments, we can produce significant changes in a simple repeating rhythm. Indeed, syncopation happens to be one of the essential principles in the production of electronic dance music. The dynamic variation in timbres created by accenting notes that occur either on or off the beat is one of the leading contributing factors to the groove.

Alongside syncopation, timing differences also contribute towards sustained listening. By modifying the timing and dynamics of each instrument, we can inject groove into a recording. If the kick drum occurs on all four beats and a snare occurs on the second and fourth, simple actions such as moving the snare's timing forward or backward by milliseconds can make a significant difference to the feel of the rhythm section. Adjusting parts to occur slightly behind the beat creates a laid-back feel while positioning them to happen just ahead of the beat produces an intense surging feeling. These are the fundamental theories behind introducing swing into music and are one of the key factors to creating Techno, Minimal, Tech House, and House music.

The use of swing in electronic dance music is of vital importance and can be traced back to the early drum machines such as the Emu SP1200 and Akai MPC 60. As these machines operated on a strict quantize grid basis, the resultant rhythms were rigid, and swing could be employed to emulate the natural timing discrepancies of human players. By increasing the swing percentage, the drum hits are randomly moved by increasing amounts away from the quantize grid. The higher the swing value, the more the randomness occurs and the further off the grid the instruments are positioned.

All DAWs today, including Logic, Ableton, Reason, Cakewalk, Cubase, Pro Tools, and Studio One, feature a wealth of swing and quantize options. But it should be noted that each sequencer applies it differently and thus produces differing results. A swing quantize used in Cubase, for example, may sound different when the same settings are applied in Studio One. For many of the older dance musicians, the AKAI MPC units have always featured the best swing. However, in my experience of both these machines, I feel this to be based on bias and myth rather than fact.

CHAPTER 17

Kicks and percussion

At least with a drum machine you only punch the rhythm in once.

Rick Snoman

The central component in the creation of electronic dance music lies with the sonic design of the kick drum. This, along with the surrounding percussive instrumentation, forms the ubiquitous drum loop of almost all electronic dance music. Typically one to four bars in length, the drum loop provides the grounding element to not only the music but also the underlying timing and rhythm that listeners will dance to, too.

For this chapter, we'll examine the conventional approaches for designing the kick drum, claps, snares, and ancillary percussion. Then, in the next chapter, we'll explore how to combine these timbres to produce drum loops.

THE KICK

The kick drum is *the single most crucial* component of dance music today. More important than your bass, leads, or vocals, the kick underpins the entire groove and provides the ultimate rhythm for your listeners. Play a great kick drum in a nightclub, and the audience will dance to it, play them a poor one, or any other instrument, and the dance floor will empty. It is essential that your kick is right and so time spent designing the kick cannot be undervalued.

For many years, electronic dance music relied on the kick drum from the Roland TR808 and Roland TR909 drum machines. While the TR808 earned its place in Hip-Hop and some forms of House music, it was the TR909 that became the more pervasive machine, mostly due to its being the only machine of the two that featured MIDI connectivity.

Released in 1983, the **T**ransistor **R**hythm 909 (TR909) was a partial analog and sample-based drum machine with a 16-step sequencer. Introduced as a rhythm

FIGURE 17.0
The infamous Roland TR909 drum machine

accompaniment section for solo gigging musicians, the sounds were considered too artificial. The TR909 was soon dropped by many in favor of other rhythm composers available at the time.

The lack of interest resulted in a sharp decline in second-hand prices, and this landed them in the hands of many of the early dance music pioneers. Gaining popularity through the use by these pioneers, the TR909 soon formed the mandatory sound of dance music during the late 1980s and throughout the 1990s. Consequently, prices increased to epic proportions as musicians fought for the limited numbers available.

Much of the TR909's signature sound was a result of the engine behind the kick drum. By generating a kick drum through analog synthesis, you could control multiple parameters such as the tuning (pitch), decay, and attack stage.

Just these three parameters let you program a diverse range of kick drums and this is why it's the most sampled drum instrument of all time. The love affair with this drum machine and its predecessor – the TR808 – is still very much alive today. Many dance producers prize them highly, and despite the rerelease of these machines in hardware by Roland in 2017 (the TR-09 and the TR-08), and software plug-ins, the TR808 is more commonly used for bass in Techno, and the TR909 kick is rarely used at all.

In more recent years we've placed a more substantial focus on the kick drum. The current aim is to create a kick low enough to flap your trousers while also punching your listeners right in the chest. And the TR909 just isn't capable of producing either of these on its own.

BASS/KICK DRUM CREATION

With current production ethics, there are two approaches to the creation of a kick drum. You can rely on some samples that are layered together, or you can build one with synthesis. Both of these approaches are then processed to achieve the desired deep and heavy results. The choice of which to use is mostly down to the producer and their skill set, but both methods can produce the requisite kick timbres.

Layering kicks

The theory behind layering kicks may appear simple, but involves careful consideration, experienced listening, audio editing, and processing. The process begins by choosing three kick drum samples based on their different sonic characteristics. The first kick is the character kick. This is the kick that displays the overall tonal quality and shape you want to hear on the dance floor, while the other two kicks are used to augment this tone, attack, decay, or depth. For example, the second kick may exhibit a deep sub bass to enhance the shape of the character kick while the third may improve the transient or decay.

A common mistake made by novices is to layer numerous samples to produce a robust kick drum, but this will not work. Layering too many samples only results in a confused mash of frequencies that lack definition. Instead, kicks are created from a combination of just three *well-chosen* samples. The skill is in the ability to pick out samples that are best suited for the creation of the final kick drum, and this can only be accredited to experience born from practice.

Once the samples are chosen, they are each placed on separate audio channels in the DAW where you can zoom right into the waveforms to ensure that the phase and starting points of each kick line up accurately. This is a time-consuming process relying on both auditory and visual senses.

Examining the waveforms from each kick is essential to ensure that the phase is aligned. When the transient phase of all three kicks are aligned, it produces an increase of gain at that point. This can increase the presence and attack – and sometimes the body – of the kick. If, however, the phase is misaligned, this can result in phase inversion and cancellation of frequencies.

Both these can be beneficial to the overall timbre, but, at this early stage, the aim is to increase the gain rather than produce phase inversion. So much so, that if there are occurrences of phase inversion, the audio may be cut at that particular point and the sequencer's phase inversion tool used to invert the phase to ensure they all line up to increase the overall gain.

Once all of the kicks are successfully aligned, balancing, effects, and processing are applied while listening to the kick. Ideally, you need to balance the gain levels of each kick first to reach close to the sound you want. This is followed by judicious application of effects such as EQ, filters, transient designers, compression,

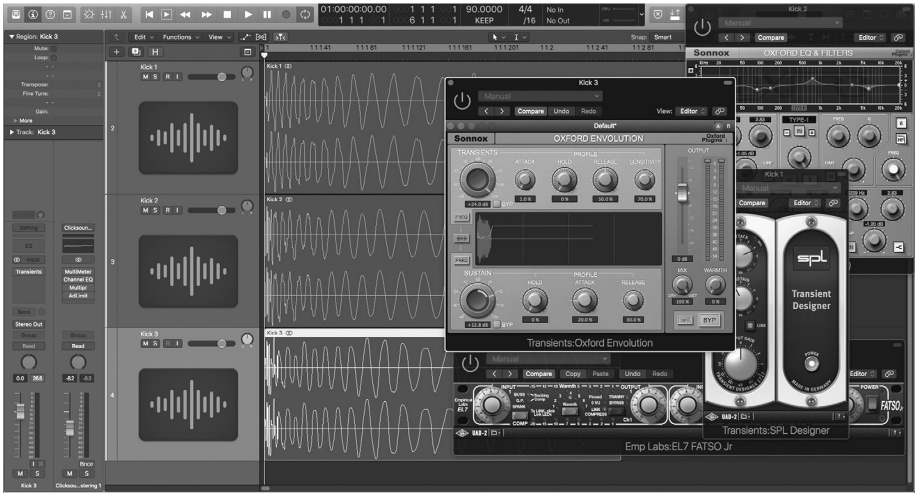

FIGURE 17.1 Layering kicks in the sequencer

distortion, tape distortion, and pitch processing. The choice of effects and processing is creative, but the goal is to produce a single coherent kick. This is born from practice, experimentation, and knowledge of effects and processing.

For example, if the body of one of the kick layers is distorting or incorrectly influencing the results, you could employ a low-pass or high-pass filter to reduce the frequency content of that particular kick channel. Alternatively, EQ could sculpt out chunks of an offending frequency to help all three kicks gel together and produce a coherent whole. EQ is rarely used to boost at this point, and should be avoided because it may produce signal overloads that distort the sound.

Individual kicks may be treated to a transient designer to modify the initial attack stage or may be bused to a group track where the transient designer is used to define the complete timbre. A compressor may be employed across the bus channel, set to a high ratio and low threshold with a fast attack to help all the individual channels gel together. Or a compressor may be placed over each kick on separate channels and used to model each kick to produce a more consistent overall timbre.

If the kick suffers from a substantial transient with little body, a compressor set with an extended attack to miss the initial transient but compress the body would result in a change of dynamic ratio between the kick's transient and its decay, in effect, increasing the gain of the body.

Similarly, you might want to apply some pitch processing to change the pitch of one kick, or, indeed all of the kicks, to produce a harder sound. This may help the different layer's gel or increase the body by lowering the pitch of a kick channel. Alternatively, careful application of distortion or analog style distortion may add further character to the kicks.

There are, of course, many processing and effects routes possible but these are a result of hours of experimentation and experience. A good understanding of the working principles of some effects and processors such as compression is obviously beneficial because there is no universally accepted practice for layering kicks.

It is your individuality, your ideas, that create the best results, so experimentation is the key. It is certainly not unusual to spend a good few hours building a great kick drum, and this should be doubled (or tripled) for the novice.

Once the layers bond well, it should be bounced into a single audio file. This audio is reimported into the current working project as a single kick drum sample.

DMM resource: The process of layering sampled kicks.

While the sample layering approach is undoubtedly a common technique for many producers, it can introduce some potential difficulties. Because dance music kicks are carefully engineered to induce bowel movement and/or thump your listeners in the chest and head, some feature a discernable pitch. This is particularly the case if the kick is being used as the bass, so if this pitch does not match the key of the music, it will result in a dissonant low-end groove. To prevent this, it's common practice to tune the pitch of the kick to the music.

Some producers argue against this form of tuning, believing that if a kick is in tune with the bass, it will result in a series of phase problems that reduce the energy of the bass. While this may have some standing with chart-based pop 'dance' music, it does not apply to genuine electronic dance music. The kick is the central constituent here, and the music is *always* composed around it. So much so, that to prevent frequency conflicts (that may reduce the impact of the kick), the bass will either not occur at the same time or if it does, the bass will be side-chained to move it out of the way of the kick.

If a kick is produced from layering samples, it should be tuned by ear because many pitch detection algorithms are incapable of detecting the pitch accurately. This is not beyond the capabilities of many producers, but you must obviously have a musical ear otherwise you may elevate the problem rather than solve it.

You can tune a kick with a pitch-shifting plug-in, or my preferred solution is to place it in a sampler and map it across a number of pitches. This way you can move the kick up and down in the piano roll editor until it falls into 'tune.' You should ideally have an acoustically treated room with a reliable monitoring system equipped with a subwoofer to do this. Standard monitors struggle to replicate the lowest frequencies of a kick drum accurately, so a subwoofer may be essential.

Synthesizing kicks

A more popular approach is to synthesize a kick. This offers more creative freedom than layering samples because not only can you specifically pitch the kick but you can also synthesize any parts you may need for further layering.

There are a good number of plug-in instruments available that are designed to facilitate synthesizing kicks, and almost all of them do the job well. As the plug-in market is continually developing, it would be futile to suggest any one in particular, but a quick search on Google will reveal the most commonly used.

If you don't want to use plug-ins, hardware instruments such as the Jomox Mbase01 and Mbase11 produce kicks with an incredible depth and body. From personal experience, these should be used with caution because they can tear the cones on a monitoring system! Both these units use analog synthesis and are only capable of modeling kick drums, but their capabilities can't be underestimated, and there is a good reason they are the mainstay of many studios. Of course, it is not necessary to use either a hardware or software specific drum synthesizers to create kick drums, and you can also create them in most capable synthesizers. But drum synths offer far more synthesis options targeted towards the creation of these timbres.

The theory behind synthesizing a kick remains the same regardless of the synthesizer. It consists of sweeping the pitch of a sine wave and then employing further techniques to add presence, body, and a harder attack stage.

In the original Roland TR909, the kick was created via a sine wave oscillator employing an attack/decay envelope controlling the speed of pitch decent. To increase the transient, a pulse and a noise waveform were combined and filtered internally to produce the initial attack stage. This process created the basic 909-kick timbre. Most drum synthesizers replicate this method, but some may let you change the sine wave to a square wave.

The first consideration should be the frequency of the sine wave because this will determine the depth and body of the kick. The fundamental frequency of a dance kick is commonly pitched between 40 to 80 Hz (E1 to E2 on a keyboard) depending on the genre, style of kick, and the scale you're working with. Possibly the best key to tune a kick to is A1 (55 Hz) because this is the fundamental frequency of a club system and kicks sound great in clubs when they hit this frequency.

An attack/decay envelope is used to negatively modulate the pitch of the sine wave so that the pitch descends rather than rises. All amplifier parameters are set to zero, and then the decay is slowly increased to produce the start of the timbre. Here, increasing the decay increases the time it takes for the pitch to fall resulting in a kick with a more substantial boom while decreasing will create a kick with a snappier feel.

On some synthesizers, the decay envelope will run in a linear fashion typical of the original TR909, but for today's production standards, this linear decay needs to be adjusted to behave exponentially. Indeed, many manufacturers of drum synthesizers today are well aware of the importance of modifying the decay potential and will permit you to alter this to modify the pitch characteristics of a kick. This is often accomplished by just clicking and dragging to modify the envelope's behavior.

By employing an exponential movement to the decay and controlling the speed of the pitch sweep, it's possible to create a much more comprehensive variety of kicks. For example, if the envelope were to bow outwards, as the pitch decays it will sweep more slowly at the beginning and begin to pick up speed as it closes towards the end of the kick.

Alternatively, if the exponential decay slope is inwards then the pitch will initially sweep faster at the beginning and slow down at it reaches the end of the decay portion. These two decay settings will provide very different results with the first producing a fatter, rounder kick while the second will create a timbre with a 'slapping' characteristic. By increasing the period of decay, these effects can be improved further to enhance the features of the kick.

If a synthesizer is not capable of adjusting the envelopes linear behavior, it is possible to create an exponential decay through other methods. For example, some synthesizers will modulate the decay parameter of the envelope with itself. By doing so, the modulation destination can be used to modulate both the oscillator's pitch and the decay parameter by negative or positive amounts, and this results in creating exponential behaviors in the decay. This effect is known as *recursive modulation*.

Alternatively, you can create exponential behavior using little more than a compressor. By inserting a compressor with a low threshold, high ratio, and a slow attack onto the kicks channel, the action of the compressor will increase the body of the kick resulting in an effect similar to 'bowing' the pitch. Similarly, using a fast attack on the compressor with a quick release, high ratio, and the threshold adjusted so that only the attack breaches the threshold of the compressor, it is possible to recreate the inwards style rate of decay. This approach should be considered the last option, and better results will often be attained through synthesis alone.

Alongside the pitch, many drum synthesizers enable you to modify the amplifier envelope. Using this you can make the kick appear shorter and snappier, or longer to exhibit more of a bass tone. The longer the bass tone, however, the more critical it is to tune it to the music!

When synthesizing, the initial timbre is rarely enough, and it is common practice to add clicks. These add more high energy to the transient of the kick to help it pull through busier mixes. Many drum synthesizers contain hundreds of samples of hi-hats and clicks to accomplish just this, and most will allow you to edit their pitch and amplitude to suit the kick.

If the synthesizer doesn't contain samples, then you can just layer a hi-hat over the kick. As strange as this may sound, layering a closed hi-hat over the kick and then employing a noise gate and transient designer to modify the dynamics will add a high transient to the kick allowing it to pull out of the mix.

Alternatively, they can be synthesized using either a sine or square wave by tuning them at double the frequency of the first kick. Using an immediate attack with a swift decay setting, you can produce a short transient click that can then be layered above the first kick.

More body or energy may be added to synthesized kicks by layering further synthesized kicks, samples, or, more commonly, a sine wave under the first kick. For many producers, a preferred synthesizer for creating and layering a sub bass is the Logic Pro X EX24 plug-in. Although this is a sampler, not a synthesizer, if no sample is loaded, it defaults to a sine wave that is perfect for drum synthesis.

FURTHER PROCESSING

Once the kick is produced, it will often benefit from further processing. Here, it can be treated with minimal amounts of reverberation. Although many publications will recommend avoiding reverberation on low-frequency instruments, they apparently haven't produced dance music.

Reverberation is almost *always* applied to the kick drum to add both character and body. This must be used with care because overzealous application will result in a kick that lacks definition and clarity. The kick is sent to a reverberation bus with EQ applied after the reverberation to sculpt the sonic signature. The settings for the reverberation and EQ depend on taste, but typically a small 'room' reverberation with a long pre-delay and very short tail is a great starting point.

The pre-delay should be long enough so that the initial transient is not molested by the first reflections and remains distinct so the kick can pull through the mix. For this, the pre-delay time will depend on the kick, but commonly settings between 15 and 30ms produce the best results. The reverberation tail is applied with a short time, typically less than 25ms. But a great technique is to set a long reverberation time of one second and then, while monitoring on headphones, gradually reduce the reverberation time until it's no longer noticeable. At this point, increase the reverberation time again by just a few milliseconds.

Alternatively, for a different style, some artists choose to employ a much longer – 500ms or more – tail but employ a noise gate directly after the reverberation. Using this, the reverberation is bypassed as an effect and the noise gate is adjusted so that the moment the kick ends the noise gate activates. This ensures that when the reverberation is engaged, there is no reverberation tail present and the reverberation itself only affects the body of the kick drum. After reverberation, more judicious compression may be applied. This is used as a finishing polish to help gel reverberation and kick, and create more equilibrium in gain level across the length of the kick.

For genres such as Techno and Tech House, reverberation on the kick forms part of the sound of the style. Here, the producers usually place a reverberation on an auxiliary send and send the kick signal into the reverberation. An EQ and a noise gate follow the auxiliary reverberation. The EQ is used to sculpt the reverberation signal to suit the artist's vision while the gate is used to prevent the tail from being too long.

DMM resource: The process of reverberation and compression on a kick.

SNARE AND CLAPS

A secondary instrument that shares a relationship to the kick is the snare or clap. Creativity dictates the choice of whether to use one or the other but genre considerations play a role in the decision. In some genres, claps are used while, in others, it's snares. Listening to the current market leaders should never be underestimated as it'll let you know what constitutes the genre and what's currently working on the dance floor.

The snare drums in the earliest forms of dance music were taken from the ubiquitous TR909, the E-mu Drumulator or the Roland SDS-5. All of these vintage machines employed a very similar synthesis method to create a snare drum by applying a triangle wave mixed with pitch-modulated pink or white noise. The clap and snare from the TR909 are still employed in music today although similar to the kick drum, and these are often layered with another snare.

With snares, two or three samples are commonly chosen, layered, aligned, phase inverted (if required) and followed with judicious use of compression, EQ, and transient designers to help the samples gel together in a cohesive whole. These do not all have to be the same instrument and often claps, and noise will be layered with snares.

Unlike kicks, layering various samples is challenging because snares exhibit a complex character. Consequently, layering only two samples and then synthesizing a final sample can often help in this respect and cut down a few days of sample editing into a few hours of programming. Although not always necessary, by using synthesis to create the third clap the timbre can be specially designed in time, frequency, and tone to help the other two-snare/clap samples merge more fluidly.

Snares can be programmed in most synthesizers, but, generally speaking, it is preferable to use drum specific synthesizers such as the Waldorf Attack as these feature parameters suited towards the creation of snare timbres.

A synthesized snare requires a minimum of two oscillators, for the first a triangle wave is used, with either pink or white noise employed for the second. The decision between pink or white noise is down to what you prefer, but generally, I prefer pink noise. This is because pink contains more low-frequency noise and produces a broader range of frequencies.

To produce the first snare timbre much of the low- to mid-frequency content of the noise should be removed via a high-pass, bandpass or notch filter. Notching out the middle frequencies will create a 'clean' style of snare timbre commonly employed in breaks, and minimal house while a bandpass can be applied to add a sonic crispness to the tone making it more suitable for tech house and techno style projects. Alternatively, employing a high-pass filter with a medium resonance setting can be used to create a more house or trance style timbre.

As with kick drum synthesis, the amplifier's EG (**E**nvelope **G**enerator) should employ a zero attack, sustain, and release while the decay can be used to control the length of the snare. Ideally, a different amplifier EG for both the noise and the triangle wave should be used so that the triangle wave can be kept short with a fast decay. The noise can be made to ring a little further by increasing its decay parameter. The more the noise decay is increased, the more atmospheric a snare will appear. It is often worth making the noise ring out for a little longer than the layered snares to help gel them together and end with a single tonal ring.

If two amplifier EGs are not available, then small amounts of reverberation can be used to merge and lengthen the timbre. Room-style settings are a favorite of many producers although euphoric and uplifting trance aficionados tend to favor hall settings for that larger more atmospheric offbeat strike.

In either case, the pre-delay is adjusted so that initial transient creeps through unmolested and the decay time is adjusted to between 250ms and 1000ms. Noise gates with a fast attack should follow this, and by employing a short hold time with a quick release, the threshold can be set to your taste. Low thresholds times create a more ambient feel to the snare while higher thresholds will cut the gate off earlier producing a snappy timbre.

If the snares are still not gelling, further tonal modification may be possible in the form of pitch envelopes. By employing positive or negative pitch modulation, the pitch of the synthesized snare can be forced to sweep through a broader range of frequencies that can help gel many samples together.

Alternatively, an LFO set to a sine or triangle wave could be employed to modulate the pitch, although the frequency of the LFO must be set by ear to ensure the pitch sweep occurs at an acceptable rate for gelling all three snares. After this, small amounts of compression can be applied so that only the decay of the synthesized hit is squashed (i.e., slow attack). This will bring up the decay in volume to help gel the samples together.

Claps

Claps follow the same layering principles as snares in that two or three samples are carefully chosen and then layered together. This mixed with proper sample alignment; phase inversion and careful application of compression, EQ, and effects all aid in the samples gelling together. Similarly, like the snare you are not restricted to claps only, often they will be layered with snares. The snare is used as the transient *bite* of the timbre while the clap produces the tail and noise.

Like the snare drum, a clap exhibits a complex sonic character and layering can be difficult. Therefore, an often used technique is to layer a clap and snare and then synthesize a third. Claps are perhaps the most difficult 'percussive' element to synthesize because they consist of a large number of 'snaps' all played in a rapid, sometimes pitch shifting, sequence.

In many of the older analog drum machines, claps were created from white noise passed through a high-pass filter and modulated with a sawtooth envelope before being treated with reverberation. This can be emulated on most synthesizers by employing a filter and amplifier envelope onto a white noise oscillator. Both envelopes should use a fast attack with no sustain or release and the decay utilized to set the length of the clap. The old analog style sawtooth envelope can be emulated by employing a sawtooth LFO to modulate the filter frequency and pitch of the timbre. Increasing or decreasing the LFO's frequency can change the sonic character of the clap significantly.

FURTHER PERCUSSION

Beyond kicks, snare, and claps most of the percussive instrumentation are collected from the multitude of sample content that is available on the market. These are inevitably in WAV and AIFF format so they can be imported into the workstations audio channel and processed.

I prefer to import these directly into a sampler, however. Here, they can be mapped across the octave and then pitched up and down to better suit the music. Additionally, you can use the sampler's envelopes, modulation engine, and filters to modify the percussive element to fit with your music. Beyond this, percussion can be treated to compression, transient designers, and EQ, through to the more common practices of employing reverberation and following the reverberation with a noise gate to remove the tail. That said, the ability to synthesize percussive elements should not be underestimated.

Many genres such as Minimal, Techno, Tech House, and Progressive House depend on any number of strange, evolving percussive sounds occurring within the rhythms. A basic understanding of how to program and synthesize these further elements can, therefore, be fundamental to the music. What follows is a run-through of how to synthesize some of the more common percussive elements. By developing on these, it should provide you with some ideas on how to program any number of percussive elements.

Hi-hats

The ability to program a hi-hat should be considered as fundamental as the ability to program a kick, snare, or clap, since there will often be an occasion where a hat is layered on top of a kick, snare, or clap to introduce a tighter transient stage.

In the original analog drum machines, hats were created through filtered white noise. This can be accomplished quickly in most synthesizers by employing a white noise oscillator with a high-pass filter. The filter should be adjusted so that the lower frequencies are rolled off leaving only, the higher frequency content. Both amplifier and filter envelopes should be customized to a fast attack with no sustain or release leaving the decay parameter to control the length of the hat. If the decay is set suitably long enough, the hat will change from a closed hat to an open hat.

Alongside employing analog synthesis to create a hi-hat, it's possible to use ring modulation or frequency modulation to accomplish the same thing. Creating a hat through ring modulation involves feeding a high- and low-frequency triangle wave into a ring modulator and then adjusting the pitch of each oscillator until it achieves the desired results. Again, the amplifier envelope should have a fast attack with no sustain or release permitting the decay parameter to determine the length of the hat.

Similarly, creating a hat through FM synthesis consists of modulating a sine wave with a high-frequency triangle. This will result in a high-frequency noise waveform that can be sculpted via the amplifier EG with the typical hi-hat settings. An advantage of employing FM is that a pitch modulation can be used to vary the timbre. Once the basic sound is constructed, you can adjust the oscillators pitch to achieve many different sonic styles.

Hats will also benefit from EQ, transient designers, and reverberation followed by a noise gate to sculpt the final timbre. Dynamic restriction via a compressor, however, should be avoided. Even if the attack parameter on the compressor is adjusted to bypass the attack stage, because the cycle is so short, many compressors will still capture the transient and remove the high-frequency detail. Consequently, compression is avoided on any high-frequency timbres such as hats and shakers.

Shakers

Shakers, like hi-hats, are more often than not gathered from the multitude of sample contents on the market but they can be synthesized in a very similar fashion to analog style hi-hats. Indeed, to program a shaker, you create a hi-hat by employing a white noise oscillator modulated by the filter and amplifier envelopes.

These employ a short attack no release or sustain so that the decay can be used to determine the length of the timbre. Here, the decay is lengthened considerably to create a tone that is longer than an open hi-hat that is then LFO modulated with a high-pass filter. Typically, a sine wave LFO with a high frequency and medium depth produces the more typical shaker effects but changing the LFO waveform and frequency can provide a multitude of different shaker effects. As always, this can then be further sculpted with EQ, transient designers, and reverberation followed with a noise gate.

Cymbals

Cymbals are created in a fashion similar to hi-hats and shakers. Again, they can be synthesized from a fundamental noise waveform, but a better alternative is to employ ring modulation or frequency modulation. Both these produce a much stronger and realistic resonance in the tail of the cymbal.

If ring modulation is available, two high-frequency square waves can be fed into the ring modulator and the output modulated with an amplifier envelope employing a fast attack and a medium decay (sustain and release should be set to 0). You then detune the two square waves from one another to run through a vast range of cymbal styles. Typically, detuning the oscillators two octaves from one another produces the most 'realistic' results but experimentation is the key, and many techno tracks have made use of three octaves or more.

If the synthesizer permits frequency modulation of oscillators other than sine waves (such as Native Instruments' FM8) a high-frequency square wave modulated by a low-frequency triangle oscillator can produce some cymbal effects. This can be further changed by adjusting the frequencies of both oscillators and increasing/decreasing the FM amount. What's more, modulating the pitch of one or both oscillators with a slow frequency sine wave LFO can create a crash cymbal that slowly changes during its cycle.

Tambourines

Tambourines, like claps, consist of some rapid successive hat strikes that are often modulated in pitch with a sawtooth envelope and have a band-pass filter applied. You can emulate this behavior by programming a basic shaker with a more prolonged decay but employing a bandpass filter rather than a high-pass.

A sawtooth LFO is then used to modulate both the bandpass filter and the pitch of the oscillator. For better results, the LFO's frequency should be modulated with another sine wave LFO so that the original LFO's frequency changes over time to produce more realistic results. Finally, you can adjust the bandpass to create a variety of different timbres. Here, full bandpass settings will recreate a tambourine with a sizeable tympanic membrane while thinner settings will recreate a tambourine with a smaller membrane.

Cowbells

If you need more cowbell, they can be formed in many different ways depending on the style required. For cowbells with a broader, more well-bodied sound you can use two square waves whereas if you want a brighter form, a square mixed with a triangle will produce lighter results.

For a cowbell with more body, both square oscillators should be detuned by approximately five tones from one another, so if one occurs at around C5 (554 Hz), the other should occur at G5 (830 Hz). The amplifier envelope should employ a fast attack with no release or sustain and a short decay, and a bandpass filter is then used to shape the overall tonal content of the timbre.

Alternatively, to produce a cowbell that exhibits a brighter color you can use a square wave mixed with a triangle wave. The frequency of the square can initially be set to around G5, and the triangle should be detuned so that it sits anywhere from half an octave to an octave away from the square. Both these oscillators should be fed into a ring modulator with the results run through a high pass filter to remove the lower frequencies. Once these basic timbres are created, the amplifiers decay can be lengthened or shortened to finalize the tone.

Congas

Congas are best created via frequency modulation synthesis by employing a sine wave and noise waveform. The sine oscillator's amplifier EG should use a fast attack and decay with no release or sustain. This will produce a basic clicking style timbre that can modulate the noise waveform via frequency modulation.

The noise waveform's amplifier EG should be set to a fast attack with zero release or sustain and the decay set to taste. By increasing the FM amount and detuning the frequency of the sine wave, it's possible to create some different styles of congas hits and strikes.

There is usually no requirement to employ a filter on the resulting sound but if it exhibits too much lower or upper energy a high- or low-pass filter can be applied to remove any offending frequencies. By employing a high-pass filter and reducing it slowly, it is possible to create any number of muted congas while adjusting the noise amplifiers decay slope to a non-linear function can produce the slapped style congas that often appear in Progressive House rhythms.

Tom drums

Tom drums are synthesized by employing the same methods as kick drums but instead utilize a higher pitch with a more prolonged decay on the amplifier's envelope generator. It's also advisable to mix in some white noise with the sine oscillator to produce some ambiance to the initial timbre and this can employ the same amplifier envelope as the original tom drum (zero attack, release, and sustain with a medium decay). An alternative method for creating a tom drum is to synthesize a common snare with a triangle and noise waveform but then modulate just the pitch of the noise with a slow triangle wave. This results in the sound sweeping down while leaving the triangle unmolested.

ANCILLARY PERCUSSION

Although here I have concentrated on the main elements in percussive dance loops, genres such as Tech House, Minimal, Techno, and Progressive rely on a large number of percussive style sounds that fall outside this category. However, despite the differences in timbre, they all follow the same basic programming principles laid out here.

For example, the primary oscillator will usually always consist of a sine, square, or triangle wave with its pitch modulated by a positive pitch envelope. This creates the initial tone of the timbre while a second oscillator is employed to generate the subsequent resonances of the percussion's skin or the initial transient. For recreating the effects of resonance, white or pink noise prove to be the most common options, while if the purpose is to create a transient, then a triangle or square wave is often used.

Both amplifier and filter envelope of the first oscillator is commonly set to a zero attack, zero release, and medium decay. This is so that the sound starts immediately on key press (the drummer's strike) while the decay controls how ambient the surrounding room is. If the decay is set quite long the sound will take longer to decay away producing an effect similar to reverberation on most instruments.

If the second oscillator is being used to create resonances to the first oscillator, then the amp and filter settings are always the same as the first oscillator. However, if it's being used to generate the transient, the same attack, release and sustain parameters are employed, but the decay is shorter.

For more creative applications, it's worth experimenting with the slope of the amplifier and filter's decay and attack parameters since nonlinear envelopes can produce strikingly different results on the same timbre. Also, by experimenting with frequency modulation and ring modulation, it's possible to create a host of new drum timbres. For instance, if the second oscillator is producing a noise waveform, this can be used to modulate the main oscillator to reduce the overall tone of the sound. What's more, both effects and processing chains shouldn't be underestimated. Many of the sounds in today's productions will be subjected to any number of processing and effects chains to add interest to the sound, and it's not unusual for the effects to be heavily modulated so that each strike is slightly different from the last to sustain repeated listening. This is one of the techniques we will discuss in more detail in the next chapter.

CHAPTER 18

Creating drum loops

1 ... 2 ... 3 ... 4 ... See, I'm already better than your f*cking drummer.

Anyone that has ever recorded a drummer

Drum loops form the very basis of all dance music genres, and are the foundation that the music will be built on. The problem is that creating a great drum rhythm is more difficult than it may appear. On casual listening, they can seem simplistic, but this belies a far more complex production ethic.

Merely throwing random drum samples into a DAW will often result in a flat, insipid, and uninspiring rhythm. To produce a professional-sounding rhythm section is a skill that requires close attention to timbre, effects, processing, timing, and modulation. Of course, drum loops are genre dependent. They are all composed differently concerning timbres and timing – and the second part of this book on genres discusses some of these approaches – but some fundamental techniques are employed within *all* professional drum loops that we will address here.

THE BASIC LOOP

The most common drum loop is the '*four to the floor*.' This is produced in a 4/4 time signature and features a kick/bass drum positioned on each beat of the bar alongside a snare or clap occurring on beats 2 and 4. Finally, a 16th closed hi-hat pattern and an open hi-hat positioned on every 1/8th, complete the underlying structure.

The synthesis of these percussive elements has been discussed in the previous chapter, but whether employing synthesis, samples, or a mix of both, it is essential to pay close attention to the timbres. Every percussive element will exhibit a different pitch, transient, sustain, dynamic behavior, and area (natural reverberation). These tonal and dynamic differences can make it difficult to produce a cohesive rhythm if they don't all match. This means that if you're employing samples you must be able to identify the most suitable candidates and – with

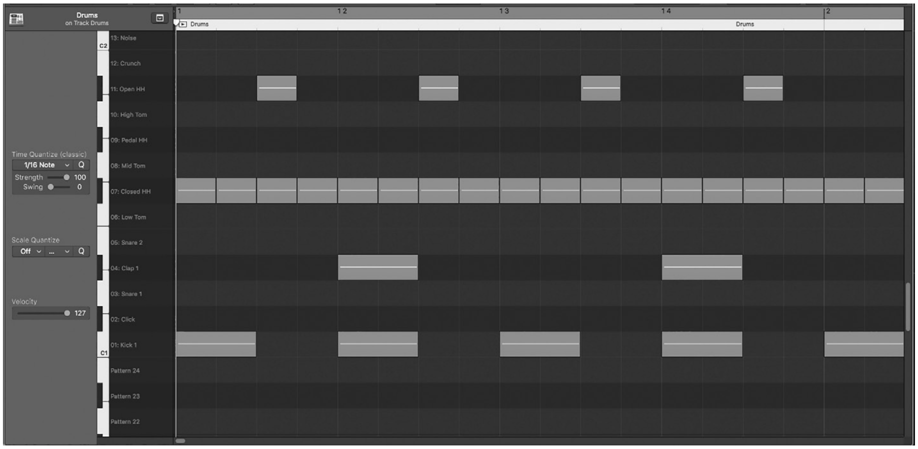

FIGURE 18.0
Logic Pro Audio displaying the atypical EDM drum arrangement (these are all contained in the same event for this image)

both samples and synthesis – employ some production techniques to create a unified rhythm.

Sample selection starts by auditioning them. Most sample content will consist of anywhere between 100 to 1000 individual percussive samples and surfing through these can be an arduous and particularly unmusical task. Therefore, rather than waste time auditioning every sample, it's preferable to pick the first 20 or so that pique your interest. Once you have 20 snares, claps, open, and closed hi-hats they can then be organized into *sample lanes*.

Sample lanes consist of employing multiple sampler instruments and mapping each batch of samples to different notes in each sampler. For example, one sampler will comprise of 20 different closed hi-hat samples allocated to the individual notes on the sampler's keyboard. Another will contain nothing but 20 snares mapped to different notes, another will hold claps, and yet another open hi-hat.

The kick is almost always laid down first. This, more often than not, is synthesized on a drum synth and already present on every beat of the bar. Following this, the snare/clap sampler is inserted on a secondary instrument channel, and a series of MIDI events are programmed into the piano roll to trigger a sample. When playback is initiated, you can then hear both the kick and snare playing together. By encircling the snare/clap MIDI events in the piano roll, these MIDI events can be moved via the piano roll, and by doing so, the connected sampler will play the next mapped snare/clap sample.

If this note movement is mapped to a keyboard shortcut, you can close your eyes and audition the samples and make a note of the relationship occurring between the kick and the different snare samples. Unless you're particularly lucky, it's unlikely that any samples will fit perfectly with the kick, but you will find a sample that is close and just requires further editing. Even if the sample appears to sit appropriately, time spent editing the sample is not wasted as it will likely help the instruments gel further.

The most influential processors for drum editing are:

- pitch
- transient
- EQ
- dynamics
- reverberation.

The first stage is to insert a pitch processor on the sampler channel and adjust the pitch in semitones. Most DAWs include a pitch processing plug-in and this provides a more natural method of changing pitch than in the sampler itself (because the notes are already mapped to different samples). Just this pitch processing can often result in a sudden cohesion with the other instruments in the loop.

Whether the pitch adjustments succeed or not, the next stage is to modify the transient and sustain portions of the sample. This can be accomplished in the sampler by employing its ADSR envelopes and filters. The action of increasing/decreasing the sustain and attack of a percussive element will profoundly influence its sonic character and its front to back perspective. This can also produce consistency in a loop, helping instruments sit in their correct perspective.

Following transient modification, it's not uncommon to employ some EQ. This is a particularly powerful processor for helping elements of a drum loop gel. The action of boosting or cutting a range of frequencies can affect sound significantly. Although the recommendations for EQ are to cut rather than boost, with electronic dance music it is not unusual to apply significant boosts and cuts to a timbre. This should be followed by compression to control the wayward dynamics resulting from heavy-handed frequency boosts.

Even if no EQ is applied, the sounds may benefit from compression. The typical application is to employ a long attack so that the compressor bypasses the transient but captures the tail. By compressing the tail, the dynamics are lessened, and this can aid with a cohesive sound.

The final effect to experiment with is reverberation. For a loop to sound cohesive, many of the sounds should appear to belong in the same space. If the snare, clap, or hats have been subjected to reverberation in the sample, a light application on the further percussive elements will help them gel into a cohesive whole.

Even if none has any reverberation applied, it's not uncommon to apply reverberation to all the percussive elements anyway. While numerous publications will warn you away from using reverberation on lower frequency elements, for dance music production it *is common* practice to apply small amounts of controlled reverberation to a kick.

This must be applied cautiously as too much will wash the kick out resulting in it losing its central focus in the mix. But if it's used lightly, it will help produce the sonic character heard in many professional records. A small room setting with a pre-delay of 20ms and a tail of less than 30ms will provide a good starting point, but this must be carefully monitored. As a rough guideline, the tail of the reverberation should have decayed entirely away before the next occurring 1/8th hi-hat to prevent the loss of focus.

An alternative technique is to employ a longer reverberation tail of 80 or 90ms but follow this with a noise gate. The gate is configured to close the moment the kick timbre ends. This permits the reverberation to add textural character to the kick but prevents the sound from becoming washed out. The noise gate must be set carefully, however. If the attack and release are too fast, some gates will track the individual cycles of the low frequency and produce distortion. Notably, some genres such as Techno, Minimal and Tech House will apply reverberation in a heavy-handed way and deliberately influence this distortion to add to the kick's character.

Snares and claps also benefit from a similar application of reverberation although often the reverberations tail is longer than the kick. The reverberation settings depend on the goal but should remain short so as not to wash over the loop entirely. Long sustained drum sounds or long reverberation will mask the silence between each of the percussive strikes, and this will considerably lessen the impact of the rhythm. The sudden dynamic adjustment from silence to transient on a snare/clap has a dramatic effect on a rhythmical loop. Too much sustain or reverberation will result in sloppy-sounding drums that lack groove.

A popular application in some genres is to apply small amounts of reverse reverberation on a snare or clap so that it draws up into the strike. This is accomplished by first duplicating a sampled clap or snare event and then reversing it in the workstations editor. Once reversed, reverberation is inserted into the track and applied with a short pre-delay and a tail of approximately 100 to 300ms. This effect is printed onto the audio (by bouncing the file to audio and reimporting it into the arrange page), and the audio data are reversed again, so they play the correct way around. The result of this exercise is a reverse reverberation effect that draws up into the transient of the snare or the clap.

Reverberation drawing into the note can be modified if required through the use of a transient designer, EQ, and the workstation's audio fade in/out parameters. The draw in should be kept relatively short so as not to smother the loop or change the appearance of the snare/claps timing. Indeed, the simple act of moving the timing (or employing effects that give the impression of timing shifts) of the clap or snare by just a few milliseconds can make a significant difference to the feel of the record and the groove in general. For example, if the snare/clap occurs slightly behind the beat, it will result in a laid-back feel while if it happens just ahead of the beat, it will often produce a more intense surging feeling.

> DMM resource: Reverberation and gate applied on the kick with reverse reverberation on the clap.

Both open and closed hi-hats are treated to small amounts of reverberation applied, but as with other instrumentation, this must be administered with caution. You *must* take into account the composition, tempo, and timbre of the hi-hats. For example, if the closed hi-hats occur on every 1/16th, a reverberation tail may wash over each consecutive hat transient resulting in an increasing reverberation field that will affect the dynamics of the loop. Consequently, if reverberation *is* to be applied to such a close-knit pattern, employ a noise gate or transient designer in serial after the reverberation so that it can remove the reverberations tail.

This problem rarely occurs with the open hi-hat due to their larger quantize spacing, but the reverberation tail should expire before the next open hat transient occurs. An approach here is to employ reverberation on a send bus and follow it with an EQ. By doing so, you can sculpt the reverberation, removing any low-frequency elements in the reverberations tail.

A common alternative to the application of reverberation on hi-hats is to treat them to small amounts of tempo-synced delay. The settings for the delay are dependent the tempo of the track and whether the open and closed hats are rhythmical or explicitly positioned at regular occurrences on the quantize grid.

Because delay can often wash over the hats with faster paced genres of music, it's worth employing side-chained compression on the delayed hi-hat signal. Here, the delay unit is placed on a bus and followed with a compressor set to a fast attack and release with a high ratio and low threshold. The compressor is then fed with the original hi-hat signal into its side-chain input, and the hi-hats are *sent* to this delay bus where a short to medium delay setting is employed.

By doing so, every time an original closed hi-hat occurs the compressor is triggered through its side-chain input. As the compressor triggers, it restricts the dynamics and lowers the volume of the delayed hi-hat. However, during the silence between the original hi-hats no compression is applied, and therefore the delay occurs at full volume.

This prevents the delay from washing over the hi-hats and creating a mush of frequencies. What's more, if you insert a noise gate onto the original hi-hat channel, the attack and release settings of the gate can be used to shorten the hi-hats. This will affect how the delay behaves producing a large variety of effects. The noise gate could then be automated throughout the track to deliver slow modulating results as the rhythm track progresses.

DMM resource: The effect of a noise gate on a hi-hat channel that is employing a side-chained delay.

GROOVE QUANTIZING

Swing, or shuffle, is essential to most drum loops for many genres. While some styles do require the rigidity of every note occurring dead on time, most styles benefit from some swing on the open hi-hats, closed hi-hats, or snares (or all three).

Swing is often overlooked when creating a dance drum loop because it's often wrongly attributed to the laid-back style of hip-hop. Although many of these tracks do employ swing (usually from the AKAI MPC), its application reaches further afield and finds its way into almost every genre of electronic dance music.

Swing can be applied automatically via a workstations swing quantize option. Here, you choose a percentage value up to 100%, and the amount of swing applied becomes relative to the current quantize setting. For example, 50% swing is considered *snap quantize* (no swing is being applied). Therefore, if the swing quantize value is set at 54% and the current quantize is set to 1/16th notes, a random selection of notes will all be moved later by 4% of a 1/16th. If this is adjusted to 46%, then a random collection of notes will be moved earlier by 4% of a 16th.

Unless you're deliberately aiming for a laid-back or syncopated feel to the music, quantize should be small enough so as not to be immediately noticeable and therefore settings of between 51% and 59% provide the required results.

Typically, the kick drum is left dead on the beat, and it is only the surrounding percussion that is treated to swing (claps, hats, and further percussion). This allows the kick to determine the exact timing and provides a context for the instruments that are affected by swing. In the example of the resource files, I applied a 9% swing over a 1/16th to the claps and open and closed hi-hat rhythms.

DMM resource: The effects of swing.

FILTER AND PITCH MODULATION

Alongside swing, drum hits should also exhibit varying volumes or slight tonal changes. This can be compared to a drummer striking the drums at random velocities and different places on the drum skin.

For this, it's common practice to employ filter modulation to many of the timbres. This involves microtonal pitch or filter modulation on the snares of claps so that the tones occurring on the second and fourth beats of the bar exhibit a small tonal difference from one another. This effect can be applied in the DAW's audio editor or via a modulating effect.

To apply a microtonal change in an audio workstation, duplicate the snare/clap event in the audio editor (to not affect all snares or claps in the project!) and then pitch this timbre up or down by a few cents. Around 10 to 20 cents usually works, but it should *not* exceed 30 cents. The purpose here is not to make the pitch movements between the two evident. Seven to 12 cents are considered to be just noticeable to our ears. Therefore, it is often best to remain below 30 to achieve this style of effect otherwise the drums will just sound out of tune.

An alternative to employing pitch shifting is to use a modulating filter effect. Sound Toys' *Filterfreak* is the preferred effect by many dance musicians as it offers a multitude of modulation options for the filter. The modulation is cyclically applied, so that first snare/clap of the bar passes through a low-pass filter unmolested but the second strike occurring on beat 4 is subjected to a very light low-pass filtering that removes a tiny portion of the higher harmonics of the timbre.

This effect introduces textural variance to the timbres. This is a crucial element to maintaining interest and focus because the mind naturally expects to hear some variation in musical tones, no matter how small. By employing these tiny fluctuations in texture and pitch, we pick up on the harmonic variance, and loops maintain sustained listening periods.

Of course, this modulation effect is not limited to filters or pitch, and almost any harmonic altering effect can be employed including distortion and bitrate reduction. However, whatever effect is applied it must be used in such a way that the harmonic differences between the strikes on the second and fourth beats are almost imperceptible. If they are applied too heavily, the groove will suffer.

DMM resource: Filter modulation applied to the snare.

FIGURE 18.1
Cyclic modulation applied via Sound Toys' FilterFreak

Alongside the snare/clap, the open and closed hi-hats can be treated to a very similar form of cyclic filter modulation. This can be applied via velocity to filter cut-off modulation but employing a filter is a faster and easier approach. This is modulated via an offset sine wave (it is possible to offset a sine wave modulation within *Filterfreak*), or via a sample and hold waveform. When this effect is applied *lightly* over the length of two bars, very slight harmonic changes occur over a period of two bars, and since the modulation source is offset or random, it prevents it from appearing as a cyclic repeated motion.

The result is that while the snares/clap are receiving a cyclic modulation every bar, the open hi-hats are receiving modulation every two bars and this creates a syncopated modulation between the two different timbres. This produces a pleasing effect to the ear.

Applying cyclic filter modulation to the closed hi-hats over three or six bars can further augment this effect. Three or six bars are chosen since as works against the standard structural downbeat of dance music and result in a cross-syncopation of modulation.

DMM resource: The effects of cyclic modulation applied to the snare, open and closed hats.

Once the basic rhythm is complete, further interest can be added to the loop by employing some displaced, syncopated hits. This approach features in *all* examples of dance music. Syncopation is a result of placing stress off the beat. Whereas typically this would involve changing the accent of notes or percussion that are already present, here the producer puts some percussive hits on unequal positions of the bar.

Figure 18.2 shows the typical 16th subdivision of the bar that is employed with many drum loops. In this configuration, all of the drum instrumentation is placed on an equal subdivision of this bar. The kick appears equally subdivided on position 1, 5, 9, and 13 (every four-step divisions), snares and claps occur on equally subdivided 5 and 13 (every eight-step divisions), and the closed hi-hat occurs on each of these 1/16th positions.

If, however, a percussive hit is placed on an unequal subdivision of the bar such as position 12 and 15, it results in rhythmical syncopation. This produces the effect that two different rhythms are playing.

In the example in Figure 18.3, I placed two short percussive strikes on position 12 and 15 in the bar and then subjected them to small amounts of delay followed by a small amount of subliminal modulation via a modulating low-pass filter.

This was set over a bar and modulated with an offset sine wave (in *Filterfreak*). To add more complexity to the two late hits, the delay was placed on a send

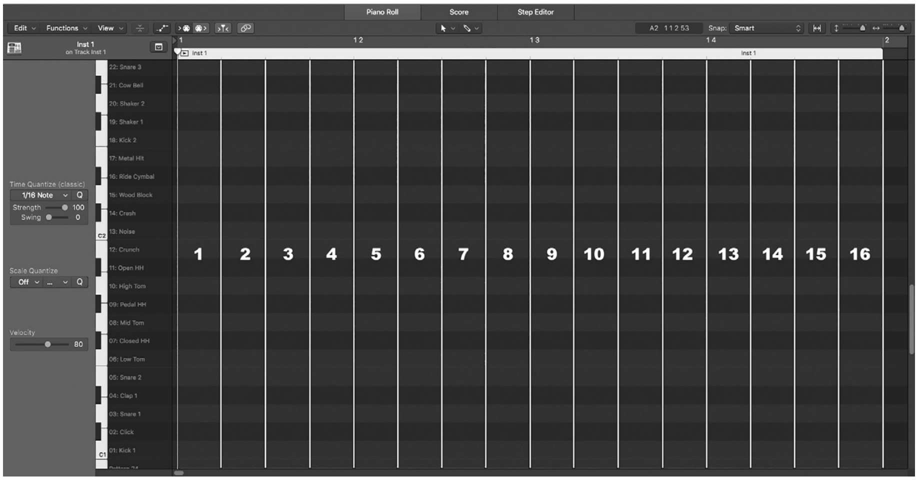

FIGURE 18.2
Equal subdivision of the bar

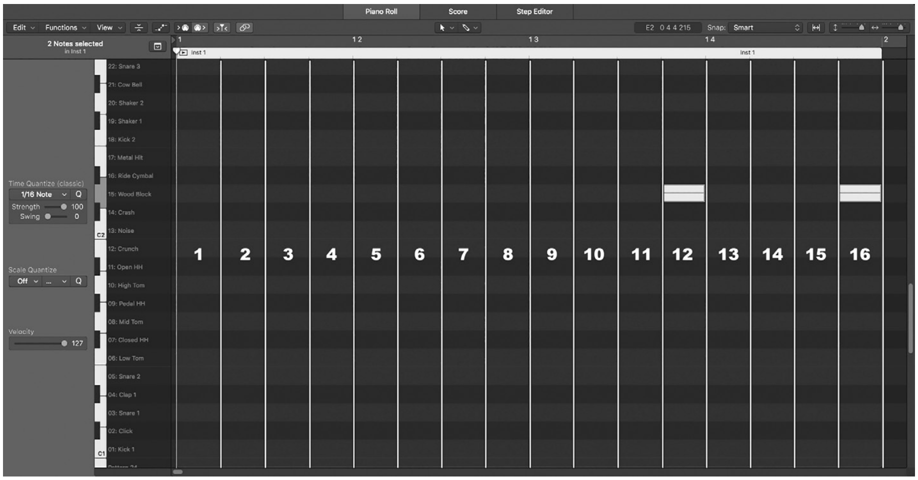

FIGURE 18.3
Hits positioned on unequal positions of the bar

channel, and EQ was applied after the delay. This results in the delayed signals sounding different again to the original percussive hits. Just this simple exercise of displaced positioning and delay treated to EQ results in a more complex rhythm that is difficult to disseminate.

Further to syncopated hits, triplet rhythms can be introduced at specific intervals during the drum rhythms. These are rarely applied throughout the entirety of a track rhythmical section since the effect can overcomplicate a steady beat but employed occasionally, they add excitement to rhythms. Known as hemiola, this again will be discussed in further chapters, but as a quick introduction, it produces the effect of cross-meter modulation.

For this effect, the workstation's quantization is adjusted to a triplet grid. This creates three equal subdivisions of the bar. By placing short percussive hits such as toms, congas, hi-hats, or, in the case of Techno and Tech House, synthesized percussive timbres or bass hits on these triplets, the pulses occur against the evenly distributed kick and snare rhythm resulting in a complex rhythmical interpolation.

In the sample you can hear in the example in Figure 18.4, I programmed some short percussive style timbres to sit on the triplet rhythm but changed the pitch of each micro-tonally so that the second note is –5 cents lower than the first and the third moves +3 cents above the second strike (making the third a total of –2 cents lower than the first note). Note that since this triplet effect clashed with the previously introduced syncopated strikes, I removed them.

This triplet can be further manipulated, and a conventional technique for this is to maintain this triplet pattern over some bars and then, after a fixed number of bars, reduce it to a single note for one bar. This *asymmetrical phrasing* creates a denial of expectation in the listener. The listener becomes accustomed to the triplet rhythm so when it is removed, it goes beyond their expectation and adds interest to the rhythm section.

Similarly, employing techniques such as polymeter can add further interest to drum rhythms throughout a track. Again, this will be discussed in more detail later but involves merely moving from the standard 4/4 to another such as 5/4 for just one percussive instrument. By doing so, it produces patterns of different lengths that run in and out of sync with one another over a specific number of bars. For this to work, the 5/4 time signature must employ a particular rhythmical pattern since if it consists of equally spaced transients, it would not be possible to determine the polymeter.

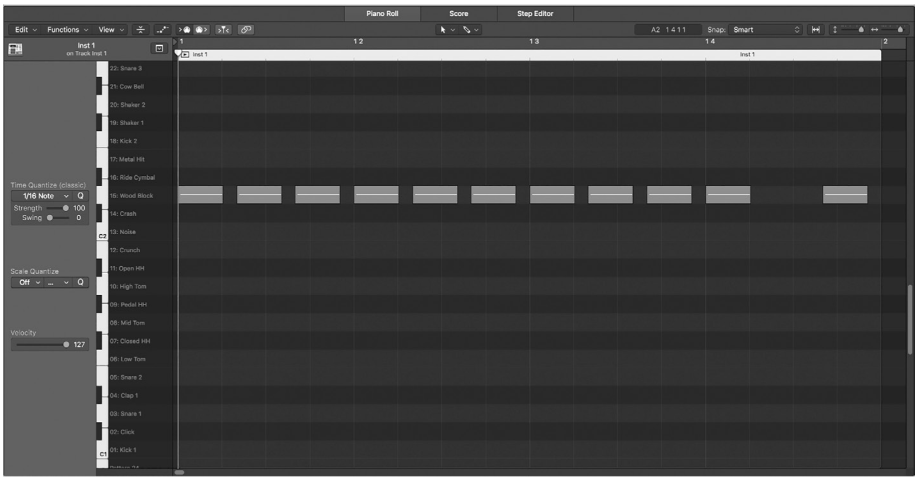

FIGURE 18.4
Hemiola

LOOP AUGMENTATION

An alternative to producing your drum loops from individual samples is to employ loop augmentation. Here, rather than use individual samples, entire drum loops are imported into the DAW. With many loops present, the producer removes different sounds from the loops, allowing the loops to mix. For example, you may remove the snares from one loop and the kicks from another, so the result is a mix of the two loops. You have the kicks from one loop and the snares from another. This technique is employed on anywhere from four to 10 drum loops to produce an entirely new rhythm. This form of loop augmentation forms the very basis of today's Tech House and Techno genres.

PARALLEL COMPRESSION

A final technique applied to the rhythm section is parallel compression. Sometimes termed 'New York compression' because it was a widespread effect in New York, it's a form of *upward* compression.

A compressors action is usually *downward*. It will restrict the dynamics of signals that breach the threshold while leaving those below the threshold unmolested. With upward compression, the opposite occurs. The quieter signals are increased in gain while what would typically be restricted remain the same. This effect is accomplished by sending the rhythm component tracks to a single group track so that they can all be controlled via a single mix fader. A compressor is then inserted onto a bus, and the group track is sent to the bus by a small amount of compression.

By doing so, both the uncompressed group signal and the compressed signal can be mixed at the desk. This way, the transient peaks of the original group track remains, but the low-level detail of the compressed track is increased (due to the compressor reducing the dynamic range) and produces a thicker, defined rhythm without increasing the transients.

The settings to use on the parallel compressor depend on the loop, but good starting points are a ratio of 3:1 with a fast attack and an autorelease (or a release of around 200ms). With these basic settings, the threshold should be lowered until it registers approximately 5 to 8 dB of gain reduction. Once this is set, either the bus fader or the compressors makeup gain can be increased until it achieves the desired effect.

DMM resource: The effects of parallel compression on a drum loop.

Compression can also be used to produce a crunchy distortion suited towards Techno and Tech House. Here, two compressors are employed in a serial configuration. The drum loop runs into the first compressor set to a high ratio and low threshold mixed with a fast attack and release. If the output gain of this compressor is pushed high enough, it results in distortion of the midrange that adds to the character to the signal. By then feeding this distorted signal into a second compressor, the distortion can be controlled to prevent it from clipping the outputs of the workstation.

Here we have only touched on the atypical four to the floor rhythms, but the same techniques can be, and often are, applied to *any* style of rhythmical percussive loop. Regardless of the rhythmical timing of any of the instrumentation these same techniques – if employed carefully – will breathe life and action into the loop, creating a rhythm with a more polished professional sound while also helping to maintain prolonged listening periods.

CHAPTER 19

Fundamentals of music theory

Could you put that up an octave just a little?

A&R representative

Once we move beyond drums and into chromatic instruments, we benefit from a basic understanding of music theory. While there are records that *have* been composed by producers with little knowledge of the theory, they are in the minority, and this shouldn't be an excuse to shy away from learning it. The fact is that an understanding of the basics of music theory will go a long way with helping you compose bass lines and leads. So, even though some of you may feel like skipping this chapter altogether, stick with it because, ultimately, it will be rewarding.

THE MUSICAL INTERVALS

Music theory is based on just one concept: the musical interval. This is the term used to describe the relationship between different notes in the scale. To understand this relationship, we should begin by studying the notes on a piano keyboard.

As Figure 19.0 shows, a piano keyboard consists of the same 12 notes repeated over along its length. Each note on the keyboard relates to a particular pitch and each is named after a letter of the alphabet.

The distance or spacing between different pitches is termed an *interval*. In western music, the smallest possible interval is a *semitone*. This describes the distance between two consecutive notes on the piano keyboard. For example, the interval between C and C# is one semitone, C# to D is one semitone, D to D# is one semitone, D# to E is one semitone, and E to F is one semitone. This interval relationship is shown in Figure 19.1.

Any two semitones can hypothetically be summed together to produce a *tone*. If you choose to skip over a semitone interval and play the next note, the two notes would be one tone apart. C to D on the piano keyboard would be considered to

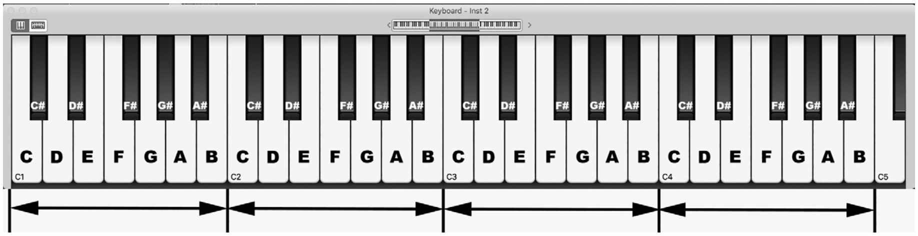

FIGURE 19.0
A piano keyboard

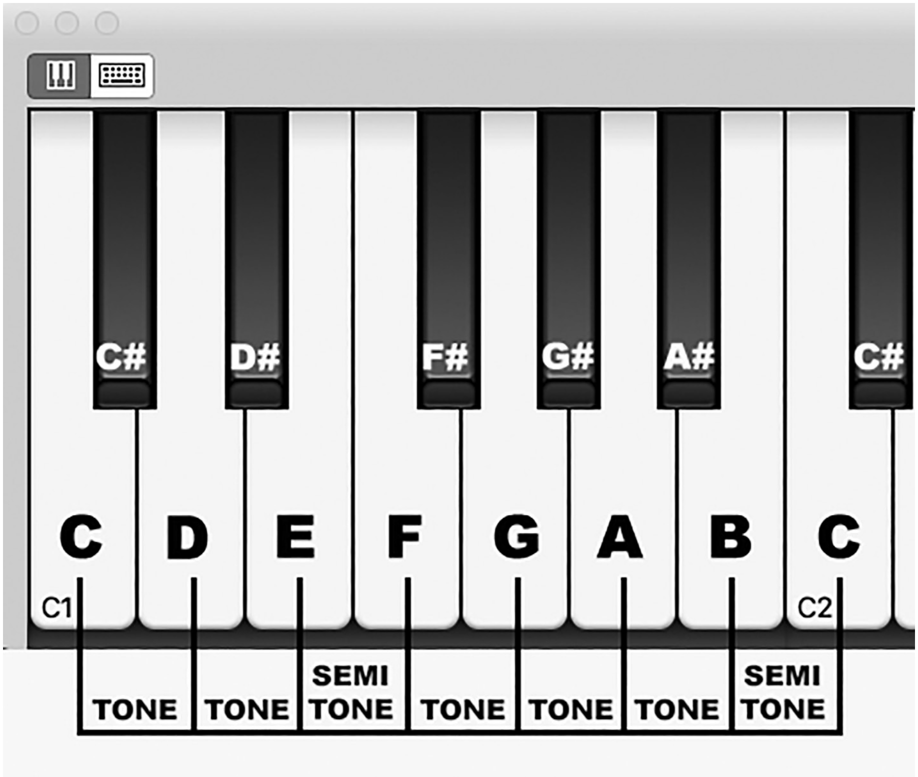

FIGURE 19.1
Intervals on a piano keyboard

be a *tone* apart because there is a note in between them (the C#). Similarly, D to E is one tone apart (E# is between them), and E to F# would also be one tone apart (the F is between them). It's this relationship between tones and semitones that forms the cornerstone of all musical theory. Armed with this understanding, it's possible to create any number of musical scales with which to compose music.

ROOT NOTES, KEYS, AND SCALES

A musical scale (aka a *key*) consists of eight notes that will sound pleasing when used together in a composition. These eight notes form the musical octave and are used to compose everything from the lead to the chords, through to the bass line and beyond. These notes are not chosen at random, however, and are predetermined by an interval pattern.

To better explain this, the first step to composing music would be to select the root note. All music will naturally gravitate towards its root note, and an excellent way to describe this gravitation would be to sing the classic line:

Doh – Ray – Me – Fah – So – La – Te …

By not performing the final *doh*, it appears incomplete. It feels as if it should resolve and finish on the next *doh*. This is the gravitation of music: it naturally wants to *resolve back to the root note*. You can choose any note on the piano keyboard for the root note of your music, but different root notes will often produce a different *feel* to the music. For the following example, the root note is C. Using this note as the starting point; we employ the following interval pattern:

Tone – Tone – Semitone – Tone – Tone – Tone – Semitone

With the interval pattern and counting upwards from C on the piano keyboard, it would produce the scale of notes shown below.

The C major scale

C – D – E – F – G – A – B – C

Tone – Tone – Semitone – Tone – Tone – Tone – Semitone

Alternatively, using E as the root note and the interval pattern mentioned above, starting at E and counting upwards would result in the following scale of notes:

The E major scale

E – F# – G # – A – B – C# – D# – E

All the compositional elements of the song would employ a combination of these eight notes. When composing music, you are free to choose any root note on the piano keyboard and using the interval mentioned above pattern; it's possible to create 12 major scales. This is because there is a total of 12 notes on a piano keyboard.

Composers employ different scales because each provides a different feel or mood for the composition. These different feels or moods can be attributed to the root note since, as previously mentioned, music will naturally gravitate towards this root note.

It is accepted by many music theorists that composing in major scales will result in music that exhibits an uplifting vibe. This is why a good number of pop music records and nursery rhymes are written and performed in major scales. However, while some dance records have been composed in major scales, they do also employ minor scales.

Minor scales are a different musical *mode* and can exhibit a more significant shift in the emotional impact and feel of the record. Minor scales are sometimes described as sad or somber, but this isn't necessarily the case. Some uplifting dance records have been composed in minor. In fact, rather than describe all minor scales as somber, minor may be better described as more serious or focused than major.

As with composing in a major scale, a minor scale can be created from any note on the keyboard only this time employing a different interval pattern:

Tone – Semitone – Tone –Tone – Semitone – Tone –Tone

Like the major scale, the minor interval pattern will result in different scales in a minor scale. For example, if you were to write in A minor, you would begin counting from the root note of A, and, following the minor interval pattern, would produce the following notes:

A – B – C – D – E – F – G – A

Similarly, if you chose to write in the scale of C# minor, using the minor formula, the notes would consist of:

C# – D# – E – F# – G# – A – B – C#

RELATIVE KEYS

With 12 scales in both major and minor, all using the same selection of notes as the root, there are instances where a major scale and a minor scale contain the same notes as one another. The only difference between the two is that they have a different *root* note.

For example, the notes in the scale of C major consist of:

C – D – E – F – G – A – B – C

whereas the notes in the scale of A minor consist of:

A – B – C – D – E – F – G – A

Both C major and A minor feature the same notes as one another. The only difference between the two is the root note of each. These two scales are known as *relatives* to one another, and every major scale will have a relative minor scale, and vice versa. This major to minor relationship offers the opportunity to move, or *modulate*, between the two different scales during composition to add more interest to a musical piece.

Although not prevalent in dance music, there is often modulation between scales in pop music. Here, the verses may be in A minor but the music moves to the C major for the chorus sections. This makes the music appear more uplifting.

You can determine the relative minor of any major scale by taking the sixth note of the major scale and using that as the root note of the minor scale. For example, in the case of C major, the sixth note in the scale is an A and therefore the relative minor root note is A. The following box shows all the relative major and minor scales.

Relative major and minor keys/scales

- C major > A minor
- C#/D♭ major > A#/B♭ minor
- D major > B minor
- D#/E♭ major > C minor
- E major > C#/D♭ minor
- F major > D minor
- F#/G♭ major > D#/E♭ minor
- G major > E minor
- G#/A♭ major > F minor
- A major > F#/G♭ minor
- A#/B♭ major > G minor
- B/C♭ major > G#/A♭ minor

In the box above, you may notice that some scales may be referred to as sharps or flats. This is because the black note to the right of C may be termed C sharp (C#) but in other scales it may be referred to as D flat (D♭) because it's also to the left of D. They both reference the *same* note but are awarded different names due to the limitations of musical tablature. As dance producers, we use the piano roll in the DAW, not the score editor (Ableton Live doesn't even feature a score editor) so I'll avoid discussing tablature.

MELODIC AND HARMONIC MINOR

What's peculiar about the minor scale is that it consists of three different scales. These were introduced over a period of years due to inconsistencies that soon become evident when you begin to compose in a minor scale.

As touched on already, the interval patterns for major and minor provide a series of pleasing intervals between the different notes or pitches in the chosen scale. For example, if you were to play A, A#, and B, one after the other it doesn't sound anywhere near as pleasing if you were to play A, then B, and finally C.

However, while the natural minor scale can provide pleasing intervals between notes, many musicians consider them to sound disjointed or displaced when composing harmonies and melodies. Because of this, the harmonic and melodic minor scales were introduced.

The harmonic minor scale is the same as the minor scale, but the seventh note is pushed up (*augmented*) by a semitone. So the G, for example, would become a G#. Without wanting to go into too much detail (for this chapter), by raising the seventh note of the minor scale by a semitone, it results in a more natural and convincing sounding chord.

While this approach suits a chord it, unfortunately, doesn't also work with a melody. Melodies work their best when they can also be performed by the human voice, whether that's humming, singing, or whistling the tune. By augmenting the seventh note, the distance between the sixth and seventh note of the minor scale are *three* semitones apart. This is difficult for a human to perform.

So, if you wish to play a melody alongside a minor harmony, you should employ the melodic minor. Here, both the sixth and seventh notes of the natural minor are augmented by a semitone each when ascending the scale but when descending the scale, it returns to using the natural minor scale.

As confusing as this may all appear, you just have to keep in mind that if you choose to write in a minor scale with chord harmonies, use the melodic and harmonic minor. If it's just melodies without any chords, you can usually just use the natural minor.

FURTHER MODES

The major and minor interval patterns are by far the most popular two modes of music. In almost all instances, electronic music producers will employ either the Ionian (major) or the Aeolian (minor) interval patterns. However, these are not the only interval patterns. To understand why we must first to investigate how interval patterns are created.

The Ionian mode (major), interval pattern exists because we begin by choosing C as the root note, and then we count up the piano keyboard employing only the natural (white) notes. When we do this, it provides the major interval pattern:

Tone – Tone – Semitone – Tone – Tone – Tone – Semitone

Similarly, with Aeolian, we choose A as the root note, and through only counting up through the natural notes, it produces the Aeolian interval pattern:

Tone – Semitone – Tone –Tone – Semitone – Tone –Tone

Following this logic, it's feasible that you can create further interval patterns by choosing different root notes and remaining with only natural notes. For example, if you were to begin to count from D upwards and ensure you only remain with the natural notes, it results in the tonal pattern of:

Tone – Semitone – Tone – Tone – Tone – Semitone – Tone

This interval pattern is known as the Dorian mode. Like both Ionian and Aeolian, you can write in any number of keys in this mode, and each will exhibit a different emotional feel.

If our root note is E and we only count up the white notes, we produce another mode known as Phrygian:

Phrygian (E)
Semitone – Tone – Tone – Tone – Semitone – Tone – Tone

Lydian (F)
Tone – Tone – Tone – Semitone – Tone – Tone – Semitone

Mixylodian (G)
Tone – Tone – Semitone – Tone – Tone – Semitone – Tone

Locrian (B)
Semitone – Tone – Tone – Semitone – Tone – Tone – Tone

Each of these modes exhibits different emotions that are (very) loosely described below:

- Ionian – happy and uplifting (common for electronic music).
- Aeolian – serious and sometimes somber (a common choice for electronic music).
- Dorian – a minor blues feel (often used for R&B).
- Phrygian – Spanish, Latino feel (can be useful for chill-out/ambient/summer styles).
- Lydian – uplifting (sometimes useful for trance genres).
- Mixylodian – a dominant blue feel (useful for R&B).
- Locrian – unstable, nervous feeling (rarely used).

While it's worth experimenting with these modal forms, to gain a better understanding of how modes can affect the emotional impact of your music, I would hesitate in recommending these for club-based tracks.

From my 25+ years of experience with writing, producing, and mixing music in this field, most dance records are written in either major or minor scales. Although almost every scale is used, the most popular are usually *A minor*, *C major*, *D major*, *G♭ minor*, *E minor*, *and C minor*.

While it could be argued that the choice for writing in A minor by many dance producers is likely due to *A minor* being the relative of *C major* – the only minor scale with no sharps – I believe it has developed into a more carefully considered practical choice than one based on the simplicity of the scale.

Club DJs employ a technique known as *harmonic mixing* that consists of matching the musical key of different records. This helps each track flow and ebb into one another to create a single continuous flow of music, an essential element of club-based music today. Many dance musicians with a musically trained ear naturally want to produce tracks that can be harmonically mixed easily with other records because it's more likely that their track will be played.

This also explains the popularity of *E minor*. While this permits the artist to be more emotionally creative, this key is harmonically closer to *A minor*. Therefore, they prove less stressful for a DJ to harmonically mix in with the large proportion of A minor records.

Perhaps the most important reason for producing in A minor, however, is concerning note frequency. On many club systems, the bass response is tuned to 55 Hz. At this frequency, they will respond their best to bass frequencies, and A1 sits at 55 Hz. Since music gravitates around the root note the chosen key, having the root note at A (55 Hz) keeps the bass energy of the record at a particularly useful frequency.

CHAPTER 20

Chords and harmony

A rock musician will play three chords in front of 30,000 people and a jazz musician will play 30,000 chords in front of three people.

Anonymous

Harmony plays an essential role in electronic dance music. While some genres may not exhibit harmony in the *traditional* sense, they will still nonetheless employ chords and progressions. As these two elements form the building blocks of harmony, an understanding of what they are and how they work is essential for producing any genre of music.

Harmony describes a series of chords that are played sequentially. This chord *progression* can accompany or support a melody or motif to create changes in mood or energy. Many recent examples of electronic dance music employ chords to form their leads, while for others it is used to *plane* chords for Techno or Tech House stabs. And for the rest, they're used as the basis for the creation of bass lines and melodies. It's no secret that artists such as DeadMau5 begin most of their tracks with chord progressions.

CHORDS

A chord consists of two or more notes played simultaneously. Traditionally, however, chords will consist of three or more notes. These notes are not arbitrary and are chosen to complement one another in a harmonious way. To understand how, we must return to musical scales.

As discussed in the previous chapter, there are eight notes in a scale, with the eighth note a repeat of the first note, only an octave higher. To construct a chord, we can ignore this eighth note and assign each note in the scale a number one through seven. For example, C major consists of C – D – E – F – G – A – B, so we would name these notes 1 through 7 as shown in the following box.

Numbering/naming the notes (using C major as an example)

Number	1	2	3	4	5	6	7
C major	C	D	E	F	G	A	B

The numbering system shown above applies irrespective of the scale. The numbering will always correspond to the same note position so if requested to play a melody consisting of notes 5-3-4-2-3-5, you could play the correct notes no matter the scale.

The most straightforward chord in the composition of music is the *triad*. This is a chord consisting of three notes that are a third and fifth apart from the first note of the chord. If you were composing in the scale of C major, using C as the first (root) note of the chord, it would consist of the notes C – E – G. This is because the note of E and G are three and five intervals consecutively above the root note of C.

With this theory, it is possible to construct a chord in any scale using any note within that scale as the root note of the chord. Provided the further notes consist of third and fifth intervals above the root note chosen, the chord will remain harmonious.

As different scales feature different notes, chords can consist of both natural and sharps or flats. The next box shows that writing in the key of E major with E as the root note of the chord would result in E – G# – B.

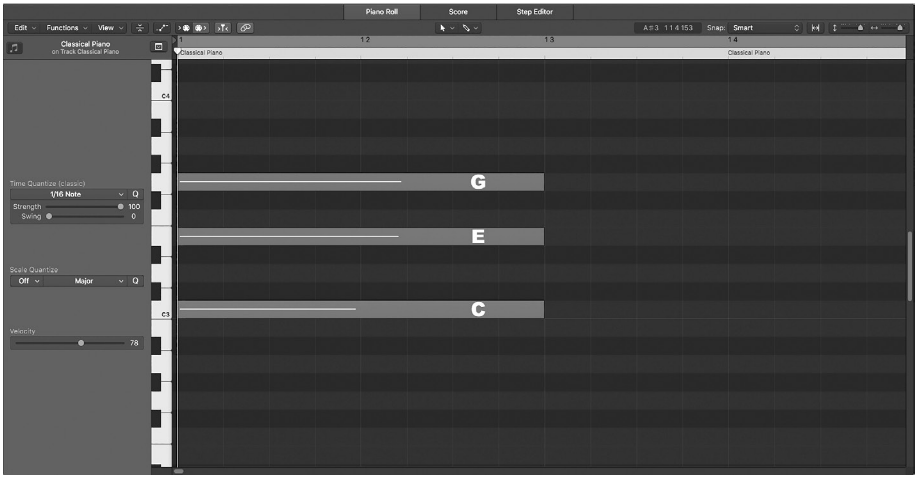

FIGURE 20.0
Logics piano roll editor showing chords CEG

The notes of the Ionian scale

Scale	1	2	3	4	5	6	7
E major	E	F#	G#	A	B	C#	D#

MAJOR AND MINOR CHORDS

Just as scales can be major or minor, so too can chords. Understanding the difference between the two is fundamental to the composition of music. Moving from a major to a minor chord (or vice versa), or employing a majority of major or minor chords in a progression will help determine the mood of the music. What differentiates a major from a minor chord is the position of the *third* interval in the chord.

In our example of a triad chord in C major, the notes consist of C – E – G. There are two intervals in this triad: the third, *bottom interval* of E, and the fifth *outer interval* of G. Because the chord's root note is C, we check to see if the bottom interval of E is present in the C major scale. It obviously is, and so this becomes a *major 3rd*, resulting in a major chord.

C major chord

Scale	1	2	3	4	5	6	7
C major	C	D	E	F	G	A	B

Remaining with C major, suppose a chord was built from the root note of D. Here, the chord would consist of the notes D – F – A. Even though we are composing in the key of C, because the chords *root note is D*, we make comparisons to the notes in the D major scale. The third interval in our chord is an F, but this note is *not* in the D major scale. Because this note is *not shared* in the scale we are composing in *and* the scale determined by the root note of the chord, the third interval is a *minor 3rd*. Any chord that has a minor 3rd is a minor chord.

Triad chord in C major with D as the root

Scale	Root	2	3	4	5	6	7
C major	D	E	F	G	A	B	C
D major	D	E	F#	G	A	B	C#

Because the third interval determines the difference between a major and minor chord, you can convert any major chord into a minor chord and vice versa. This is accomplished by either *augmenting* (raising by a semitone) or *diminishing* (lowering by a semitone) the third interval.

PERFECT FIFTHS

Alongside major and minor, chords may also contain a perfect fifth. Perfect fifths have an important relationship with the 'tonic' note of a scale. To determine whether a chord includes a perfect fifth we must again compare two scales. In our D minor chord (D – F – A), the fifth interval is an A and this note exists in both C major and D major scales. Consequently, the A is a perfect fifth.

While this shared note theory probably appears convoluted, it is essential to understand the relationship. All the musical scales are derived from the same chromatic notes, and therefore they are all inextricably linked with one another.

For example, if you were to compose in C major with a root note of E, we would compare the notes in C major to the notes in the key of E to see if both scales share the same third and fifth intervals. If both share the same third interval, the chord is a major, and if they share the same fifth interval, the chord contains a perfect fifth. The importance of relationships between major and minor chords and perfect fifths will be discussed further in this chapter.

Of course, comparing scales to determine whether a chord is major or minor is a longwinded approach. To simplify the process, Roman numerals are used to describe the chords in a scale. An example of this is shown below.

Roman numerals in place of decimal (major scales)

Note	1	2	3	4	5	6	7
Numeral	I	ii	iii	IV	V	vi	vii

The numerals are a mix of uppercase and lowercase. The uppercase denotes major chords, and the lowercase indicates minor chords. In any major scale, if you were to construct a chord using the first, fourth, or fifth note (I, IV, V) as the root note, it would result in a major chord. And if you were to use the second, third, sixth, or seventh as the root note (ii, iii, vi, vii), it would result in a minor chord.

To summarize:

1. If you construct a triad chord in a major key using the first, fourth, or fifth note as the root of the chord, you will create a major triad.
2. If you construct a triad chord in a major key using the second, third, or sixth note in a major key, you create a minor triad.

3 You can change a minor triad into a major triad by augmenting (raising) the bottom interval (3rd) by a semitone.
4 You can change a major triad into a minor triad by diminishing (lowering) the bottom interval (3rd) by a semitone.

TRIADS IN MINOR KEYS

So far we've been creating triad chords from the major scale, but they can equally be formed in the minor scale. If you understood the concept behind the creation of triad chords in a major key, a glance at the box below will be self-explanatory.

Major and minor chords (minor keys only)

Number	1	2	3	4	5	6	7
Numeral	i	ii	III	iv	V	VI	VII

In the box above, the fifth note in the minor scale is shown as a major chord but, in theory, this would be a minor chord. This is one of only a few instances where the theory is manipulated to suit composition.

The fifth note of the scale is the dominant note and therefore, any chords that are composed of the fifth note of the scale – whether in major or a minor scale – should always be a major chord. This is so important that, even in a minor scale, the composer will augment the bottom interval by a semitone to produce a major chord!

FURTHER CHORDS

The next logical step up from a triad chord is a seventh chord. This is created by adding a fourth note to the previous triad examples and sits at the seventh interval above the root note of the chord, resulting in the intervals: 1-3-5-7.

If composing a chord in C major, using C as the root note, the chord would consist of the notes C – E – G – B. Similarly, if working in E major with a root note of E, the notes in a seventh chord would consist of E – G# – B – D#. This is shown below.

Seventh chords (using C major as an example)

Scale	1	2	3	4	5	6	7
C major	C	D	E	F	G	A	B

Seventh chords (using E major as an example)

Scale	1	2	3	4	5	6	7
E major	E	F#	G#	A	B	C#	D#

Seventh chords can be expanded on yet further by adding another note, this time the ninth note in the scale. Continuing in C major with C as the root note, a ninth chord would result in C – E – G – B – D. This same procedure can continue for further, more extensive chords, known as 11th and 13th chords. If you have followed the logic behind the creation of triad chords, sevenths, and ninths, these further chords should not require further explanation.

NOTATING CHORDS

Chords are always named after the root note and followed with some basic terminology of what they are. For example, a major chord is written as the root key followed by the word *major* or more simply *maj*. Similarly, if the chord is minor, the root note is followed with *min*.

If the chord is augmented, it will be followed by *aug.* and if it is diminished its followed with *dim*. And if it's a seventh chord, you just add a 7th at the end and so forth. The box below should clarify this with an example of a chord written in E♭ flat.

Chord naming using E♭ as an example

Chord	Notes	Names
E♭ major	E♭ – G – B♭	E♭, E♭ maj.
E♭ minor	E♭ – G♭ – B♭	E♭ min., E♭ m, E♭-
E♭ diminished	E♭ – G♭ – A	E♭ dim., E♭°
E♭ augmented	E♭ – G♭ – B	E♭ aug., E♭+
E♭ major 7th	E♭ – G – B♭ – D♭	E♭ maj. 7th

TRADITIONAL HARMONY

It is the progression from one chord to the next that creates harmony. The choice of chords and how they progress from one to another will determine the energy and mood of the music. In theory, there is nothing to stop you from composing random chords within a scale and playing them one after another to create a progression. However, this approach will often result in a piece of music that lacks direction or purpose. To ensure than harmonies ebb and flow naturally from one chord to the next, there are some guidelines we can follow.

For ease of explanation, it is best to view good harmonies as consisting of a series of paired chords with each pair linked together to produce a progression. Furthermore, each matched set of chords will exhibit a feeling of familiarity or introduce an unexpected twist.

The familiarity, or even predictability, of some paired chord progressions, is the outcome of a *natural* or *strong* progression. These progressions are a result of chords that share a note with the following or preceding chords. One of the most popular relationships is with the fifth note of the scale.

If you were to play C major and G major chords one after the other, it would result in a natural sounding progression because both chords share the G note. What's more, both these chords also contain the *perfect fifth* discussed earlier. Listeners latch onto this close harmonic relationship between notes and experience a comfortable progression.

By reversing this arrangement, and moving from the V to I will create what is perhaps the most robust progression of all. Understanding why is essential for building progressions so we should first name all the notes in the scale.

Numbering/naming the notes (using C major as an example)

Number	1	2	3	4	5	6	7
C major	C	D	E	F	G	A	B
Name	Tonic	Supertonic	Mediant	Subdominant	Dominant	Submediant	Leading tone

The important note is the seventh, known as the *leading tone*. This is because it naturally leads back onto the tonic, or the *tonality*, of the music. Because the leading tone is present in the dominant chord (V), we feel an instinctual urge for it to resolve back to the tonic (I). This can be so crucial to music that the third interval of a chord is deliberately augmented in the minor scale to produce a major chord. This ensures we create a form of resolve from the dominant to the tonic.

The emphasis on the tonic of the music plays a vital role in harmony and factors into many musical compositions. Many chord progressions are composed to create the yearning feel to return to the tonality of the music. It appears as though the music has a desire to move back to the tonic and once it does, it results in *cadence*, the feeling of resolution.

Many records have relied solely on the I–V and V–I chord progression, but it is only short. To make it longer, we can introduce further chord and a popular choice would be the subdominant. This is a major chord and is a 'sub' of the dominant so naturally leads into the dominant chord. In fact, so compelling is the IV –V – I progression that thousands on thousands of records have been composed with just these three chords.

INTRODUCING MINOR CHORDS

To add some interest to a progression, we can introduce minor chords. As discussed, the V–I is commonly employed as the finale of harmony, and therefore new chords could be added before the V. If we want to continue with a strong progression, we can maintain the tonal relationship of the leading tone. We can use any chord provided that the root notes are either four notes up or five notes down from the leading tone.

For example, by counting five notes *down* from the leading tone of the V chord we arrive at the note D. As you can see from the box below, D forms the tonic of the ii chord. Therefore, the progression could now consist of:

ii – V – I

The triad chords that naturally occur in C major

Name	**Tonic**	**Supertonic**	**Mediant**	**Subdominant**	**Dominant**	**Submediant**	**Leading**
Numeral	I	ii	iii	IV	V	vi	vii
Notes	C E G	D F A	E G B	F A C	G B D	A C E	B D F
Chord	Major	Minor	Minor	Major	Major	Minor	Minor

We could add another chord to this progression by counting up four notes from the root note of the ii chord. This results in the vi chord, therefore the progression would now consist of:

vi – ii – V – I

This has now provided two pairs of chords that deliver a natural sounding progression. This progression can also be heard on the resource pack (Figure 20.1).

You could continue to construct a chord progression using this method for as long as required. However, because it only results in strong progressions, it does form a predictable progression. Indeed, any chord progression that is constructed entirely of strong progressions may appear trite.

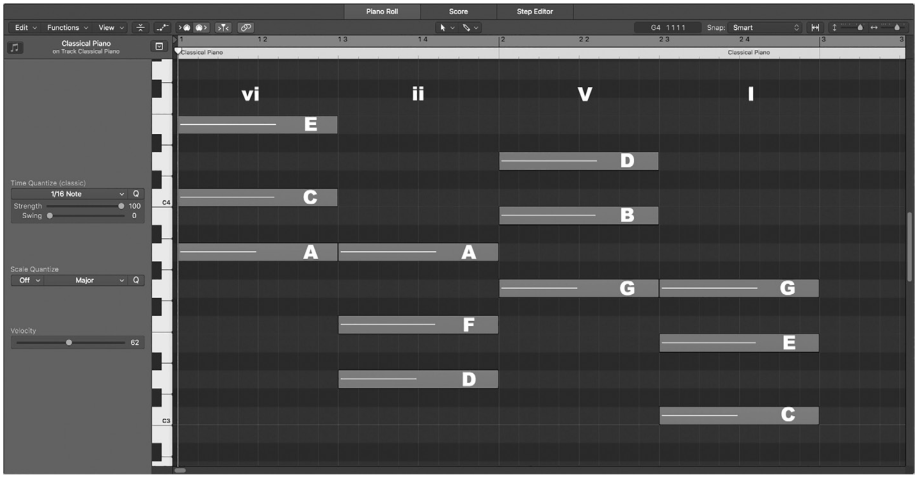

FIGURE 20.1
Chord structure in a piano roll

While chord progressions should remain reasonably predictable, we should also throw in the odd unexpected chord. This will inject some spice into the music. An unforeseen or weaker chord partnership is part of the progression that doesn't follow the previously discussed formula. Generally speaking, the more weak chord pairs that are introduced into a progression, the more unique the music will sound. But if too many weak chords are added, the music will appear disjointed or incomplete.

Although there are no rules to progressions, an accepted technique is to lead into a weak chord pairing but immediately follow that with a strong chord. That is, you move from a strong chord into a weak chord by choosing a chord that doesn't share any notes with its predecessor. This weak insertion can make the progression appear disjointed, but the following chord then shares a note with the weak chord and resolves the disjointed feel almost immediately. The result is a *strong–weak–strong* progression, a technique that reaffirms the harmony and prevents it from sounding confused. Of course, this approach shouldn't be taken too literally, otherwise all chord structures would seem the same (although many do; check out the Axis of Awesome on YouTube). Instead, it should be considered as a rough guideline for building progressions that will make 'sense' to the listeners.

DMM resource: A simple chord progression.

MINOR SCALE PROGRESSIONS

When creating a progression in a minor scale, the most commonly used chords are I - iv and VI, compared to the I – IV and V of the major scale. Additionally, when composing a progression in a minor scale, there is an almost instinctual urge to move to a major scale. So, it isn't unusual to move from a minor progression to major during parts the music.

Aside from these exceptions, composing in a minor scale follows the same guidelines as writing in a major scale. However, solidly adhering to these guidelines will limit your musical palette. Although plenty of music has been composed using just these guidelines you can create more interesting harmonies by adapting and thinking outside the box.

FURTHER PROGRESSIONS

Often, when we need to add more excitement to a progression, we can use the same chord but change the positions of the notes within that chord. These are known as inversion and can be useful if used with a *pedal tone* bass. This is where the bass instrument is held at the same pitch through some chord changes. In some dance music styles such as House, it's not unusual to use inversions of the same chord while the bass remains at either the tonic or dominant note of the chord.

To understand an inversion, we must return to earlier examples of chords. A basic triad chord consists of the first, third, and fifth notes of the scale played simultaneously. For example, a C major triad contains the notes C – E – G but the position of these can be altered. We could create a first inversion by moving the root note of this chord an octave above its usual position. This would result in E becoming the root note of the chord, producing the chord E – G – C as shown in Figure 20.2.

This could be further modified by moving the (now) root note of E from the first inversion an octave higher. This now results in the G (the former fifth) becoming the root of the chord, thus creating a second inversion.

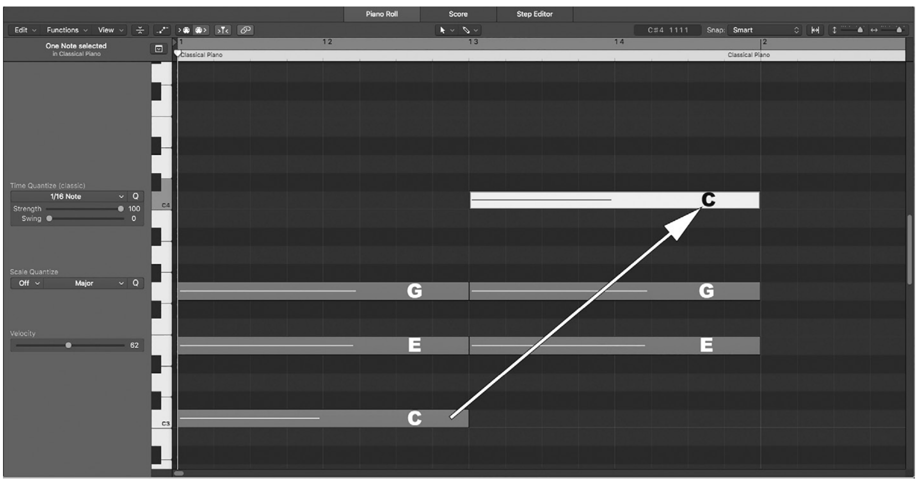

FIGURE 20.2
Piano roll editor showing chord and first inversion

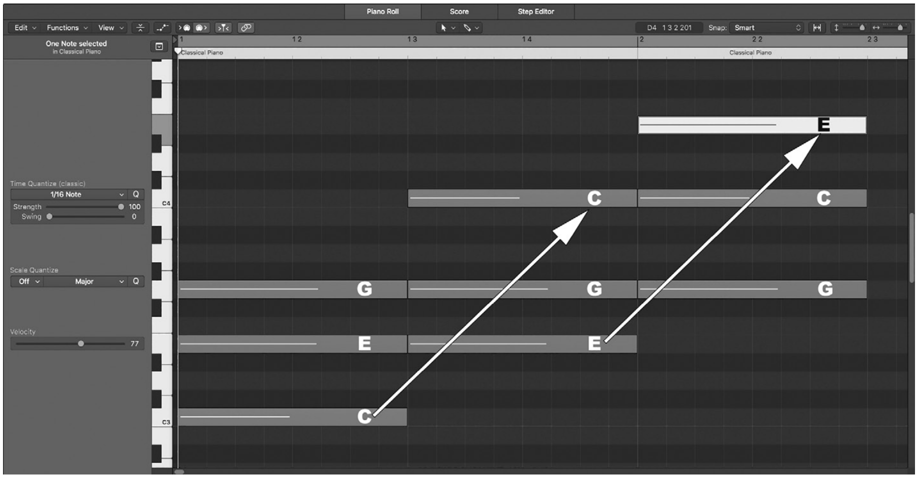

FIGURE 20.3
Piano roll editor showing chord, first, and second inversion

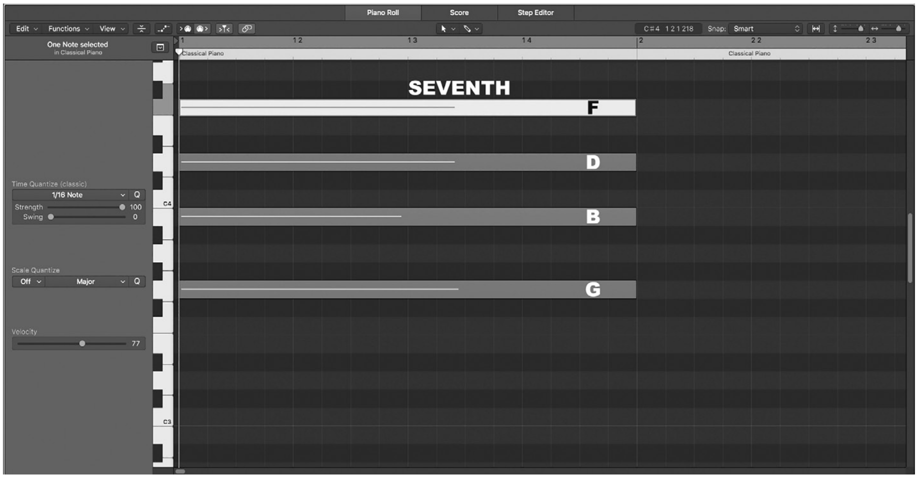

FIGURE 20.4
Creating a 7th on the V

Because inversions share the same notes as the original chord, they can introduce variety without wandering too far from the previous chord. Inversions do sound weak – with the second inversion weaker than the first – but they are not as weak as other chords with less of a relationship.

SEVENTHS

Another way to add interest is to modify some of the chords by introducing sevenths. Typically, the I, IV, and V benefit from the addition of a fourth note, but the most popular approach is on the V chord. The creation of a seventh here creates a *dominant function*. This creates an intense feeling that the music wants to return to the tonic or key of the music. So much so, that it's often used to lead the listener into believing the progression is about to return to the tonic. You can then add a twist by moving to another chord rather than to the tonic.

SUSPENSIONS

Another alternative is to suspend a chord. Chords can be suspended by merely augmenting the bottom interval (3rd interval) by a semitone. To suspend an I chord in C major; you would augment the E to F. And to suspend the V chord, you would augment the B to a C.

It's not by accident that I chose both the I and V chords as examples since these are the most common chords to be suspended. Of course, this is only music *theory*, so naturally, any chord is open to this modification.

Irrespective of what chord you suspend, it should be resolved afterward by returning to a standard chord. Resolve is essential for a *suspended third*, more commonly known as a *sus4* (the bottom interval is augmented by a semitone) because it creates a significant amount of tension that needs to resolve. The suspension most commonly used after *sus4* is *sus9*. Similar to the creation of the *sus4*, raising the tonic, or root note, of the chord by a tone, creates a *sus9*. Generally speaking, this doesn't produce as much tension as a s*us4*, and therefore it isn't as important to resolve, although it is dependent on the chord.

SECONDARY DOMINANTS

Secondary dominant chords can give the impression that the tonic of the chord has changed. To understand what these are and how they are applied, we need to revisit the dominant chord.

The relationship between the dominant and the tonic plays a vital role in music. This is because the dominant chord contains the leading tone, a central note that is one semitone below the root note of the tonic chord. So, when we hear the dominant chord, we feel the instinctual urge for it to be resolved by the tonic. We can emulate this type of behavior using two different chords and changing the arrangement of the notes within the chord.

For example, consider the first minor chord in the C major scale – the ii chord – consisting of the notes D – F – A. This is a minor chord in the C major scale because the F is not shared in the D major scale. Instead, D major scale contains the note F# and therefore, when in C major, it produces a minor chord. If you chose to augment the bottom interval of the ii chord, it would result in a chord comprising D – F# – A. By doing so, you would accomplish two things:

1. It would introduce the note F# – a note that does not exist in the C major scale, resulting in an accidental note for the scale.
2. It would also change the minor chord into a major chord.

This could be viewed as *borrowing* the I chord from the D major scale, even though in theory you are working with the second chord in the C major scale, and this is always a minor chord. Nevertheless, if you were to play a chord progression in C major and during the progression introduced the second augmented chord as a major instead of its usual minor, it will grab the attention of the listener.

When you add a secondary dominant, it is a commonplace to resolve it with a chord whose root note is either a fifth up or fourth down, emulating the effect of moving from a dominant to a tonic. This adds a resolve and prevents the note from sounding out of place in the progression.

SCALE MODULATION

Changing scale part way through a song is a popular choice for many of the pop-oriented dance records. It can also be used to provide additional energy and anticipation for a drop, but it can sound particularly cliché.

If it occurs in electronic dance music, it will happen in one of two places: when moving between different sections of an arrangement or the drop. Scale changes in these two situations are often applied delicately or abruptly depending on the track. If applied suddenly, it is usually at the drop or at the point where the music is moving from the dominant chord to the tonic chord. When used here, the dominant is played in the original key, and the tonic enters in the new key.

If the transition needs to be more delicate, then the harmony will work up to a dominant seventh before moving to a triad in the new scale. This is the best approach because you can rearrange the notes in the dominant seventh chord so that it shares as many notes as possible with the triad in the new scale. If at all possible, a smooth a natural progression could be achieved by ensuring the leading tone of the dominant is a semitone below the tonic of the new key.

Another option to smooth over the transition is to introduce a secondary dominant chord or a 'modulating' chord. A modulating or 'pivot' chord is a chord that is shared by the current key and the key you want to move to. For instance, if you're working in the key of D with the idea of moving to the scale of A, you could use the I chord. This is because the D chord's tonic (I) contains the same notes as the subdominant (IV) in A, so you could play D –E7 – A (a seventh is introduced to help ease the transition).

Whenever modulating to a new scale, it's best to use it near the end of the music, and then move upwards in tone rather than downwards. By doing so, it produces the final energetic punch of the music, driving the message home, and can be particularly useful if the vocals or rhythm section increases with this modulation. While it is possible to move down in scale, this type of movement kills the energy of the song, and is only really useful if you're drawing the track to a close.

Above all, modulation must be approached carefully and even more so if a vocalist is performing with the music. Apart from sounding cliché if overused, modulating to new scales can create problems for the vocalist, because they may not be able to perform in the new scale.

HARMONIC RHYTHM

With any progression, it is crucial they feature rhythmical symmetry. Without this, even if the chords all form a natural or strong progression, the music will appear incoherent. Harmonic rhythm is the name given to the length, pacing and rhythmical structure of a chord progression. All music is constructed from a series of interlinked passages or phrases that tie together to produce the final result. These must all work together to provide a rhythm that augments the surrounding melody and bass. This is accomplished through pairing chords together to create strong or weak progressions, and then arranging them to occur both symmetrically and rhythmically.

For example, you may choose to pair chords one and two, three and four, five and six, and finally seven, returning to chord one that would result in a 'crossover pair.' Figure 20.5 shows how paired chords could work together.

In Figure 20.6, the seventh chord is dominant that naturally resolves to the tonic (I) at the beginning of the progression. These two chords create the paired crossover progression because the tonic is not only paired with the preceding dominant (V) at the end of the previous bar, but also paired with the following subdominant chord (IV). Each paired 'set' of chords are strongly harmonious with one another but may perhaps sound 'weaker' when moving from one strong pair to another strong pair in the progression. This style of motion is shown in Figure 20.6.

The choice of where to place weaker progressions is a creative choice, but music should be largely predictable with the odd unexpected event thrown in to maintain interest. And, while there is no limit to how many pairs of chords can be used to create a progression, it shouldn't be too long.

Harmony is perhaps best viewed as a series of interconnecting links. Each paired set of chords produces a musical phrase, but all these chords together produce another musical expression. This latter musical phrase determines the start and end point of a much larger phrase within the music.

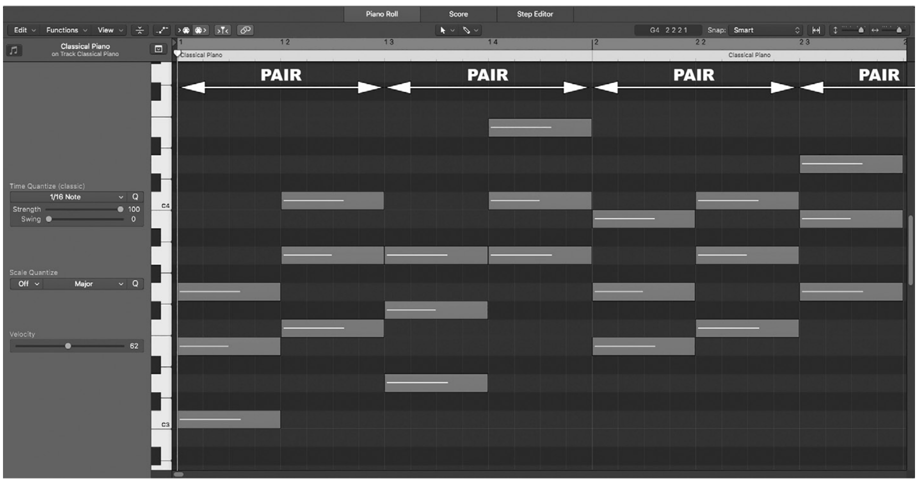

FIGURE 20.5
Chorded pairs

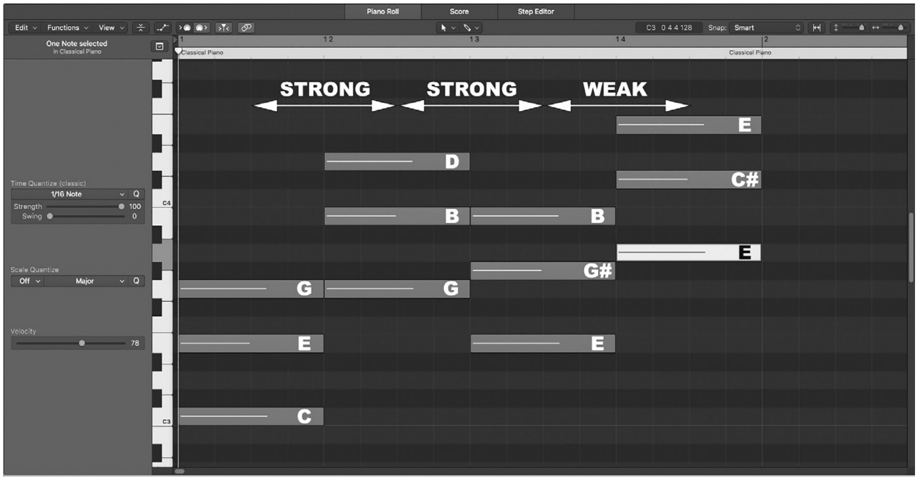

FIGURE 20.6
C to G = strong, G to E = strong, E to A = weak

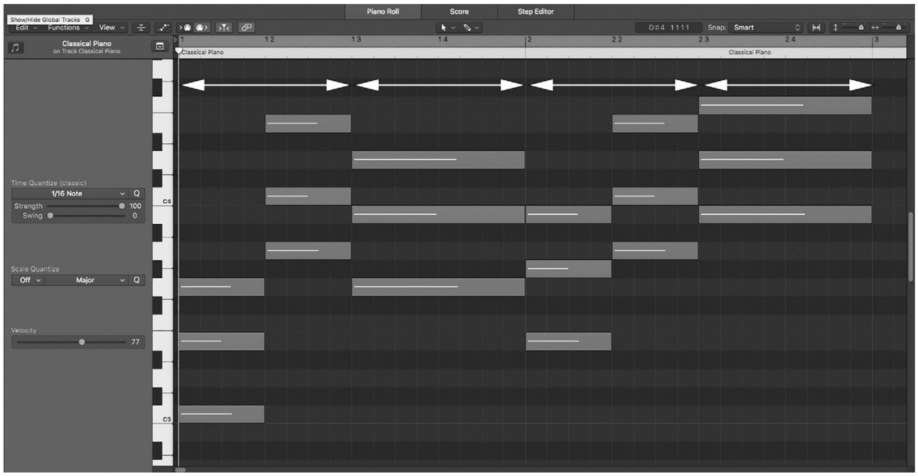

FIGURE 20.7
Two chords – one chord – two chords – one chord

Just as the producer must carefully create a progression to introduce a right amount of predictability to the listener, they must do the same with the placement and speed of the chords to establish a harmonic rhythm. Here, the aim is to make the rhythm predictable to the listener. So although you may inject the odd surprise into the chords themselves, the harmonic rhythm remains foreseeable.

For a harmonic rhythm to accomplish this effect, a chord progression should be symmetrical. This is done by equally spacing the chords over a given number of bars. For example, you may have a progression running over four bars and to attain some symmetry, you could position two chords in the first bar, one in the second, two in the third, and, finally, one in the fourth as in Figure 20.7.

Alternatively, you could employ one chord per bar for all four bars, two chords per bar for all four bars, or one chord in the first bar, two in the second, three in a third, two in the fourth, three in the fifth, and, finally, one in the sixth.

Provided you aim for symmetry to the progression the listeners can predict when the next chord change will occur. This is important because, without it, the music will appear disjointed or incomplete. The resources pack features an example of an asymmetrical progression followed by a symmetrical progression. While they are both *theoretically* correct, the second example is more coherent and complete.

Alongside the symmetry, you must also consider the speed or frequency with which the chord changes occur because this will influence the feel of the music. For example, a conventional method to add excitement to music would be to begin with a slow harmonic rhythm and gradually increase the speed of this rhythm as the music progresses. This is a favorite technique employed in some genres of Trance, the intro, and body section will use a chord progression with perhaps one chord per bar while in the reprise (the uplifting part), the chord progression changes to two chords per bar.

On the other end of the scale, genres such as Lo-Fi, Chill Out and Trip Hop employ a slow harmonic rhythm with a slow rhythm to create the laid-back feel of the music. Other styles, such as Disco House, Dubstep, and Future House will employ a slow harmonic rhythm mixed with a faster bass or melody line. This latter approach can be a critical compositional element of electronic dance music. This is known as a *canonic progression*, where the chord progression will complete in eight bars, and the melody will finish in four.

This is best thought of as a series of different sized cogs all turning at different speeds, moving in and out of one another. For example, the drum rhythms may repeat every bar, the bass every two bars, the leads every four and the chords every eight. This results in a series of instrument rhythms that continually move in and out of sync with one another over the period of the music. This approach is especially important with the more minimalistic genres such as Tech House and Techno. Here, it's typical for the bass to repeat every half bar, the drums to repeat over one bar and a chorded stab to rhythmically repeat every two bars.

CADENCE

It would be careless to close a chapter dedicated to harmony without touching on *cadence*. To discuss this thoroughly would involve an exhaustive examination of harmonic analysis and musical form that could easily require the rest of this book, so this should be considered an overview of the subject.

As touched on numerous times in this chapter, music takes the listener on a journey. A cadence is as a way of bringing the story to a close or to signify the end of a particular musical phrase or loop. In many respects, cadence also occurs in films and TV dramas. Although these are often full of unexpected twists and turns, the viewer more or less knows what to expect in the way of scenes and the film in general.

The viewer will intuitively feel each scene working towards to the next and, ultimately, the final stage with the build-up to where the protagonist overcomes all. If there were no tension or build to this final scene, or if the protagonist failed, the story wouldn't resolve, and it would be disappointing.

While receiving something that isn't expected can sometimes be rewarding, we have a feeling of greater reward if we receive what we expect. The way in which a director builds up into the next scene and constructs the grand finale, building on the viewer's expectation, is what separates a great film from an average one.

When cadence is employed in music, it is to achieve this same goal. The music is building on the listeners' expectations, guiding them through the music, and building to signify that they are arriving at the end of a musical phrase or passage. The way in which we use cadence to build on the listener's expectations is often what separates a great piece of music from a run of the mill piece.

Notably, cadence occurs in all the elements of music and not just the harmony. The rhythm section with its drum rolls, the harmony, the bass, and even the musical structure will all utilize cadence to lead the listener into the next stage of the song. Of all these, however, it is the harmony that forms the essential cadence, and this is why returning to the tonic, or home 'key' of the music can be important.

CHAPTER 21

Composing and designing strings

I see dance being used as communication between body and soul, to express what is too deep to find for words.

Ruth St. Denis

After the drums, a common starting point for many musicians is the chord progression. This is because they often form the genesis of the music. With the chords written, you can determine the melodic motion of the record, build the bass and leads from them or even use them as the lead instrument.

The theory behind building chords and composing progressions have been discussed in detail in the previous chapter, and those techniques can be applied to create a progression for any genre of music. However, if you listen carefully to electronic dance music, several chord progressions present themselves time and time again. These popular chord progressions are shown in the following box.

The most popular chord progressions in electronic dance music (in no order of preference)

1 1-5-6-4
2 5-6-4-1
3 6-4-1-5
4 4-1-5-6
5 1-6-4-5
6 1-5-6-4
7 6-7-1-3

Each of these progressions shows the root notes of the chord and their respective position in the scale. For example, if composing triad chords in A minor, using the first chord progression would result in the following progression:

Notes in A minor

Number	1	2	3	4	5	6	7
A minor	A	B	C	D	E	F	G

(1) A – E – G/(5) E – G – A/(6) F – A – B/(4) D – F – A

Of course, these chords could be sevenths or ninths, augmented or diminished, or even inverted. What matters here is that initially, the chords root note follows the progression pattern.

While this may seem overly simplistic, or perhaps even cheating, there is no escaping the fact that from every chord progression we have available, these particular progressions appear consistently throughout the music.

Although many chord progressions will start on the root note of the scale (1), you'll notice that some of these progressions begin on the fourth, fifth, and sixth notes of the scale. By doing so, it can produce a sequence that appears to have no end and can work to significant effect.

An example of this can be heard in *Daft Punk*'s 'Get Lucky.' Here, the chord progressions are cleverly composed so that the song never seems to resolve. They accomplish this by using an ambiguous progression. In 'Get Lucky,' the scale appears to be F# minor, playing the chords I III v IV. However, the music does not resolve to the tonic (F#) and instead cadences to B minor.

This makes the music appear to be in B minor because the chords exist in both these scales and the ambiguity of the tonic results in a song that never seems to resolve. Combined with the climbing vocal structure, it makes us want to repeatedly listen because we're waiting for a resolve that never happens. For the rest of the examples in this chapter (and on the resource pack) we'll use the 1-5-6-4 structure in A minor (A > E > F > D).

With this basic chord structure down, we have the entire basis of a track. It is common in dance music for the bass to follow the root note of the chords, but even if this isn't the case, it is possible to draw a bass line from the chords and use the chords to produce both the lead and harmony. The techniques for composing bass and leads will be discussed in following chapters, but with a chord progression composed we can begin to look at the sound design.

Although rhythmical staccato chords (short notes) can be used as leads, the more common use of harmony is to supply the strings and pads. These rarely form the central focus of dance music but understanding how the basic form is created will help towards understanding how LFOs and envelopes work together to produce compelling evolving timbres. While there are no definitive methods for creating these sounds, there are some guidelines that we can follow. Typically, pads and strings are used to provide one of three things:

- To enhance the atmosphere in the music, this is especially the case with Chill Out and ambient music.
- To fill any 'holes' in the mix between the rhythm and groove of the music and lead or vocals.
- To be used as a lead itself through either noise gating or producing a timbre.

Depending on the purpose will determine how it should be synthesized. Although many of the sounds in dance music will employ an immediate attack stage on the amplitude envelope, strings and pads will typically use a slower attack.

We determine tones that start abruptly to be louder than those that do not, but we also perceive sounds with a slow attack stage to be 'less important' even though this may not be the intention. Consequently, when pads or strings are employed as 'backing' instruments, they should fade in, but if they are being used as a lead, it would be preferable to employ no attack at all. This abrupt start to the timbre will help it cut through the mix and also provide the impression that it is an essential part of the music.

A favorite technique to demonstrate this is the gate/pumping pad. This consists of an evolving, shifting pad that is played as one long progressive note throughout the record. A noise gate is inserted on the pads channels with a side-chain signal used to trigger the noise gate. Provided the side-chain signal is rhythmical; it will rhythmically pulse the pad. This can also be accomplished through third-party plug-ins such as Xfer LFO Tool. This can be placed onto a channel, and you can draw in the pulsing behavior.

FIGURE 21.0
The XFer LFO tool

For this pulsing gated pad to be genuinely useful, it must evolve texturally throughout. If this technique is used on a sustained tone with no textural movement, you may as well just retrigger the timbre whenever required. Indeed, it's this textural movement and development that's essential to creating a great sounding pad or string. If the tone remains static throughout without any variation, it becomes tiring to listen to.

Perhaps the most straightforward route for pad and strings in your mix is to use Spectrasonics' Ommisphere. This has been used by countless producers since its release in 2008 and holds an astounding number of strings, pads, basses, and leads that have been sampled from synthesizers, orchestras, and nature. You can choose the instrument and then apply effects and envelopes to modify the sample further.

While Omnisphere is, without doubt, the most famous plug-in for pads and strings, you will eventually want to program your own, even if it is just for layering further synthesized tones on top of these samples. Analog modeled synthesizers are one of the best for the creation of pads and strings because the gentle, random fluctuations of the oscillators pitch can sometimes provide an additional sense of movement. Alternatively, wavetable synthesis can produce some exciting sounds provided that you select appropriate tables. Whichever you choose, both will require modulation from envelopes and LFO to gradually increase or decrease the harmonic content/pitch/oscillator mix/wavetable scanning, etc. during the progression of the pad. To better explain this, we'll look at how envelopes are used to create the beginnings of a great pad or string.

By employing a fast attack and a long decay long on an amplifier envelope, we can determine that the sound will take a finite amount of time to reach the

FIGURE 21.1
Spectrasonics Omnisphere 2

sustain portion. Provided that the sustain is set just below the amplifier's decay stage, it will decay slowly to sustain and then continue to play until the note is released. At this point, it will progress to the release stage.

This creates the basic premise of any pad or string, to continuously play until the note is released. Assuming that the pad has a rich harmonic structure from the oscillator section, textural development could be introduced by modulating or increasing a low-pass filter cut-off while the pad plays. By doing so, there will be a gradual increase in harmonics.

Using a filter envelope with a long attack stage, the action of the filter will be introduced slowly, gradually increasing the harmonic content of the pad. Alternatively, if the modulation envelope were augmenting the wavetable scanning, as the attack stage grows, the wavetable would move through its various tables.

It is important to note here that the amp and modulation envelopes are designed to operate over different periods for the pad to evolve in harmonic content. This arrangement can only be one way because if the function of these envelopes were reversed (the modulation envelope has a fast attack, but the amplifiers are longer) the modulation would have little to no effect because we would be unable to hear the pad because it's still increasing in volume.

To texturally evolve pads or strings, they must be designed to exhibit a high harmonic content because we require a sound that is rich in harmonics for a filter to sweep. In many analog synthesizers, we should employ saws, pulse waves, and noise oscillators. The choice of oscillator depends on the style of pad required, but as a general starting point for practice, two saws, or a saw and a pulse would provide a harmonically rich timbre. These oscillators should be slowly detuned from one another in semitones while carefully listening to the results. You should then stop detuning when the tone appears 'thick' enough or just before the oscillators separate into two distinct pitches. If the sound seems light, then you can add further weight with a third oscillator set to a sine or triangle wave, detuned so that it sits an octave below the other oscillators. Also, changing the phase of the oscillators and introducing unison will always thicken up the sound.

With wavetable synthesis, it's best to choose wavetables with high harmonic content, the richer, the better because these can always be filtered if needed. You can't add further harmonics to a wavetable that are not already present. Depending on the wavetables, two detuned from one another is usually enough. After choosing them, experiment with detuning, unison, blend, and wavetable scanning to find the most exciting combination of the two sounds.

Alongside using modulation envelopes to modulate the wavetable position or filters (or both), we should also employ LFO modulation. This is because many envelopes are one shot, so once the modulation envelope has ended, we require a constant cyclic modulation to maintain some interest. Here, a saw, triangle, or sine wave can be used to slowly and gently modulate other parameters. Common destinations here include:

- oscillator pitch
- filter
- resonance
- pulse width
- wavetable scanning
- wavetable blend
- FM amounts
- volume
- pan
- effects.

To help lay down some foundations, what follows is a guide to how some of the pads and strings employed in dance music are created. This is by no means the definitive list, but its purpose is to offer some general starting points. In the examples, I'll be using analog synthesis, but you can replace any of the oscillators with wavetables if you're using a wavetable synth. The only difference is the sound (with the obvious benefit that you can modulate the wavetable scanning).

CHORD PLANING

Chord planing is a classic technique that was first used in the 1990s and has since become an incredibly popular effect. The process starts with a basic chord, either a triad or a seventh. You can create a more significant chord if you wish, but the larger the chord, the more out of tune it can sound when planed. You can program a patch in a synthesizer, or more commonly, artists will use an instrument such as a piano, violins, or guitar.

The chord is rendered to audio and then imported into a sampler. With the note played at its original sampled pitch, the chord repeats as you would expect but as you play different notes, the chord changes in interesting ways. This technique appears on countless records and is especially prevalent in Techno and Tech House to produce the chorded stabs.

CHORD LEAD

The chorded lead is another favorite effect that is used in countless dance records, from DeadMau5 to Major Lazer. There is no one definitive way to create this style of lead, and you can use either wavetable or analog synthesis to generate different styles of sound, but the general process remains the same. We'll keep this example simple using only a couple of saw waves and a pulse but feel free to experiment with different oscillations and wavetables.

Detune the saws from one another until it begins to produce a thick, full sound. You can use unison to add further width if necessary, then detune the pulse wave an octave down from the saws to add additional depth to the sound. Because this chord is used as a lead, it requires an immediate attack, with a fast decay, full sustain, and a quick release.

Use a 12 dB low-pass filter with no resonance and a low setting on the filter cut-off. Then modulate the filter cut-off with a fast attack, long decay, and no sustain or release. Send the modulation envelope to the filter by 50%. With the chord playing, reduce the modulation envelopes decay, and the chord should start to pluck. This produces the primary starting point for the lead. Experiment with the amplitude release and the modulation envelope to complete the sound.

Almost all leads benefit from reverb but if this is applied, employ a long pre-delay so that the transient can pass the reverb unmolested because if the transient is captured, the sound will be pushed into the background of the mix. Also, use a large reverb room with a long release but follow this with a noise gate to remove the tail of the reverb before the next note starts.

RISING PAD

This pad is constructed by employing two saw oscillators detuned from one another. Start detuning the second oscillator from the first in in semitones until just before the sound begins to split into two separate oscillations; this will produce a sound that is thick and full. You can enhance the sound further with unison, or if you need the sound to exhibit more weight, then you can add a pulse wave tuned an octave below the saws.

Use a fast attack, short decay, medium sustain, and extended release on the amplitude envelope. Use a 12 dB (or 24 dB) low pass filter with the cut-off and resonance set quite low to produce a static harmonically rich timbre.

From this, set the envelope of the filter to a long attack and decay but use a short release and no sustain and set the filter envelope to maximum positive modulation. Finally use the filter's key follow so that it tracks the pitch of the notes being played. This results in the filter sweeping up through the pad before slowly settling down. If the pad continues for an extended period, then you may also want to modulate the pitch of one (or both) of the oscillators with a triangle or sine wave LFO and modulating the cut-off or resonance of the filter with a square wave LFO. These can use a medium depth and a slow rate.

BRIGHT STRINGS

You can create bright strings by mixing a triangle with a square, or triangle and pulse-width oscillators and detuning them from one another. Detune the oscillators from one another in semitones until the sound begins to thicken up. Similar to the previous pad, use a fast amplifier attack with a short decay, medium sustain, and long release but the filter envelope should employ a long attack, long release, high sustain, and quick decay.

Use a 12 dB low-pass filter with the cut-off set low but resonance around ¾ so that the timbre exhibits a high resonant quality at the cut-off point. Finally, modulate the pitch of the triangle oscillator with a sine wave LFO set to a slow rate and a medium depth and employ a positive filter key follow.

The LFO modulation creates a pad that exhibits a natural analog 'character' while the filter tracks the pitch and sweeps in through the attack and decay of the pad and then sustains itself through the amplifier's envelope. If the pad is to continue for an extended period, it is worth employing a sine, pulse, or triangle wave LFO to modulate the cut-off or oscillators of the filter pitch. If a pulse wave has been applied, modulating the pulse width can produce further motion.

MOVING PADS

This pad consists of two detuned saw oscillators. As is common, the detuning amount depends on the synthesizer, so detune the oscillators from one another by semitones until the sound becomes as full and thick as you need it. Use a medium attack, sustain and release with a short decay for the amplitude envelope, with a fast attack and release with a long decay and medium sustain on the filter envelope. You can use a 12 dB low-pass filter to modify the tonal content and start by setting the filter mid to low and the resonance approximately half way.

Add a triangle wave to mix, but pitch this up an octave above the saws, and then tune it further up via semitones (usually +3, +5, +7, or +9) until it produces a bright tone above the saws. Now, gently modulate the pitch of one of the saw waves with a pulse or saw wave LFO. This must be applied slowly, and only by a small amount to prevent the timbre from being too wild.

LIGHT STRINGS

You can produce a light string with a single oscillator. You can use a single saw or pulse waveform for this with some unison to make it slightly stronger. For the amplitude envelope, you can use a medium attack, sustain, and release with a fast decay on the amp envelope. Use a 24 dB low-pass (or high-pass) filter and then modulate the cut-off slowly with a triangle wave. Using the resonance, you can increase the brightness of the pad until it sits with the rest of the record.

EFFECTS

All pads and strings will benefit from a series of effects. You can use any effects you see fit here, but the most typical applications are reverb, phasers, and chorus effects. Reverb is often used to push the pad or string into the background of the record, while both phasers and chorus can be used to add interest and movement. These effects usually benefit from slight modulation if possible to maintain movement. It's also common practice to pump or side-chain both strings and pads to move with the pulse of kick drum. There is no requirement to pump these aggressively, but a gentle motion helps to accentuate the all-important kick. If they are not pumped, the kick will not appear as powerful as it should be.

CHAPTER 22

Composing and designing leads

Melody is the essence of music.

Wolfgang Amadeus Mozart

Except for only a few genres, electronic dance music does not typically employ melodies and will rely instead on a simple motif. Also known as a *motive*, *hook*, or *theme*, this is a rhythmical idea that is repeated throughout the music. Its aim is to provide listeners with a reference point for the track.

The opening bars of *Eric Prydz*' 'Pjanoo,' the repeating chord patterns in *Major Lazer's* 'Lean On,' or *Avicci's* 'Levels,' then we have *Calvin Harris*' use of rhythmical chords in 'Let's Go,' and the growing synth rhythm in *DeadMau5's* 'Faxing Berlin' are all great examples of motifs. As soon as you hear them, you can immediately identify with the track, and, if they are composed well, they can be compelling tools.

Motifs follow similar guidelines to the way that we ask and then answer a question. The first musical phrase often asks the question, and this is 'answered' by the second phrase. This type of arrangement is known as a 'binary phrase' and plays an essential role in all electronic dance music.

To demonstrate this, we'll examine the motif in Avicii's 'Levels.' Admittedly, in the fast-moving world of electronic dance music, this is an 'old' track, but it was nevertheless famous and instantly recognizable to many. Plus, the theory employed in this motif is simple to understand, and can be applied to all genres of music. 'Levels' is composed in C# minor or E major (they are relatives). The notes in C# minor consist of:

C# – D# – E – F# – G# – A – B – C#

The lead motif consists of two instruments layered together mixed with side-chain and a tape-stop effect. The first lead layer consists of a series of stabbed chords that back up the main lead, these chords consist of a minor and three majors in the scale: i > III > VIII > VI. Figure 22.0 shows these on a piano notation.

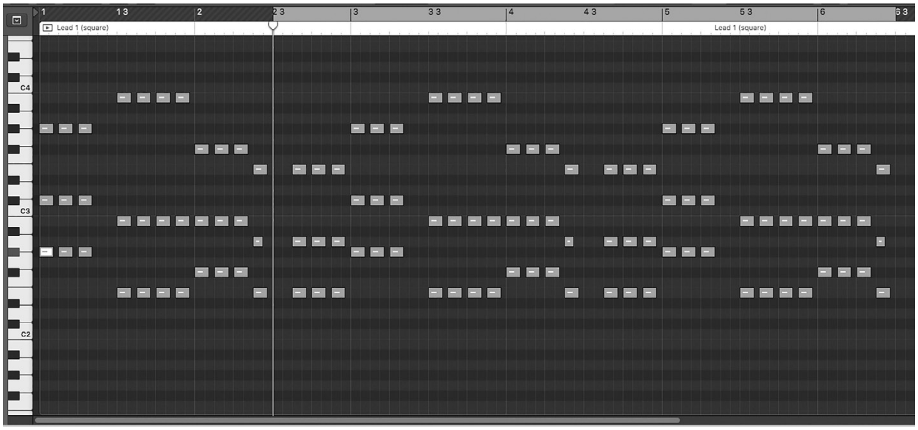

FIGURE 22.0
The chord backing

The track employs an asynchronous motif. The start of the chord and lead motif occurs at beat three of bar two rather than beat one of bar one. The bass and vocal motif, however, begin at the start of the first bar. This produces the asynchronous feel that has discussed in previous chapters to allow for repeated listening. The lead layer that sits on top of the chord motif (and starts at the same time as the chords) can be broken down into a binary phrase as discussed previously. The first phrase consists of:

C#4 – B – G#3 – F#3 – E3 – E3 – E3 –
E3 – E3 – E3 – *D#3 – D#3 – E3 – E3*

and the second phrase consists of:

C#4 – B – G#3 – F#3 – E3 – E3 – E3 –
E3 – E3 – E3 – *C#3 – C#3 – B2 – B2*

DMM resource: You can hear the melody played via a piano and also access the MIDI file.

Both phrases of the lead motif remain at the same pitch and it is only at the *end* of each phrase that there is any change between the two. At the end of phrase one, the pitch remains constant with only a slight variation introduced when it drops a semitone for two notes at *D#3*. It then rises again back to *E3* to complete the first part of the phrase.

Phrase two repeats the same rhythm and pitch as phrase one with the only change occurring at the end. Here, the pitch drops further than the previous phrase, moving from the *D#3* in the first phrase to *C#3* and then, rather than return to *E3*, it moves further down the scale by a whole tone to *B2*. Aside from this small amount of pitch variation both phrases remain simple and almost identical regarding rhythm.

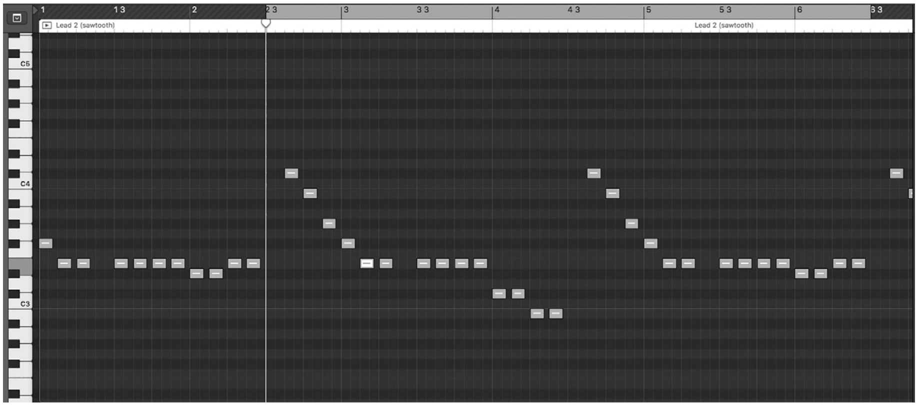

FIGURE 22.1
The lead motif

This style of repetition is often mistaken to be exclusive to electronic dance music, but every other form of music is based on the repetition of single or multiple themes. In fact, the underlying theory behind writing any great memorable record is to bombard the listener with the same theme so that it is embedded and becomes an earworm.

An earworm, or to use its accurate term, Involuntary Musical Imagery (IMI), is a phenomenon whereby a melody remains in the listener's head long after the song has finished. We've all experienced this at one time or another, but it is the central focus for many musicians and artists – to get their melody stuck in *your* head. This is accomplished via excessive repetition and occurs in every genre of music, thus it is not only limited to dance music. The only difference is that the repetition is upfront in dance music whereas, in other forms, various techniques are employed to detract from the repetition of the music.

For example, listen to the pop song 'When Will I See You Again' by Owl City (from the movie *Wreck-It-Ralph*). Ignoring the vocals, listen to the theme played by the bell tone when the vocals start (it's in the background). This same two-bar motif repeats over and over and over again throughout the entirety of the track – non-stop. After the first four bars, strings are introduced playing the same theme, and after another four bars, the bass is added, yet again playing the same theme while a side-chained bass plays the chord structure. When the song enters the chorus, the strings and bell play the same theme yet again.

So why don't we hear the repetition?

First, there is a chord change occurring in the song. The first two bars are D – A – Bm – G and then they move to D – A – Em – G for the next two bars. These same eight chords are repeated throughout the song but are syncopated into the pre-chorus. The last verse is extended so that chords D and A play in the verse and the pre-chorus starts with the Bm and G.

This alongside the vocal melody and the introduction of further instruments (even though they're playing the same motif) disguises the repetition. This is not exclusive to this particular track either, and all pop music follows the same structure if you listen carefully enough (which you should – ear training is essential).

The same technique is employed in classical music. Here, the same initial motif is repeated over and over again, but as the motif is repeated, further instruments are added. Examples of Classical motifs are Mozart's *Eine kleine Nachtmusik*, Beethoven's *Moonlight Sonata*, and Vivaldi's *The Four Seasons* (yes, all four of them).

COMPOSING A MOTIF

There are three essential elements to creating an earworm motif or theme: these are simplicity, rhythmical repetition, and variation. When it comes to creating one, it is best to compose the rhythmical content first using a single note pitch. Then when a rhythm is down, you can move onto introducing pitch changes between phrases at a later stage.

By far the easiest method of composing a rhythm is to connect either a piano keyboard or a pad trigger (such as the Arturia Beatstep or Push for Ableton) and tap out rhythms while recording the subsequent MIDI data. You could use any sound for this because you're only aiming to produce a rhythm.

After 20 minutes, listen back to the MIDI recording and pick out the best rhythms you came up with or mix and match them if required with the DAW's editing tools. Alternatively, you could use a synthesizer's or DAW's arpeggiator to record MIDI patterns and rhythms that you can later edit. The aim here is to do nothing more than produce a good, catchy, rhythmical pattern.

A benchmark to a memorable rhythm lies with its pace. The rhythm should be between 120 and 140BPM and be no more than a bar long (two or four bars if you include variation). It should also be something that you could pace to. The best rhythmical motifs in dance music tend to be those that it would be possible to go jogging with and used to keep time.

Beyond the rhythm, we have pitch, and this can (somewhat briefly) be condensed into two simple theories, direction and amount. This is known as the musical contour of the motif. Here the structure should be kept simple, but this too should also follow a rhythmical pattern. For example, play the following notes on a piano:

1 C – C – G – G – A – A – G
2 F – F – E – E – D – D – C

If you chose to play the above, you would have recognized it immediately; it's 'Twinkle Twinkle Little Star.' There are two things to take note of here:

1 The melody rises in pitch, only to then fall and rise again.
2 The pitch moves rhythmically.

If we compare 'Levels' to 'Twinkle Twinkle Little Star':

'Twinkle Twinkle Little Star'

1 C – C – G – G – A – A – G
2 F – F – E – E – D – D – C

'Levels'

1 C#4 – B – G#3 – F#3 – E3 – E3 – E3
2 E3 – E3 – E3 – D#3 – D#3 – E3 – E3

I am not suggesting that 'Levels' is a copy of 'Twinkle Twinkle Little Star.' Instead, I want to draw attention to the similarities in composition. Both feature a simple repeating rhythm, and both have a rhythmical pitched melody. The notes move either up or down in pitch, then the opposite, and then back again. Equally important is the distance between the different notes. In almost every case of a dance music motif, the notes will remain within a single octave, and remain close together. That is, the intervals between each note are not massively separated. It is far easier for us to remember stepwise motions than it is to remember leaps across a large number of notes. A great motif should be natural for anyone to play with just one hand.

BINARY PHRASING

The final essential element for any great motif is variation, and this is created through binary phrasing. As touched on already, a binary phrase consists of two rhythmical or melodic ideas of equal duration but with pitch or rhythmical differences in the second phrase from the first. This was explained with the previous example of 'Levels,' but it does not always need to occur in the lead instrument. For instance, if we examine Eric Prydz' 'Pjanoo' (composed in G minor), it is the bass phrase and not the lead that provides the phrasing.

However, the bass plays:

G – G – F – F – D# – D# - D# – D# – D# – F – C – C – C

followed by:

C – C – D – D – D# – D# – D# – D# – D# – F – F – G – G – G

Here, both parts of the bass employ the exact same rhythmical motions and it is only the pitch that changes. Although there is a significant amount of pitch movement in this particular example (or at least there is for dance music), it is offset by the lead piano that consists entirely of a single phrase. It does this by maintaining the same rhythmic and pitch movements throughout all four bars of its motif.

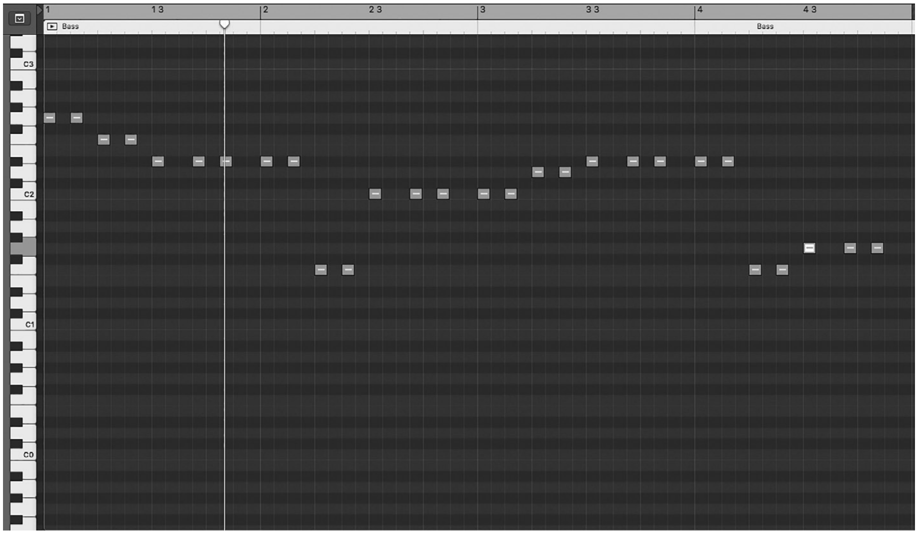

FIGURE 22.2
Eric Prydz' 'Pjanoo' bass

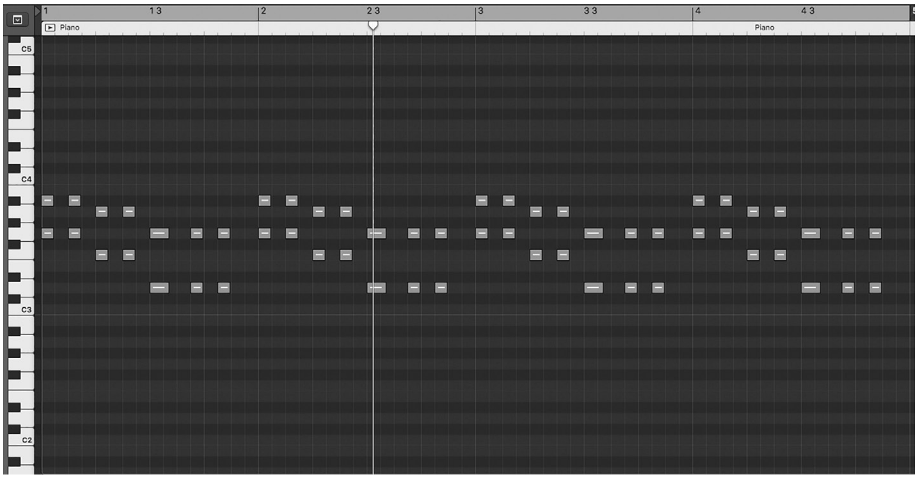

FIGURE 22.3
Eric Prydz' 'Pjanoo' lead

Binary phrasing is not limited to motifs and will occur within all other elements of the music. In particular, it will occur within rhythmical elements. This technique was employed in an earlier chapter on drum loops. A modulating filter was used to manipulate the frequencies on the hi-hats while pitch shifting and/or frequency modulation was employed on the snare.

Both of these techniques constitute a form of binary phrasing and result in the loop bearing sustained listening. Indeed, genres such as tech house, techno, and minimal often employ a more complex form by employing slow modulating effects and micro-scale tuning on the drum rhythms and single instrument hits.

This permits sustained listening of what *appears*, on the surface, to be little more than a repeating, sustained loop.

CALL AND RESPONSE

A similar technique to binary phrasing is call and response. This originates from African music and involves one instrument making the call and a secondary instrument responding. The secondary instrument is usually a different timbre than the first but is not always the case.

For example, in Tech House and Techno, it is common practice for the snares to exhibit the call and response behaviour. Here, typically a percussive element will play a specific pattern in the first bar that is answered by the same percussive element responding to this call with a slightly different pattern in the second bar.

DMM resource: Example of percussive call and response.

In Dubstep, two different instruments are employed to produce the call and response. These will consist of two different styles of bass timbre but some may use a bass to provide the call and this receives its response in the form of a higher pitched lead sound. In the DMM resource, the call is provided by the first bass and then responded by a secondary, different bass timbre.

DMM resource: Example of Dubstep track call and response.

There is no requirement for a call and response or binary phrase to occur one directly after the other and often the call may repeat a number of times before it receives its response and is resolved. This patience technique was employed in the House example of this book. In this example, the phrase is repeated three times before it is finally resolved.

DMM resource: House riff example.

SOUND DESIGN

Producing a great motif is only half the battle, because sound design plays an equally important role. Most tracks will employ a different style of lead, and so there are no definitive guidelines for their creation. But while it's not possible to suggest ways of designing a lead to suit any one particular track, there are some generalizations that we can apply.

Almost all lead instruments will employ a fast attack on the amp and filter envelope so that the sound starts immediately with the filter introducing the harmonics to help it to pull through a mix. After this, the amplitude, decay, sustain, and release will depend on the lead you want.

If you want to create a timbre that exhibits a pluck, this is produced by modulating a filter with an envelope decay. The amplitude envelope would typically begin with a fast attack, medium decay, full sustain, and a medium release. Employing a saw wave or pulse wave oscillator (or a mix of both) will give the filter some harmonics to act on. The filter should be set to a low-pass 12 dB or 24 dB with the cut-off set about halfway. Finally, an envelope is used to modulate the filters cut-off frequency. Here, it should employ a fast attack, zero sustain, and release and the decay and modulation send amount should be adjusted to your preference. The shorter the decay parameter (or, the lesser modulation amount applied), the more pluck the timbre will exhibit. If the decay is too long, the filter will not close as quickly, and the pluck effect is reduced.

These simple guidelines produce the basis of *all* plucked style timbres so the only real defining elements will be your choice of oscillators, how they're stacked (unison), and how you choose to detune them against one another.

To show this in action, let's design a plucked timbre similar to that used in Zhu's 'Faded.' This is a bright pluck that can be programmed on any subtractive synthesizer:

1. To create a pluck, we require harmonics for the filter to sweep. In the track 'Faded' by Zhu, it exhibits a reasonably bright tonal character that we would typically associate with saw waveforms so we'll use this as the basis for the sound.
2. A single saw doesn't produce enough harmonics so several should be stacked together; this can be accomplished through unison parameters. If possible, we need each saw to start at a different phase so switch off any retriggering.
3. In the track, the sound starts immediately and exhibits a sustain portion as the filter sweeps the frequencies. The timbre does not die immediately but fades away so it would employ an amplitude release. A good starting point for this sound would be to use an immediate attack, a fast decay, full sustain, and a medium release.
4. The pluck, although bright, is not excessively so. This would lean towards a 24 dB filter rather than a 12 dB. This is because a 24 dB filter exhibits a faster slope, so the brightness of the filter decays more quickly when modulated with an envelope.
5. Adjust the filter cut-off so that it's around a quarter of its full open value and keep the resonance fully closed.
6. Using a modulation envelope, set the attack and sustain to zero and employ a quarter length release.

7. Finally, send the modulation envelope to the filter by a small amount and then increase the decay. You'll reach a point where the pluck is at its strongest. This is the setting to remain at.
8. Finally, the sound would benefit from some EQ, chorus, delay, reverb, and phaser effects to polish the final result.

The alternative to a plucked lead is the direct lead. This can be programmed in a multitude of ways using analog, wavetable, frequency modulation, or a mix of all three. The approach is to construct a harmonically rich sound that can be filtered and modulated with the envelopes and LFOs. These last modulation options play a vital role in producing a lead timbre because they require plenty of textural momentum to maintain interest. In particular, a minimal amount of pitch modulation to the oscillators at the transient of a note is a widespread effect.

To show this in action, we'll design a lead typical of the artist Disclosure. Listening to the leads, they're typically bright in harmonics and dissonant:

1. To make the sound bright, we require high harmonic waveforms so we can start with three sawtooth waveforms.
2. The sound is a lead so wants to begin immediately but hold for the duration that the note is depressed, so start with a fast attack, no decay, full sustain and a short release.
3. To produce the dissonance in the sound, we need to tune the saws away from one another. Detune one saw to +7 semitones and another to +15 semitones (this is up by an octave plus another three semitones). This will produce a harsh rasp from the synth.
4. Static sounds will send our listeners to sleep, but we don't always have to employ modulation for movement, often effects such as flangers or phasers can add plenty of action. Use a phaser with the LFO set to 2/1 with high feedback.
5. Finally, apply some EQ, delay, and reverb to polish the sound.

Most genres of dance music will feature rich and deep sounds, and, for these, it is important to layer the leads. Layering different synthesizers to produce a single cohesive tone is a mix of both art and science. Typically, a lead will consist of three layers: the component layer, the energy layer, and the HF layer.

The layering process for synthesis is similar to layering a kick drum. You start by designing the component layer; this is the overall tone and style of the sound you want in the music. The component layer for leads usually seats in the mid to upper mid frequency range so the energy layer should be used to add body and weight to the middle and lower mid frequencies. This is best approached by listening to the component layer while designing a sound on a secondary synthesizer. Sometimes you may need to pitch the motif of the energy layer down by −3, −5, −7, −9, or −12 semitones so that it adds the required depth to the sound.

Using the filters on the synth, you can sculpt the overall sound to complement and fit with the component tone. It is essential to use different amplitude and filter envelopes for these two layers so that the whole timbre is a result of the two morphing with one another regarding harmonics and dynamics. Movement is a crucial factor with any instrument and employing a fast attack with a slow release or decay on one while using a slow attack and a rapid decay or release on the other will produce an exciting lead.

Once the two layers work together, you can add the third HF layer. This contributes to the upper mid and high-frequency range of the instrument. Again, using the same approach as mentioned above, you can pitch the motif up (if required) and also employ a different amplitude and filter envelope than the previous two layers. If this particular layer is too heavy, you can use the synthesizer's filters or EQ to reduce its content.

By layering a sound in this way, the mid frequencies of the component sound mix with the middle of the energy layer, while the upper mid of the component mix with the HF layer. All the while, the energy and HF layers are adding low- and high-frequency energy to the sound. This approach results in a sonically complex, interesting sound that is compressed and treated with EQ and effects chains. While you can apply effects individually to each synthesized layer, it's advisable to leave effects such as reverb, delays, and compression to be administered via a bus.

Alongside layering, artists may also apply doubling. Here, rather than use timbres from some synthesizers, you would duplicate the track (MIDI and instrument) and then transpose the copied MIDI track up by a third, fifth, seventh, ninth, or an octave. Finally, effects and processing chains also play an essential role in creating a lead timbre. The most typical effects are reverb, delay, phasers, flangers, distortion, and chorus along with noise gates, compression, EQ, and envelope modifiers.

What follows are some simple examples of how to design a few lead sounds. As with the previous chapter, this is not intended to be a comprehensive guide. Instead, its purpose is to give you some hands-on experience with a synthesizer and to then develop on these ideas further through experimentation.

Euphoric Trance lead

The uplifting Trance lead is probably the most elusive lead to design, but in many cases, this is just because it requires layering. A basic Trance lead timbre can be constructed with four oscillators. Two employ pulse or saw waveforms, while the third is often a saw wave and the fourth, is noise to add some high-end harmonics. The two pulses (or saws) are detuned from one another by a varying amount until a thick, harmonic sound is created.

You could detune these to the point where they are just about to separate into two distinct timbres. The saw wave is detuned from these and unison is applied to thicken the sound further. Finally, white or pink noise is added to produce the high harmonics of the sound. This should be set to a lower volume than the other oscillators otherwise it'll be too prominent.

The amplifier's attack should be to zero with a medium decay, a short sustain, and a rapid release. This release can be lengthened or shortened further to suit the programmed melody. The filter envelope can employ the same settings as the amplifier, although it is worth experimenting with the decay to produce a sharper pluck. The filter key follow should be applied so that it tracks the pitch of the notes being played and treats the filter respectively. A 12 dB low-pass filter with a low to mid resonance along with a mid to high cut-off will produce the basic timbre.

Further interest can be applied to the timbre through pulse width modulation but if at all possible, two LFOs should be employed so that each width can be modulated at a slightly different rate. Here, the first LFO modulates the pulse width of the first oscillator with a sine wave set to a slow to medium rate and full depth, while the second modulates the pulse width of the second oscillator and employs a triangle wave with a slighter faster rate than the first.

A typical Trance consists of three layered instruments to apply the above techniques again, but change the tuning and some waveforms to produce slightly different sounds that add to the whole.

Place all three instruments in a group track and apply both reverb and delay. The reverb should be a room or hall setting and used as a send effect. This should employ 50ms of pre-delay so that the transient pulls through unmolested and the tail set quite short to prevent it from washing over the successive notes. You may want to follow this with a noise gate to remove the subsequent reverb tails, as this will prevent the sound from moving to the background and create a more significant sound.

Delay is also applied, but again this is commonly used as a send effect so that only a part of the sound is sent to the delay unit. The settings to use will depend on the type of sound, but the delays should be set to less than 30ms to produce the granular delay effect to make the timbre appear prominent in the mix.

TB303 leads

Although we have already discussed the use of a TB303 during bass programming, it remains a versatile machine and is equally at home creating lead sounds by merely pitching the bass frequencies up by a few octaves. The most notable, if old school classic, an example of this was on Josh Wink's 'Higher State of Consciousness.' This same effect can be recreated on most analog-modeled synthesizers as it only requires the use of a single saw oscillator.

With the amplifier's attack, sustain, and release set to zero, the decay can be adjusted to suit the style of sound you want to achieve. A 12 dB low-pass filter is employed with a low cut-off setting, but the resonance set just below self-oscillation. The filter envelope is a copy of the amplitude envelope's setting although the decay should be set just short of the amplifiers decay stage. Filter key follow is often employed to create additional movement, but further modulation is usually applied through velocity modulation of cut-off so that the harder the note is struck, the more the filter opens.

Finally, the timbre is run through a distortion unit, and the results are filtered with a plug-in filter. Depending on the melody, it may also benefit from increasing the amplifier's decay so that the notes overlap one another and employing *portamento* on the synthesizer, so the notes slur into one another when played.

Distorted leads

There are hundreds of distorted leads, but one that often appears is the distorted/phased lead that is often employed in some of the harder House tracks. The basic patch can be created with a single saw oscillator, but typically, two saw oscillators detuned from one another produce a broader, more upfront style of timbre.

The amp envelope is set to a zero attack with a full sustain, a short release setting, and the decay to around a quarter. As is common, it is this decay that will have the most influence on the sound, so it is worth experimenting with increasing and decreasing to produce the style required. The filter is a 24 dB low-pass with the cut-off quite low, and the resonance pushed high to produce overtones that can be distorted afterward with effects. The filter's envelope is typically set with zero attack, sustain, or release, but the decay is set slightly longer than the amplifier's decay parameter.

Distortion followed by phasers or flangers are commonly applied, but the distortion should come first in the chain so that the subsequent phaser is also subjected to the distortion. It is important here not to use too much distortion, otherwise, it may overpower the rest of the mix. Ideally, you should aim for a subtle but noticeable effect.

Hoovers

The Hoover sound originally appeared on the Roland Juno synthesizers and has featured consistently throughout all genres of dance music including Techno, Tech House, Acid House, and House. In fact, these remain one of the most popular sounds employed in dance music today. Initially, in the Juno this was somewhat aptly named '*What the …*' but due to their dissonant tonal qualities, they were renamed 'Hoover' sounds by dance artists since they share similar tonal qualities to those of a vacuum cleaner.

The sound is best constructed in an analog-modeled synthesizer since the tonal qualities of the oscillators play a significant role in creating the right tone. Two saw oscillators are detuned as far from one another as possible but not so far that they become two individual timbres. The amplifier's attack is commonly set to zero with a short decay and release and the sustain parameter set to just below maximum, so there is a little pluck evident from the decay stage. The filter envelope employs a medium attack, decay, and release but no sustain is applied. This augments a 12 dB low-pass filter with the cut-off set quite low and a medium resonance.

To add the dissonant feel that 'Hoovers' exhibit, the pitch of both oscillators are modulated with a pitch envelope set to a fast attack and a short decay. This, however, should be applied negative rather than positive so that the pitch bends upwards into the note rather than downwards. Finally, depending on the synthesizer recreating the timbre it may need some widening, and this can be accomplished by washing the sound in a chorus or preferably stacking as many voices as possible using a unison mode.

'House' pianos

The classic House-style piano is making a comeback in some forms of Deep House and Tech House, and this is direct from the Korg M1 (preset Piano 16'). This is usually left unmodified and treated with a series of effects chains including reverb, noise gates, compression, EQ, and delay. The Korg M1 is currently available as a software instrument, so you don't have to purchase the hardware (and this is probably why the piano is becoming popular again).

Organs

Organs are commonly used in the production of House with the most frequent choice being the Hammond B-4 drawbar organ (as used in DeadMau's 'Ghosts 'n' Stuff'). The general timbre, however, can be emulated in any subtractive synthesizer by applying a pulse and a saw oscillator.

The amplifier envelope employs a zero attack with a medium release and maximum sustain (note that there is no decay since the sustain parameter is at maximum). A 12 dB low-pass filter is employed to shape the timbre with the cut-off set to zero, and the resonance increased to about halfway. The sound doesn't require filter modulation, but if you need a 'click' at the beginning of the note you can turn the filter envelope to maximum but the attack, release, and sustain parameters should remain at zero with an abrupt decay stage. Finally, the filter key follow is adjusted so that the filter tracks the current pitch that will produce the typical organ timbre that can then be modified further.

GENERAL PROGRAMMING TIPS

- Avoid relying on one note while programming; it will not give you the full impression of the patch. Always try to play the motif to the synthesizer before programming.
- Ears become accustomed to sounds quickly, and an unchanging sound can become tedious and tiresome. Introduce sonic variation into extended timbres through the use of envelopes or LFOs augmenting the pitch or filters.

- The simpler the motif, the more movement the sound should exhibit. So for basses and motifs that are quite simple assign the velocity to the filter cut-off so the harder the key is struck, the brighter the sound becomes.
- Don't underestimate keyboard tracking. This can breathe new life into motifs that move up or down the range as the cut-off of the filter will change respectively.
- Although all synthesizers share the same parameters, they do not sound the same. Merely copying the patch from one synthesizer to another can produce completely different results. This can be useful when layering.
- The noise oscillator can seem useless when compared to the saw, sine, triangle, and square waves but it is one of the most valuable. It is used for everything from Trance leads to hi-hats so don't discount it.
- To learn more about the character of the synthesizers you use, dial up a patch you don't like, strip it down to the oscillators, and then rebuild it using different modulation options.
- If you have no idea of what sound you require, set every synthesizer parameter to maximum and then begin lowering each setting to sculpt the sound into something you like.
- Many bass sounds may become lost in the mix due to the lack of a sharp transient. Synthesize a click or use a woodblock timbre or similar to enhance the transient.
- The best sounds are created from just one or occasionally two oscillators modulated with no more than three envelopes: the filter, the amplifier, and the pitch. Try not to overcomplicate a patch by using all the oscillators and modulation options at your disposal.
- Always layer leads but use a different amplifier and filter settings on both. For example, using a slow attack but quick decay and release on one timbre and a fast attack and slow decay and release on another will 'hocket' the sounds together to produce interesting textures.
- Don't underestimate the LFO. A triangle wave set to modulate the filter cut-off on a sound can breathe new life into a dreary timbre.
- We determine a considerable amount of information about a sound from the initial transient. If you replace the attack of a tone with the attack portion of another, it can create exciting timbres. For instance, try replacing the attack stage of a pad or string with a guitars pluck.
- For more exciting transients, layer two oscillators together and at the beginning of the note use the pitch envelope to pitch the first oscillator up and the second one down (i.e., using positive and negative pitch modulation).
- For sounds that have an extended release stage, set the filters envelope attack longer than the attack and decay stage but set the release slightly shorter than the amps release stage. After this, send the envelope entirely to the filter so that as the sound dies away, the filter begins to open.

By the very nature of dance music, sounds are not only produced in a synthesizer and sound design through sampling other artists' records is equally commonplace. It's not uncommon for a dance artist to sample another artist's track to produce the basis of the song and for other artists to then sample that track. Skrillex, for example, has sampled numerous records and in turn, has subsequently been sampled by other artists. This style of musical 'recycling' is extensive and occurs throughout every genre of dance music from Trance to Techno to Minimal to House and Tech House.

The sampling can undertake numerous forms depending on the genre. It's not unusual for Tech House and Techno to borrow samples from other records, but here it is typically such small parts that are sampled and mangled that they become almost unrecognizable.

For instance, the artist may sample a guitar from an old 70s' record, time-stretch or time-compress it, apply EQ, reverb, and a variety of other effects chains to produce a single stab hit. On the other end of the scale, French house will likely sample eight to 16 bars of a complete record and repeat these over and over while applying numerous automation effects to create an arrangement.

Regardless of how you choose to approach sound design, it is a time-consuming process. There is no 'quick fix,' and it is a result of practical experience and experimentation. Many professional artists have admitted that some of their characteristic timbres were a result of experimenting and serendipity. Therefore, if you want to program great tones, patience and experimentation are the real key. You must be willing to set time aside from writing music and learn sound design.

CHAPTER 23

Composing and designing bass

Bass and drums are brothers in the basement, cooking up the groove that makes people move.

John Densmore

The drums and bass from the backbone of electronic dance music. In fact, there have been many records that consist of little more than the drums, bass and a vocal (i.e., *most Tech House and Techno*). For this chapter, we'll look at the general music and sound design theory for creating bass lines.

We have already discussed the creation of motifs at great length in the previous chapter and this same theory can equally apply to bass lines. So rather than reiterate it all again here, I would recommend referencing the previous chapter. That said, as the bass will typically underpin the groove and augment the lead, there are some different approaches we can take that we will examine in this chapter.

A bass line can be approached in two ways: you can compose the motif and then design a timbre to suit. Or, you can develop/find a great bass sample and compose a theme from that. Both approaches have advantages and disadvantages, but, in many cases, the method is determined by the genre.

Disco House, for example, benefits from a groovy melodic bass line. Often, the artist will employ a simple bass tone to compose the melody. Once the theme is down, she will search out/program a more suitable bass timbre. Similarly, as some trance genres employ very similar melodic styles of bass lines, it can be easier to program the melody first and then program a bass to suit that melody.

Alternatively, for genres such as House, Techno, and Tech House, you would often source a sampled bass – or design one – and drop it into sample playback software. This is because these bass lines are usually straightforward and rely more on the timbre than the melodic composition. Obviously, these approaches can work equally well vice versa and when it comes down to it, it's whatever you feel the most comfortable working with.

COMPOSITION

Composing a bass line obviously depends on the genre of music, but there are some general theories we can use to create them. Once we have these fundamental theories down, they can be further modified to produce the groove.

The most basic, yet still fashionable, bass is the root. Here, the bass follows the root note of the chords. If we take our example chord from the previous chapter in A minor, we have the following root notes in the chords: A > E > F > D. This formed what is known as the *optimistic* chord progression. For the bass, we would merely employ a sine or triangle wave (I prefer triangle waves for the additional harmonics) and play them an octave or two lower than the chord.

The amplitude envelope should be set to a fast attack, zero decay, maximum sustain, and short release. By doing so, the note is audible from key press and will hold for the length of the note. Some synthesizers may create an audible click on the transient of a sine or triangle; if this is the case, it can be removed by lengthening the amplitude attack. If you're using a sine wave, because it only consists of a fundamental tone, there is no need to employ a filter.

DMM resource: Drums and sine bass.

While the bass isn't particularly exciting, we can add some interest and excitement through the use of side-chain pumping. In the following example, I employed Xfer's LFO tool to add a curved action to the bass as shown in Figure 23.1.

DMM resource: Drums and sine bass (side-chain).

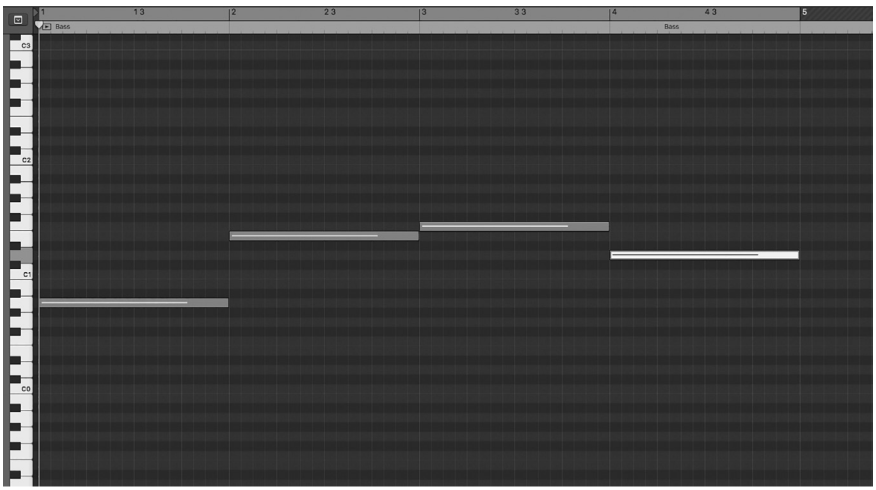

FIGURE 23.0
The root bass

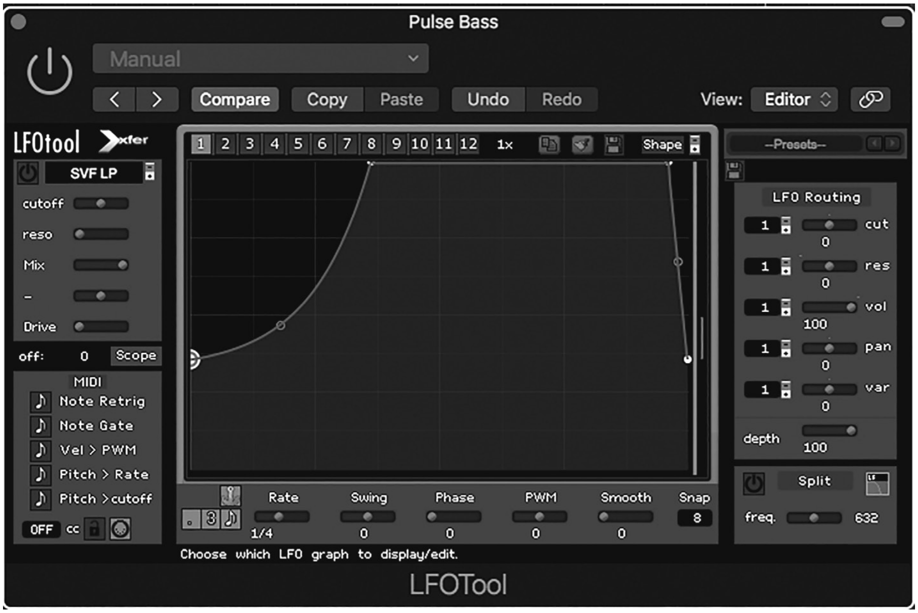

FIGURE 23.1
Xfer LFO tool

This style is commonly employed in both Tech House and Techno, although often the bass is augmented with effects or a further oscillator. This is because such a low-frequency tone isn't evident when played on a smaller speaker system. Consequently, many producers will layer an additional square or saw wave oscillator on top of the original wave. This employs the same amplitude envelope and is tuned to the same frequency or a fifth, seventh, or octave higher. Then, using a low-pass filter, the higher harmonics of the upper oscillator are removed until you have the sound you like.

If you want to add more interest to the sound, you could employ a modulation envelope to control the cut-off or resonance of the filter (or both). The modulation application is down to creativity but using a long attack and a short decay with no sustain or release on the filter; the harmonics would gradually increase as the note plays before dying away back into the sub. Or, with a fast attack and a slow decay setting, the note would start initially bright and decay down to the original wave. Provided the filter isn't fully closed, there will still be lower harmonics present after the modulation cycle has ended.

Rather than employ an additional oscillator, you could equally apply distortion, bit crushing or overdrive. This will add further harmonics to the sound so that it can be heard on smaller speaker systems.

Many styles of Techno employ a *nervous* bass; this is a low sine tone that modulates in pitch slightly producing an almost edgy tense feel to the sound. This can be created in a couple of ways. Perhaps the easiest is just to duplicate the bass note and play it an interval higher, so you have two pitches, an interval apart, playing the same melody. Because the two notes are so closely related, this results in a phase relationship occurring between the two that produces a rumbling sound. Alternatively, the sine wave could be treated to an LFO to modulate the pitch ever so slightly; this would provide a similar effect.

DMM resource: Two sine waves an interval apart.

Rather than employ a sine wave, a favorite technique is to employ a kick from the Roland TR808. If tuned low with a long amplitude attack and release envelope (to miss the transient and lengthen the timbre) it produces a bass tone similar to a sine wave but with a harder sonic edge. Some genres may just use the TR808 kick alone (with a fast attack and extended release) to act as both the kick *and* the bass.

OFFBEAT BASS

An advancement from the root bass is the offbeat bass. This again follows the root notes of all the chords, but this time the notes are a 1/8th (semiquaver) in length and occur in between the pulses of the kick drum. This composition style is shown in Figure 23.2.

It is essential for this style of bass to feature a prominent attack stage so that it can interact with the transient of the kick drum. There are a multitude of permutations for the bass, but the easiest is to employ a single saw wave. Set the amplitude envelope to a fast attack and medium decay with no sustain and a short release. The release can be timed so that the bass note just ends before the next kick occurs.

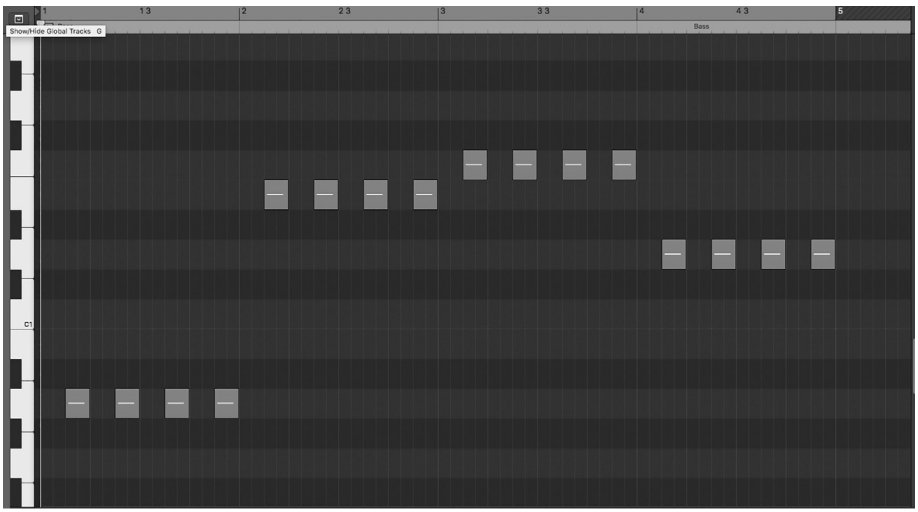

FIGURE 23.2
The classic offbeat bass

To create a hard transient, you can employ a modulation envelope on a filter to create the classic filter pluck. For this, set the filter cut-off and resonance to half way and then set a modulation envelope to a fast attack, zero sustain, and a short release. You need to set the modulation decay rate approximately 20% shorter than the amplitude decay and then send the envelope to the filter by 70%. This will produce a basic filter pluck that you can hear in the example.

DMM resource: Drums and offbeat bass.

With the primary sound, you can experiment by adjusting the decay and release of the filter modulation envelope, the filter cut-off amount, the modulation send amount, and the filter resonance. This will produce a variety of harmonic changes in the sound, so you just need to settle on what you prefer.

Once the basic sound is prepared, it can be augmented with further oscillators. For example, a sine wave tuned an octave lower will create a bass with more depth while the addition of a square wave at the same frequency (or tuned an octave below) will add body to the bass. You could also experiment with a pulse wave, modulating the pulse slightly with an LFO to add further interest into the timbre.

If wavetables are available on the synthesizer, you can experiment with tables that have a high harmonic content. Because these are also fed to the modulated filter, the pluck remains but the timbre will grow richer. It's just a case of experimenting with the oscillators/wavetables that are available in the synthesizer.

Alongside experimenting with oscillators, effects play an essential role in a plucked bass. Saturation, wave-shaping, and distortion can all add further character to the bass. Also because of the plucked nature of the bass, delays can work particularly well. A 1/8th or 1/16th note delay (or both in a *ping pong* configuration) with short feedback can add a bouncing action to a bass line, although for this to sound coherent the amplitude envelope of the bass should be shortened.

Reverb can also help with bass. Provided a long pre-delay is employed so that the reverb bypasses the transient and the decay is kept short, it can give an additional exciting character to the sound.

DUAL-BEAT BASS

A step further than the offbeat bass is to add a bass note to occur underneath the kicks. In this application, the delay is usually set to a 1/16th, and side-chain compression (or LFO tool) is used to duck the note underneath the kick. This way, the bass moves out of the way of the kick, but the delay occurring on the bass is present directly after the kick.

For this to work correctly, the bass fundamental frequency should either sit above or below the kick's fundamental frequency and the side-chain effect

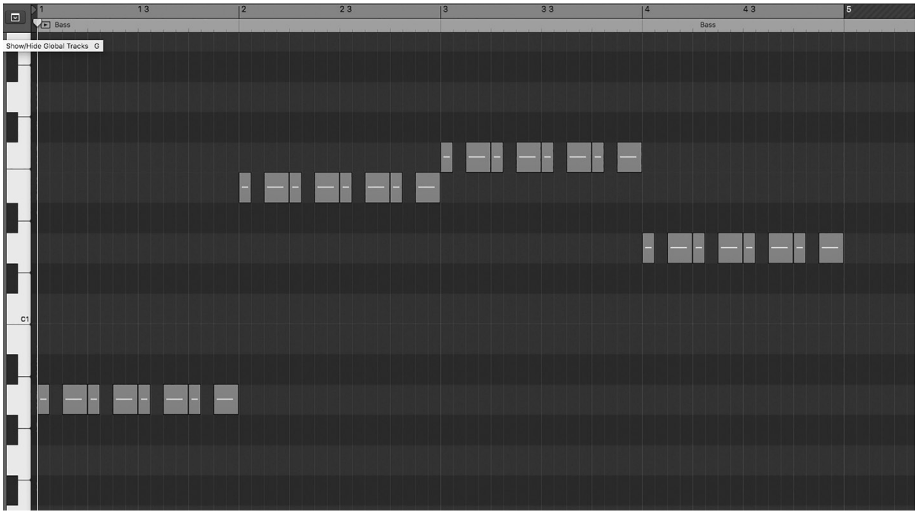

FIGURE 23.3
The dual-beat bass

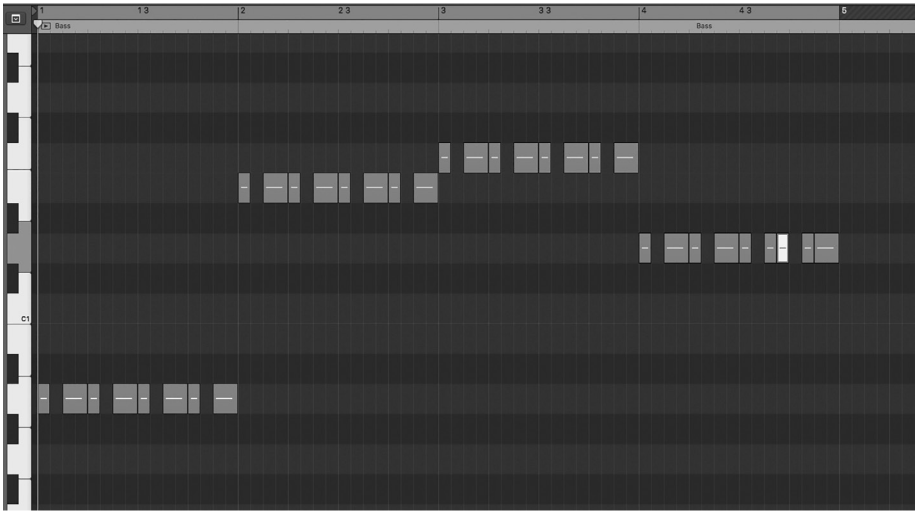

FIGURE 23.4
Adding rhythm to a bass

should occur before the delay, not after. If the side-chain effect occurs after the delay effect, then the delays will also be treated to *ducking*. However, if the side-chain happens before the delays, the subsequent delays will pass unmolested.

To make the bass more interesting, we can change the rhythm of some of the notes. The first place to add rhythm is usually at the end of every two or four bars where we would typically introduce a new instrument or make a change to the arrangement. This can be a simple as adding a couple of shorter notes to occur in between the offbeat bass or by moving the position of some of the offbeat bass notes.

So far the bass notes have been changing pitch to follow the root note of each chord, but this isn't always necessary. In many genres, the bass will remain at one pitch throughout. This is sometimes called a *pedal* bass. While the chord and other instrumentation move through different pitches, the bass remains at either the root note of the scale or plays the fifth note of the scale. The produces a repeating, rhythmical bass line that reinforces the existing groove.

If you take this approach, it's advisable to add some form of modulation to bass to make it more interesting to the ear. The simplest way to achieve this is to map velocity to filter and employ velocity changes in the MIDI. By doing so, the bass timbre will change and maintain some interest to the listener. A proper technique here is to syncopate the velocity of the bass throughout the different bars. For example, the bass could consist of medium velocity with high-velocity values only occurring on every second offbeat. But as the song progresses, the high velocity could change to happen on every first offbeat. Although this change is small, it adds an exciting syncopation to the music.

If employing a pedal bass, it's sometimes beneficial to shake things up a little, particularly in places where a chord may hold for longer than before. This could occur during a tense part of the music such as before or during the breakdown, or perhaps after a small break and build in the music. Here, you would play the root note of the chord with the bass but after the break, move the bass to the fifth, or the octave above, for a few bars before eventually returning to the root. This style of movement was one of the defining elements of disco and was the precursor to the octave shifting bass. Here, the bass alternates from the root note to an octave above and back again. As disco formed the genesis of dance music, these bass techniques are still employed in dance music today.

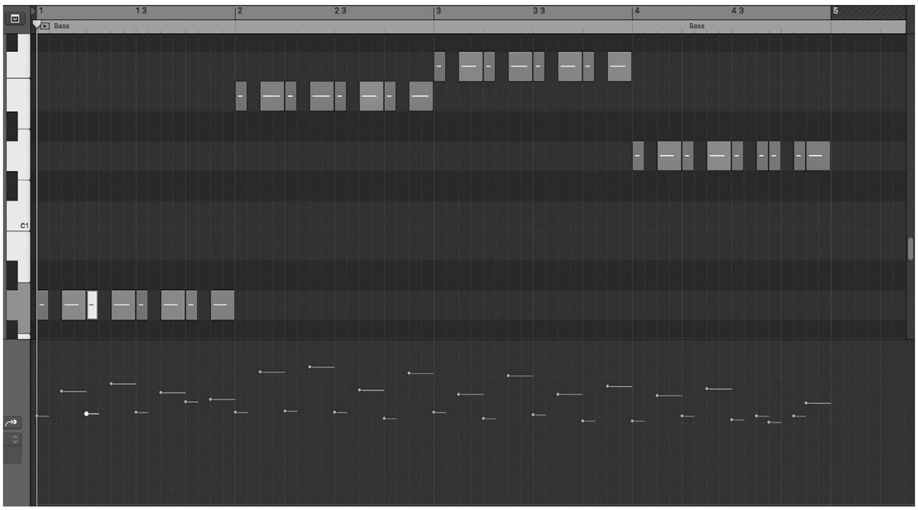

FIGURE 23.5
Adding velocity syncopation to a bass

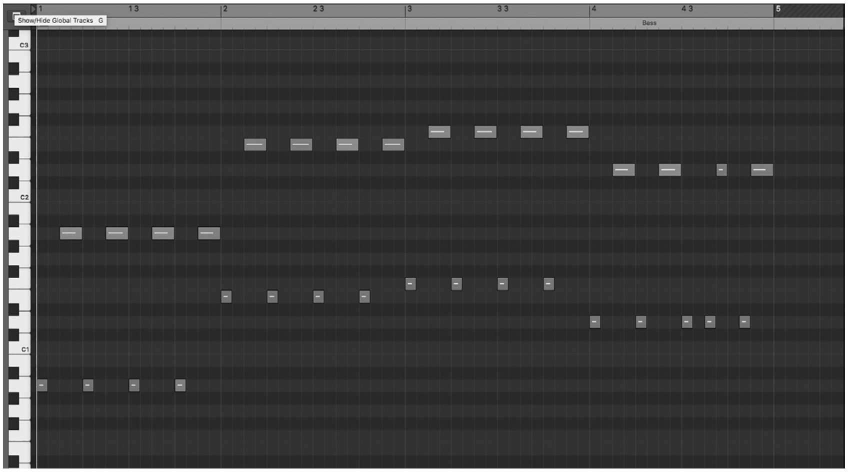

FIGURE 23.6
The classic Disco bass line

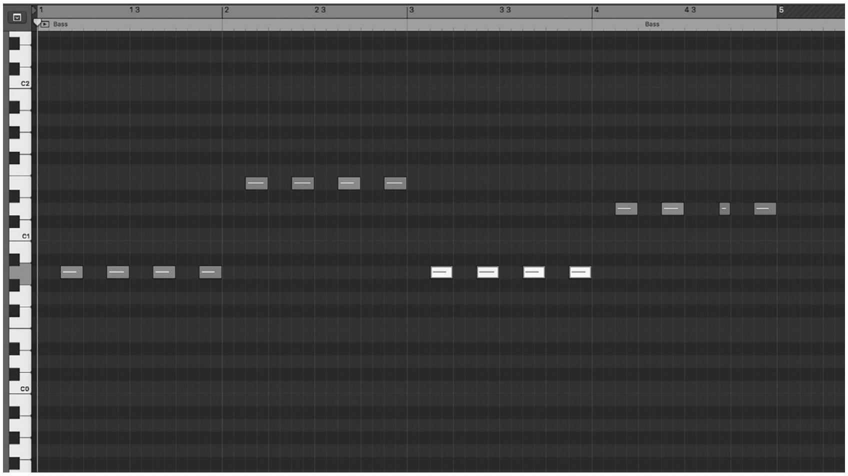

FIGURE 23.7
A slash chord bass with the chord roots of A – E – A - D

A step further is to employ *slash chords*. For this, rather than have the bass play the root note of each chord, it will occasionally play an alternative note in the chord. So, it might play the third, fifth, or seventh note of a chord rather than the root. For example, take the chord progression A > E > F > D in A minor:

- Chord 1 notes = A – C – E
- Chord 2 notes = E – G – B
- Chord 3 notes = F – A – C
- Chord 4 notes = D – F – A.

In a typical application, the bass would follow the root note of the chords, playing the A – E – F – D. With a slash chord, the bass may play A – E – **A** – D. Note how on the third chord, the bass plays the third note of the chord rather than the root. Just this small change can add funk to a bass line and increase its interest.

One further option than the slash chord is the walking bass line. This is perhaps the most complex bass to compose and is typical of the genres where the focus is on a definite groove. This involves moving up (or down) in chromatic steps between the chord's root note. Using the previous example of A – E – F – D, each chord lasts for two bars. To produce a walking bass line for this progression, you could play the beginning A for one bar, then, in the second, play B and D in half notes. Then play the E for two bars, and then the F for a bar and a half before playing two quarter notes at F and E to walk down the D.

With this approach, experimentation is the key. Depending on the genre of music, the walking may sound better occurring on the weaker beats (second and fourth) rather than the stronger (first and third). Timing and velocity both play an important role, so you will need to move notes around to produce the right feel.

Samples of a real bass guitar are best suited to the walking bass line. These are available with most samplers, and many feature numerous articulations to create a realistic sound.

The trick to programming a convincing real bass is to take note of how they're playing and emulate this action with MIDI and a series of CC commands. Bass guitars use the first four strings of a regular guitar, E – A – D – G, that are tuned an octave lower. This results in the E being close to three octaves below middle C. They are monophonic instruments, so the only time notes overlap is when the resonance of the previous string is still dying away as the next note is plucked. You can copy this action by leaving the last note playing for a few MIDI 'ticks' after the following note in the sequence has started.

The strings on a bass are either plucked or struck, but the two techniques produce different results. If the string is plucked, the sound exhibits a brighter transient and has a more extended resonance than if it were merely struck. To emulate this action, map the velocity to the filter cut-off so that higher values result in a filter that opens more. Not all notes will be struck at the same velocity, however, and if the bassist is playing a fast rhythm, the following notes will have less velocity because the talent has to move his hand and pluck the next string quickly.

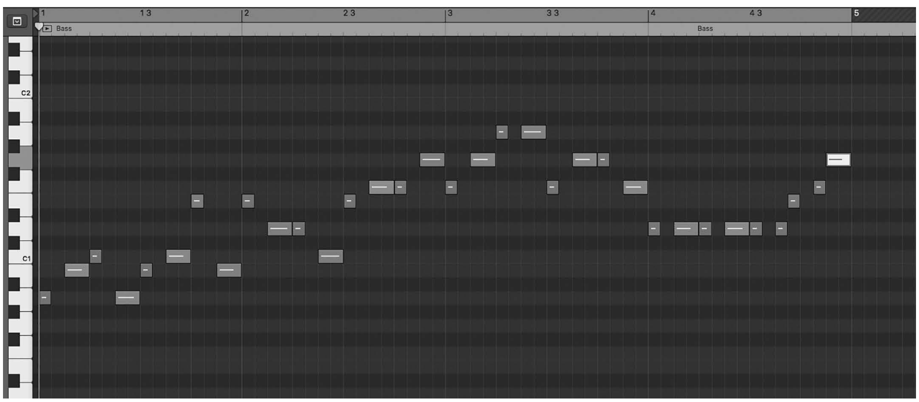

FIGURE 23.8
A walking bass line

Some bassists may also use a technique called 'hammer on.' Here, they play a string and then hit a different pitch on the fret. This results in the pitch changing but not accompanied with another pluck of the string. For this effect, you'll need to make use of pitch bend. This should be set to a maximum bend limit of two semitones because guitars don't 'bend' any further than this.

Program two notes, for instance, an E0 follow by an A0 and leave the E0 playing underneath the successive A0 for around a short period. Then just before the second note occurs, program a pitch bend message to bend the tone up to A0. If this is programmed correctly, on play back, you'll notice that as the E0 ends, the pitch will bend upwards to A0 simulating the effect. In addition to all this, it's also worthwhile employing some fret noise and finger slides. Most good sample-based plug-in instruments will include fret noise that can be dropped in between the notes to emulate the bassist's fingers sliding along the fretboard.

CALL AND RESPONSE

The final bass is the call and response. This is typical of Dubstep, Future House and House alongside many other genres. It consists of one timbre playing a motif that is answered by a second, more straightforward motif, often with a completely different tone and intensity. In some cases, the motifs are very simple consisting of little more than two successive notes creating a call and a single note producing the response. In these instances, the timbres are often layered and processed to produce exciting textures to the ear. You can hear an example of a call and response in Dubstep on the DMM resource file.

> DMM resource: An example of Dubstep call and response.

The interplay between the bass and the kick form the foundations of your groove so ideally when designing a bass you should have the kick present. This way, you can hear how the bass interacts with the kick drum.

For most bass timbres, the amplifiers attack should be short so that the note starts on key press and the decay is employed as a release (sustain and release can be used but for many basses, an AR envelope is enough). The filter's envelope is also set in a similar method because if the filters attack stage too long, there will be no transient.

The body of the bass should exhibit some textural or pitch momentum. No matter how energetic the bass melody, if the tone features no movement the ears will turn off. In fact, this lack of change is a contributing factor as to why some dance rhythms appear not to exhibit any real groove.

Movement in bass can be accomplished in different ways: Dubstep employs pitch commands to create warping moving basses while genres such as Deep House, Techno, and Tech House will use side-chaining techniques. Some producers will often assign the modulation wheel to control parameters such as filter cut-off, resonance, LFO rate, or pitch. By doing so, the wheel can be moved in real time to experiment with sonic movement.

Although not always necessary, you may need to layer the bass with further timbres. Different layers are better sourced different synthesizers so that each continuous layer exhibits a different sonic signature, but this is not always necessary. The most common reasons for layering a bass are if it's lacking bottom end weight, if the artist wants to employ more of the stereo field, or if it needs more character.

If the bass is missing the low end, the simplest solution is to layer a sine wave, triangle wave (my favorite), or a pulse wave underneath. If employing a triangle or pulse wave, you will need to use a low-pass filter on them, but any of these pitched an octave lower than the bass will always add the necessary low-end energy and detail.

On the other end of the scale, if the bass is lacking in high-end energy, you can layer a saw or pulse wave over the bass. Here, a high-pass filter should be used to remove the low-end power from the timbre so that only the upper harmonics remain. Further modulation such as LFO or modulation envelope to filter resonance or cut-off can add additional exciting character to the layer.

Some dance tracks will employ a stereo bass, but this must be approached with caution. Low-frequency energy should be shared by both speakers to ensure accurate reproduction, and therefore any frequencies below 150Hz should not be stereo. Some plug-ins – such as the brainworx_bx_XL V2 – allow you to select a crossover frequency whereby all frequencies below this become mono while

FIGURE 23.9
brainworx_bx_XL V3

those above remain in stereo. But an alternative approach is to employ a mono bass layered with a stereo bass. The typical configuration here is a mono sine or filtered triangle/pulse to provide the low-end energy while saws and pulse waves are layered on top to produce the stereo image.

As mentioned, to prevent the layered timbres from becoming overpowered for the mix, in many instances, you should employ different filters on each synthesizer. For example, if the first synthesizer is producing the low-frequency energy, then it should use a low-pass filter to remove all the higher frequencies. Similarly, the second upper layered synthesizer should employ a high-pass filter to remove the lower frequencies. The removal of frequencies means both timbres are less likely to interfere with one another.

Alongside layering, effects and processing chains play a central role in the creation of bass timbres. Typically, these consist of distortion followed by compression then EQ and finally flangers or chorus-style effects. The distortion is usually employed to create additional upper harmonics to provide the bass with presence while the following compression is used to control the dynamics and prevent any peaks or unwanted distortion.

The EQ follows compression because this can be used to sculpt the distortion into the required texture, while flangers and chorus can be used produce more textural variation. In this configuration, the bass is commonly sent to the effects bus, so you can carefully mix the original bass weight with the affected results.

SOUND DESIGN EXAMPLES

What follows are some simple examples of how to design a few bass sounds. This is not intended to be a comprehensive guide or to cover the latest sounds (which would probably be outdated before the book is released). Instead, its purpose is to give you some hands-on experience with a synthesizer and to then develop on these ideas further through experimentation.

Sub bass

The sub bass is created through a sine wave, triangle wave, or square wave. My personal favorite is the triangle wave because it features more harmonics than the sine and with nothing more than a filter, you can sculpt it into an excellent sub bass that also exhibits some character.

Set the amplifier's envelope to a fast attack, medium decay, no sustain, and a quick release. Note that some synthesizers may produce an audible *click* if the attack is set to fast, so you may need to lengthen it to reduce any audible artifacts. The decay setting will depend on what style of sound you require.

If you want to add more interest to the sound, try modulating the sub-pitch with an envelope generator set to a slow attack, medium decay, and no sustain or release. By doing so, the note will bend slightly each time it is played.

Alternatively, if using a pulse or triangle, you can use a modulation envelope to control the filter cut-off to sweep the harmonics as the note plays. A good starting point for this is a fast attack, medium decay, no sustain, and no release.

Deep heavy bass

Start with a sine wave oscillator and modulate its pitch with an attack-decay envelope. Employ a fast attack and medium decay setting. If the transient is lacking, add a second pulse wave oscillator and pitch it down to suit the first oscillator.

Moog bass

For this, use either a sine or triangle wave mixed with a square wave. The oscillators are detuned from one another until you reach an impressive tonal character. Set the amplifier envelope to a fast attack medium decay with no sustain or release. For the filter, use a 12 dB low-pass filter set to midway with a low resonance. Using an envelope to modulate the cut-off of the filter, use a fast attack with a medium decay. By lengthening the attack or shortening the decay of the filters envelope, it is possible to create a timbre that 'plucks' or 'growls.' Filter key follow can be employed if the bass melody is particularly energetic in pitch, but, if not, some character can be applied by modulating the pitch of one, or both, of the oscillators with a sine wave LFO set to a slow rate.

Sweeping bass

The basic sweeping bass was typical of UK garage but has recently been morphed by the producer to form the atypical Dubstep-style bass. It consists of a tight yet deep bass that sweeps in either pitch or frequencies depending on the style required by the producer.

This bass can be created with two oscillators, one set to a sine wave to add depth to the timbre, while the other is set to a saw oscillator to introduce harmonics that can be swept with a filter. These are commonly detuned from one another, but the amount varies depending on the type of timbre required. So grab that detune parameter and keep going until it produces a thick pleasing sound.

The amplifier can be set to a fast attack and no sustain, but the release and decay should be set about midway. The decay setting here provides the 'pluck' while the release can be modified to suit the motif being played from the sequencer.

Start with a low-pass 12 dB and set the filter cut-off and resonance to the midway point. Using a filter envelope, set it the same as the amplitude envelope and then experiment with reducing the decay setting. Finally, positive filter key follow should be employed so that the filter will track the pitch of the notes being played, which helps to add more movement.

Further movement can be applied to the timbre via LFO modulation. Here a saw provides the best results so that the filter opens, rather than decays, as the note plays. The depth of the LFO can be set to maximum so that its applied entirely to the waveform and the rate should be configured so that it sweeps the note quickly. If the melody consists of notes being played in quick succession, it's prudent to set the LFO to retrigger on key press, otherwise, it will only sweep appropriately on the first note, and any successive notes will be treated differently depending on where the LFO is in its current cycle.

CHAPTER 24

Sound effects

Music is the only noise for which one is obliged to pay.

Alexander Dumas

Sound effects (FX) are perhaps the most underrated component of all styles of electronic dance music. They not only play an essential role in a musical transition, but they also add atmosphere, interest, and motion to the record. However, they often go unappreciated because their function goes mostly unnoticed by the listener. If they're done correctly, they shouldn't draw any attention to themselves, but if they're removed, a record will appear jarring or incomplete.

If you listen intently to any electronic music track, and specifically take note of the background sound effects, you'll notice these will occur every four bars at a minimum. The audio example in the resource pack features the same track, with and without effects, and this should give you a good idea of just how crucial sound effects are for music.

DMM resource: An example of a track with and without sound effects.

There are countless forms of sound effect and plenty of sample content available to purchase. These feature various effects from sweeps and risers, to falls, explosions, impacts, and everything in between. While typically many producers will use effects directly from sample content, a growing number of producers are creating their own to separate them from the crowd.

Techno and Tech House producers are leading the way in this regard. Many of these producers own handheld/portable digital recording devices. These are digital stereo recorders that are capable of recording audio in various formats (usually 24-bit Wav or Mp3) to store direct onto an SD card. Typical models are the Tascam DR05, Roland R07, the Olympus LS100, and the Zoom H5. The Zoom

FIGURE 24.0
The Zoom H5 handy recorder

H5 is one of the more popular choices because you can detach and swap the microphone heads to suit different recording environments. Of course, it is possible to record audio directly to most smartphones, but the quality from a smartphone is rarely consistent, or of high quality, so it is worth investing in a portable recorder.

Recording sounds from the real world is known as field recording, and sounds that are recorded are called *found sounds*. This is a new approach to classic sampling and was discussed in an earlier chapter, but, here, rather than record into a sampler, audio is recorded via the handheld device and transferred into the DAW for editing.

You can record anything and *everything* from the real world. Carnivals, traffic noise, or general chatter in a coffee shop through to bathroom fans and the sound of striking objects are all excellent sources. If it sounds interesting (which most sounds from nature are, of course), they can be recorded. It just takes the effort to actively listen to the real world and a willingness to experiment.

After recording, these sounds are transferred to the DAW for further editing. The editing techniques are as diverse as the found sound. Sounds can be time-stretched or time-compressed, EQd aggressively, and subjected to various effects and processors. There are no guidelines to this; it is merely a case of experimenting with the DAW's editing functions and any effects and processors you own. The resource pack for this book contains some 'before and after' sounds that I recorded and then distorted with plug-ins from Soundtoys and Sugarbytes.

DMM resource: Recorded sounds, before and after editing.

While found sounds can be great for impacts and atmospheric beds to sit behind music, it is a mix of found sounds, synthesized tones, and audio manipulation that will produce the best results. We'll investigate these approaches throughout this chapter.

Sounds such as risers and drops are best created via synthesis. Some plug-in instruments are dedicated to creating these forms of effect, the most notable of which is Air – The Riser synthesizer.

FIGURE 24.1
Air – The Riser

While you can use these effect specific synthesizers, it is not necessary, and it's possible to create a variety of effects within any capable synthesizer and some forethought. We'll examine how to create some of the more popular sound effects here, starting with the noise riser.

THE NOISE RISER AND FALL

The noise riser is a classic effect used in almost *every* electronic dance music track; it consists of a filtered noise that increases in intensity and is usually accompanied by a series of other effects such as a pitched riser, reverse crash, and sometimes a snare roll.

These are perhaps the most straightforward FX to create. Begin by selecting a noise waveform, white noise is the preferred choice, but you can experiment with pink, red, or brown. The amplitude envelopes attack and decay should both be immediate, with a full sustain and a 50% release. Set the filter to a 12 dB band-pass and adjust the resonance and filter cut-off values, so they're about a quarter way open. Finally, using a modulation envelope to modulate the filter cut-off positively, use the longest attack setting with no decay, no sustain, and a release about half way. This envelope should fully modulate the cut-off of the filter.

When you depress a note on the synthesizer, the cut-off of the filter will slowly open, allowing more of the bandpass noise to pass through. This produces the rising sound you can hear.

By shortening the attack time, you can make the riser reach its apex faster. The only effect this typically requires is some reverberation with a long tail to add aura and dimension to the riser. Here, it's common practice to automate the reverberation wet amount so as the sound grows, the reverberation becomes more evident. When the riser stops, the reverberation continues, but the wet amount is automated to reduce so that the reverberation washes over the mix and slowly dies away. You may also want to EQ the reverberation so that it doesn't take up too much energy in the mix.

To create the opposite of this effect so that the riser falls, you just have to open the filter to 3/4 way and invert the modulation envelope. By setting this envelope to negative modulation, the filter will begin at 3/4 open and then sweep downwards, closing at the rate set by the modulation envelopes attack rate.

THE PITCHED RISER AND FALL

The pitched riser is self-explanatory; rather than filtered noise, it consists of a single note pitch that rises and creates a build. To create this sound, we must begin with the sound itself. You can use any waveform for this, but the preferred options are a saw or pulse wave. For this example, I'll use a saw wave with some unison and detuning to thicken the sound and create a more exciting tone. Because this is a rising pitch, the oscillator should be pitched down to –2 or –3 octaves.

The amplitude envelope requires an immediate attack, decay, and release with a full sustain. There is no requirement for a filter (yet), and you only need to modulate the pitch of the oscillator with a modulation envelope. Set this envelope to a slow attack with an immediate decay, sustain and release value. Finally, set the modulation amount to around 3/4.

When you play a note, the modulation envelope will increase the pitch of the oscillator from its primary pitch (–2 or –3 octaves) and at a speed determined by the modulation envelopes attack parameter. By adjusting the modulation send amount and the attack speed, you can create a variety of pitch based effects. To generate a pitch fall effect, merely reverse the modulation envelope to a negative amount, or you can reduce the attack to zero and increase the modulation envelope's decay. By increasing the decay, the pitch will fall rather than rise. Both these effects will also benefit from reverberation (as discussed in the noise section) but depending on the record, they may also benefit from some delay too.

While creating pitch risers, it's worth experimenting further with the synthesizer. For example, you could add some LFO modulation to the pitch riser. To do this, use a sine wave as the LFO waveform to modulate the pitch and send it to the pitch by a small amount. If LFO is set to a fast rate, the pitch will warble as it increases (or falls) adding interest to the motion. If possible, you could also use a modulation envelope to modify the modulation send amount of the LFO. Using a long envelope, the LFO would gradually increase as the pitch rises.

IMPACT EFFECT

An impact effect can be created with little more than a modulated and distorted noise waveform. To build the basis for this effect, use a white noise waveform passed through a 12 dB low-pass filter. Set the cut-off at 1/4 way open with a median resonance value, and if available, set the filters drive to the maximum.

The amplitude envelope should have a fast attack and sustain with a long decay and release. Using a modulation envelope to affect the filter's cut-off, set this the same as the amplifier envelope, using a fast attack and sustain with a long delay and release. Send the envelope to modulate the filter by a small amount, and then add some clip or overdrive distortion with a small amount of reverberation. This will produce the primary impact effect, but it can be further augmented with LFO. For example, using a saw or triangle waveform, you could modulate the filter cut-off at a fast rate, so both the modulation envelope and LFO act on the filter, creating a more exciting impact effect.

RISING FILTER

Another favorite effect is the rising speed filter, whereby the rise speed increases as the filter opens further. You can use a saw, pulse, or triangle wave for this effect, but a saw produces the most interesting results.

Both the amplifier envelope and filter envelopes use a fast decay and release but with a long attack and high sustain. Use a triangle or sine wave LFO set to a mild positive depth and very slow rate (about 1 Hz) to modulate the cut-off of the filter. Finally, use the envelope of the filter also to modulate the speed of the LFO so that as the filter opens, the LFO also speeds up. If the synthesizer doesn't allow you to use multiple destinations, you can increase the speed of the LFO via the workstation's automation parameters.

FALLING FILTER

This is the opposite of the previously described effect so that rather than the filter rising and simultaneously speeding up, it falls while simultaneously slowing down. As before, both the amplifier and filter envelopes are set to a fast decay and release with a long attack, but this time no sustain is used.

A triangle or sine wave LFO is set to a mild positive depth and fast rate to modulate the cut-off of the filter. Finally, the envelope of the filter also modulates the speed of the LFO so that as the filter closes the LFO also slows down.

ARRANGEMENT SWEEPS

Arrangement sweeps are best created with saw oscillators as their high harmonic content gives the filter plenty to work with, but triangle and pulse can also be used depending on the result the producer requires. Two oscillators are necessary for this effect, both set to the same oscillator waveform and detuned from one another as far as possible without becoming individual timbres.

The amplifier envelope should be configured to a fast attack, decay, and release with the sustain set just below maximum. A slow saw or triangle wave LFO is then used to modulate the pitch of one of the oscillators. A bandpass will produce the best results if set to a medium cut-off but a very high resonance and a saw, sine, or triangle LFO is then used to modulate the cut-off of the filter to produce the atypical sweeping effect.

GHOSTLY NOISES

Ghostly noises can be designed in a synthesizer with two triangle waveform oscillators. Using a 12 dB low-pass filter, set the cut-off quite low but employ a high resonance and set the filter to track the pitch of the keyboard (filter key follow). The amplifier envelope should be set to a fast attack with a long decay, high sustain, and medium release and the envelope of the filter should be a quick attack, long decay, but with no sustain or release. Finally, an LFO sine wave is used to modulate the pitch of the oscillators slowly. Play chords in the bass register to produce the timbres.

The above are all quick examples to get you experimenting with a synthesizer. As you should realize by now, the creation of sound effects lies with modulation envelopes or the LFO, or sometimes both. For instance, modulating a sine wave's pitch and filter cut-off with an LFO set to a sample and hold or noise waveform will produce strange burbling noises while a sine wave LFO modulating the pitch can produce siren-type effects. There are no limits to this application and experimentation is key.

To generalize the application, however, the LFO waveform will contribute a great deal to the sound effect you receive. Often, triangle waves are suited for bubbly, almost liquid sounds, while saws are best for zipping style noises. Square waveforms are useful for short percussive snaps such as gunshots, and random waves are helpful for creating burbling, shifting textures. All of these used at different modulation depths and rates, modulating different oscillators, and parameters will create wildly differing effects.

Of course, not all sound effects are the result of synthesis, and many effects are created by manipulating audio. What follows are a few examples of how you can modify audio to create some classic effects. As always, this should not be considered the definitive list but is intended to get you started.

REVERSE CRASH

The reverse crash is one of the simplest yet most effective techniques for transition or development of a record. This is easily accomplished by placing an audio sample of a crash cymbal into the DAW and reversing it in the sample editor. While this alone can work, it often benefits from a variety of effects.

Often it is worth applying some reverberation to the crash before it's reversed. If the pre-delay of the reverberation is kept short but uses a long tail, when the audio is reversed, it creates a more extended, interesting riser to a cymbal. If the cymbal appears too long after applying reverberation and modifying, you can always use a DAW's fade in tool to create a shorter transition.

It's also advisable to apply some delay to due to the transient nature at the end of the reversed crash. With 1/16th, 1/8th, or 1/4 note delay, the cymbal can ring out after the transition. If this is mixed with a growing reverberation tail that rings out after the transition, it can work towards an exciting build, especially if combined with a white noise riser.

KICK DROP

The kick drop is a deep kick that resonates across the mix and is often used on a breakdown when the music falls right back to its essential elements before the drop occurs. To build this effect, you need a kick drum sample. Apply a reverberation to the kick as an insert effect with a short pre-delay, a 70% mix of wet to dry signal and a six second or more tail. Follow this with an EQ to remove all of the higher frequencies and some of the very low to produce a kick with a tail. You can then bounce this down to audio, and when reimported (with the effects applied), you can use pitch shifting software to lower the pitch of the tone to create a booming effect. If the tail is too long, you can use the DAW's fadeout tools to reduce the size but if it's too short, you can always time-stretch the audio to create a more interesting timbre.

PITCH DOUBLING

Pitch doubling provides an easy way to make a sound more interesting. It consists of placing a pitch shifting plug-in onto an auxiliary bus and sending the sound effect to that bus. By setting the pitch shifter to drop (or raise) the pitch of the effect by a few semitones and mixing it in with the original effect, it produces harmonically complex sounds. These can be treated with delays, reverberation, and distortion to add further interest.

Of course, these are just suggestions of how you can create some widespread effects, but experimentation will always yield better results. For example, excessively time-stretching or time-compressing audio can produce some beautiful sounds, while automating effects and processing as the sound effect occurs can add further interest. There is also a variety of plug-ins available that are designed to contort audio in weird and wonderful ways. These, combined with field recording, synthesis, audio manipulation, and the mass availability of sample content dedicated to FX, will provide more than enough sound effects for your music.

CHAPTER 25

Vocals: recording and editing

I think I may have too many microphones.

No audio engineer, ever

If we examine the most popular and acclaimed electronic dance music tracks released over the past 20 years, almost all of them have featured vocals. This is no surprise; a great vocal is the most expressive and mesmerizing instrument we can employ in our music and they will always attract a listener's attention. So while they are not always necessary for the production of electronic dance music, I can guarantee that, as you grow as an artist, at some point in your career, you'll want to feature vocals in your music.

While there is an abundance of vocal sample content available, the majority feature such poorly written vocals lines that they're unusable. And of the limited number that are useful, they are so overused that they're cliché. Consequently, just as learning music theory will help you in creating music, learning how to record, edit, and mix vocals can be equally important.

Recording professional vocals is represented as expensive and overly convoluted, requiring specific vocal booths and thousands more lavished on microphones and esoteric pre-amplifiers. This isn't always the case, and it is possible to record studio-quality vocals on a modest budget. If you know what you're doing.

THE TECHNOLOGY

The first link in the vocal chain lies with the microphone. A microphone converts analog sound into an electrical signal through the use of a diaphragm. This is a lightweight ring of thin plastic (commonly Mylar or gold-spluttered Mylar) that vibrates with air pressure. This vibration is converted into an electrical signal and delivered to the recording device.

There are various types of microphone available, but ultimately, for recording vocals, there are two main options: the dynamic and the capacitor. Both employ

FIGURE 25.0
A dynamic microphone

a diaphragm but use a different design, and so they provide a different tonality from one another. The dynamic microphone is perhaps the more recognizable microphone.

On a dynamic microphone, the diaphragm sits at the head with a lightweight wire coil attached to its rear that is suspended over a tubular magnet. When sound strikes the diaphragm, the coil is forced into motion over the tubular magnet. As the wire moves over a magnetic field, it produces a small electrical current.

Because this diaphragm design has a metal coil permanently attached to it, it's a relatively heavy assembly, and, therefore, the diaphragm is not particularly sensitive to changes in air pressure. This means that it will readily capture the lower and midrange frequencies, but less of the higher. Consequently, these microphones often exhibit a nasal-like quality. This is not to suggest that dynamic mics are inappropriate for vocals. Some talent (your vocalist) will sound her best through a dynamic microphone, and some great studio albums have been captured with them. Models such as the Shure SM7B continue to be employed in professional studios worldwide due to their specific vocal character.

Microphones such as the Shure SM57 and AKG D5 are also widely recognized as some of the best choices for live performances. Although part of their charm is that they're sturdy, so will survive heavy-handed roadies.

CAPACITOR MICROPHONES

The preferred choice of microphone for studio recording is the capacitor or electrostatic microphone. Sometimes called a *condenser* microphone, these employ a different diaphragm arrangement and are more sensitive to changes in air pressure.

Capacitor microphones employ a diaphragm that is suspended and separated from a metal backplate by a few microns. When an electrical charge is applied to the backplate, an electrical *capacitance* field is generated between the diaphragm and backplate. As air pressure strike the diaphragm, it alters the distance between the diaphragm and backplate, changing the capacitance. This produces changes in electrical current that reproduce the audio signal at the end of the chain.

This capacitance design permits manufacturers to employ much lighter diaphragm assemblies than the dynamic variants. Consequently, these microphones are far more sensitive to changes in air pressure and capture a fuller frequency with more accuracy.

Unlike a dynamic microphone, because an electrical charge is required to provide the capacitance between the diaphragm and backplate, capacitor microphones need either batteries or, more commonly, an external +48v power supply.

In many situations, this power is provided to the microphone via a pre-amplifier and delivered via the XLR cable connecting the microphone to the pre-amplifier. This power supply is known as *phantom power*, because the power is provided to the microphone on the same cable that is used to carry the audio signal.

FIGURE 25.1
A capacitor microphone

Alongside delivering the voltage, a pre-amplifier will also increase the microphone's signal voltage. This is a requirement for many capacitor microphones, but some dynamics also require large amounts of amplification, too. By employing a quality pre-amplifier, the low signal level generated by microphones can be increased by up to 60 dB.

Many USB, Firewire, and Thunderbolt audio interfaces will feature pre-amplifiers. The recording quality varies considerably, however. Many of the cheaper, consumer-based audio interfaces suffer from a lack of low end or exaggerate the highs resulting in muddy or harsh recordings. Others, such as the UAD Apollo feature high-quality pre-amps that can employ real-time software emulations of infamous amplifiers. Similarly, manufacturers are producing completely transparent microphones that use sophisticated software algorithms to model the sound of highly sought-after (i.e., expensive) microphones. With the vocals recorded, it is then possible to add the character of different microphones to suit both the talent and the music.

While this form of modeling can be attractive because it offers access to these rare and expensive microphones, it also relies on the manufacturer to keep up to date with the drivers. Something as simple as an operating software update can render your microphone useless until the drivers catch up. Or, worse still, a second version of the product can often lead to the abandonment of drivers for the older model. This happens more often than you would imagine, and it can turn expensive software-driven pre-amp/microphone emulations into a basic microphone, or worse still, a doorstop.

Alongside modelled microphones, there are a growing number of USB microphones on the market. These do not require an audio interface or a pre-amplifier because they draw power from the USB port and act as a recording interface, too. While these do reduce the need for most external hardware, the audio quality varies greatly. Many are suitable for YouTube videos, narration, and the odd vocal shout, but it's difficult to recommend them for recording vocals for music. Although a great idea will always trump the technology, the AD converters built into USB microphones are mostly lacking in quality. Many exhibit a harsh midrange or an exaggerated high frequency that makes it difficult to sit the vocals into a mix.

It's because of this that many professionals opt for a quality hardware pre-amp and microphone that operate independently of a driver-based interfaces and USB microphones.

There is a vast range of pre-amplifiers and microphones available, ranging in price from £100 to over £14,000. There are choices between tube, FET, and solid state microphones. Pre-amplifiers can be basic, offering little more than amplification and phantom power while others provide a complete recording channel. These feature the most common parameters you would require when recording vocals or instruments, including de-essers, EQ, compressors, and limiters.

While these channels are useful when recording, unless you know what you're doing, you should avoid processing during recording except for perhaps some light compression to prevent clipping. Once processing is applied to a signal through a pre-amp, it cannot be removed later. If too much EQ or limiting is used at the recording stage, the only solution is to re-record everything. A better approach is to employ a basic standalone pre-amplifier and apply processing *afterward* via plug-ins while the signal is in the DAW.

The choice of pre-amplifier and microphone is down to personal taste and what performs best with the talent currently in the studio. Different combinations of microphones and pre-amplifiers produce different sonic results with different talent.

For example, two different microphones connected to the 1073 or API 512c will sound different to one another regarding tonal quality. And, to make matters worse, some talent will suit specific microphones and pre-amp configurations more than others. The differences between all these aren't *night and day*, the differences are often subtle, but a well-matched microphone, talent, and pre-amp will produce truly exceptional results.

Some of the most lauded microphones are the Neumann U87, Neumann U47, AKG C414, Telefunken U47, and the Telefunken ELA M251. On the pre-amplifier side, there is the Neve 1073, the Great River MPV500, UA610 MKII, Tube-Tech MP-1A, API 512c, and the Focusrite ISA. A Neumann U87 paired with a Neve 1073, UA610 MKII and an LA2A compressor were used for the examples in this chapter.

Further tonal changes can result from the size of the diaphragm of the microphone. A larger diaphragm exhibits a more substantial mass and reacts more slowly to changes in air pressure compared to a smaller one. These smaller diaphragm microphones will respond more quickly to changes in air pressure, resulting in a sharper and more defined sound.

In theory, this would suggest that, to capture a perfect performance, we should employ a small diaphragm model, but, in practice, this is not the case. Large

FIGURE 25.2
The author's UA610 MKII

diaphragm microphones will feature a diaphragm diameter of approximately 25mm. If this is exposed to a wavelength of equal size (10 kHz has a wavelength of 25mm), there is a culmination of frequencies at this point. This results in more directional sensitivity at higher frequencies.

If this is compared to the reaction of a smaller diaphragm microphone that typically features a 12mm diameter, the build-up occurs much higher in the frequency range (20 kHz has a wavelength of 12.5mm), and this is beyond the threshold of human hearing. So, while a smaller diaphragm will capture a more accurate recording, the higher directional sensitivity of a more substantial diaphragm imparts a warmth to a signal that is considered to be more suited and flattering to the human voice.

The distance the talent is from the microphone can also influence the frequency response of the microphone. As the talent moves closer to the microphone, the bass frequencies become more pronounced. This is known as the *proximity effect* and can be used to make smaller diaphragm microphones sound like the more expensive larger diaphragm models. However, care has to be taken because if the talent gets too close to the microphone and he's loud, the bursts of air can damage the diaphragm.

Consideration must also be given to the polar pattern of the microphone since different patterns will affect the sensitivity based on its directional axis. A microphones *polar pattern* determines how sensitive the microphone is to signals arriving from different directions. The most common types are cardioid, omnidirectional, figure of eight, and hyper-cardioid.

Cardioid is usually the first choice for recording vocals. This is most sensitive to sounds arriving from the front with its sensitivity reducing as it moves towards the rear of the microphone. It's called a cardioid pattern because, when the pick-up pattern is presented on a graph, it looks like a heart.

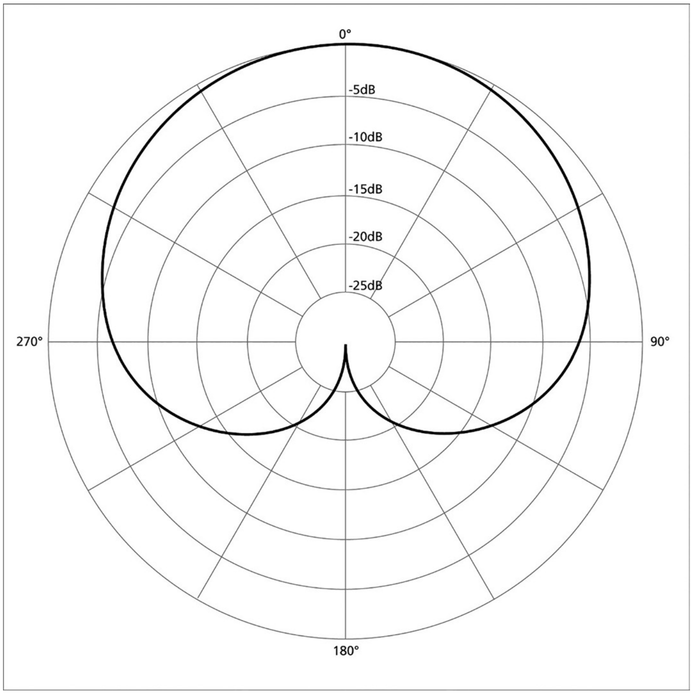

FIGURE 25.3
Cardioid pattern

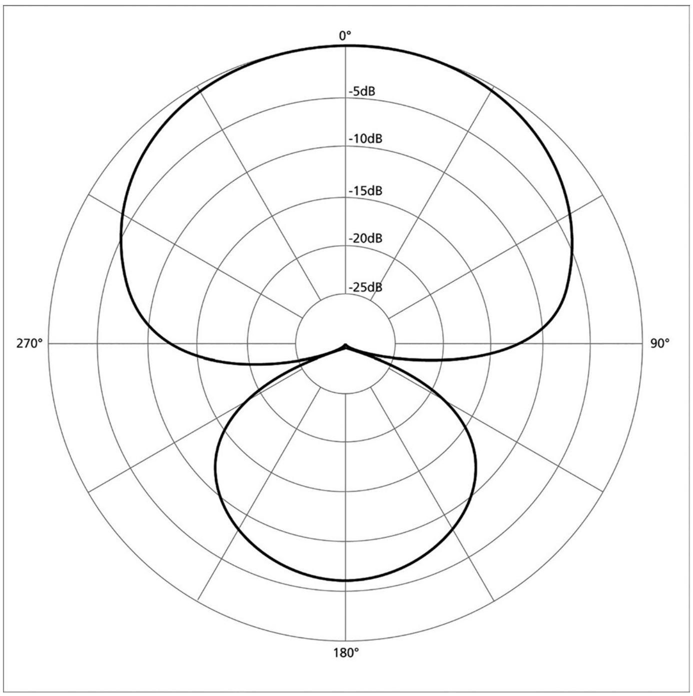

FIGURE 25.4
Hyper-cardioid pattern

While these microphones are more suited towards vocals, they are also the most prone to the proximity effect. So, when in use, the talent must try to maintain an equal distance from the microphone during the performance.

Hyper-cardioid microphones are similar to the cardioid models but exhibit a narrower pattern, so they do not pick up as much from the sides. But while the side sensitivity is reduced, they pick up sound from the rear.

These microphones are more focused and best used when recording instruments or drum kits because they are less susceptible to overspill from the sides. The signal received at the rear of the microphone often adds to the ambiance when recording instruments or drum kits.

The figure of eight microphone captures sound from both the front and rear but rejects sound from the sides. In many cases, these employ a *ribbon* diaphragm. This is a thin strip (imagine a thin piece of aluminum foil) secured at both ends. Some engineers prefer these to cardioid microphones since the ribbon assembly is lighter and therefore more sensitive. These microphones also do not suffer from a proximity effect. However, ribbon microphones roll off higher frequencies so vocals can appear dull or muted through them. While this makes them less suitable for singing, they are very suited for spoken word.

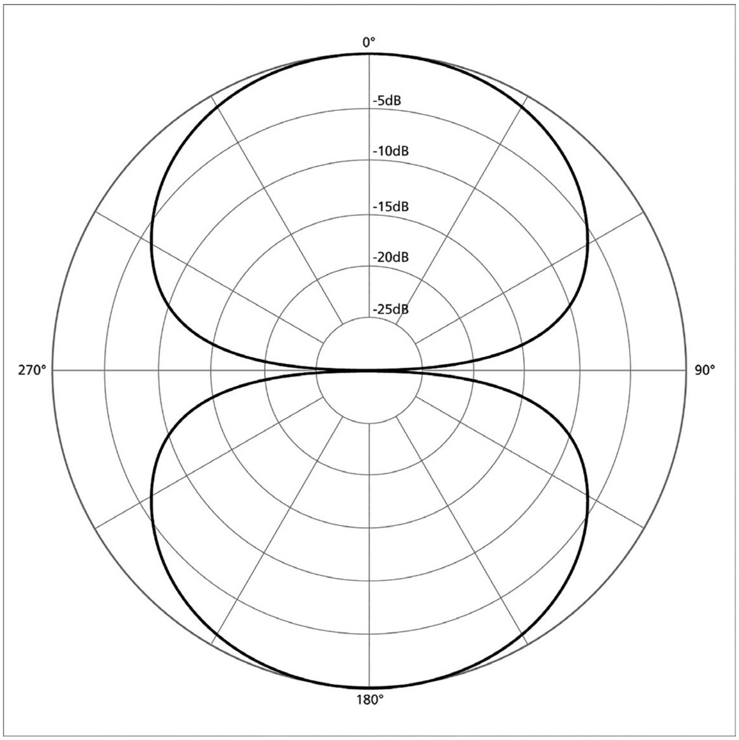

FIGURE 25.5
The figure of eight

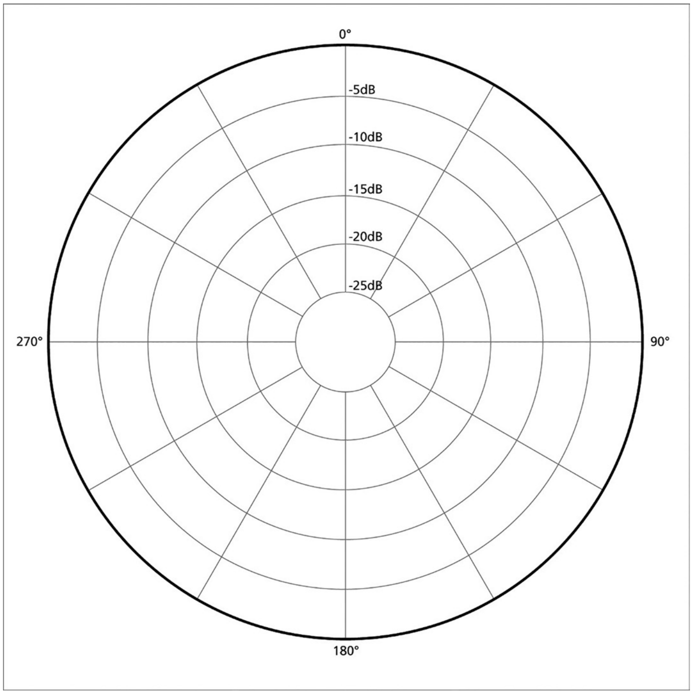

FIGURE 25.6
Omni-directional pattern

FIGURE 25.7
The author's Neumann U87

The omni-directional microphone has an equal sensitivity through its entire axis and will pick up sound from any direction. Sports interviewers employ this microphone because it doesn't require any prior knowledge of where to speak. It is sometimes used to record small string sections provided the room has a pleasant ambiance.

For electronic dance music, the preferred choice is a capacitor with a cardioid pattern and a large diaphragm. This is accepted as providing the best overall response that's characteristic of most genres, but any large diaphragm will suffice provided that the frequency response isn't heavily manipulated. Some models employ peaks in the midrange for vocal clarity, but in my experience, it can often accentuate sibilance with female talent. This is the result of overemphasized 'SH' sounds that create unwanted hissing in the vocal track.

Although microphone choice is as individual as your creativity and the vocalist, the most famous microphones are the Neumann U87, Shure SM7B, Rode NTK, and the Neumann TLM 103.

RECORDING THEORY

Beyond the equipment, an excellent recording begins with the room. You should aim to record vocals with as little of the room's natural ambiance as possible. If a significant amount of ambiance is captured during recording, it cannot be removed later. Any further effects will then not only apply to the vocals but also to the recorded ambiance.

There are various methods to employ for capturing dry vocals. The most widely publicized include hanging drapes or curtains behind the talent through to having her perform in a large cupboard or portable sound booth. While both these approaches do work, neither offers the perfect solution for a great vocal take.

It can be challenging to hang heavy drapes behind your talent unless you plan to record near a window with the curtains drawn. This can be fraught with problems because they will not absorb *all* the acoustic energy. Some sound waves will pass through the curtains and strike a flat glass object. This results in sonic reflections that can be more evident than if the talent had nothing but a bare wall behind him.

Similarly, while a portable vocal booth may provide the perfect solution, they are expensive, require a lot of room, and are only mobile if you own a large van to transport them. More importantly, your talent is a human being and packing her into a small enclosed area can be stressful. So while you may capture an excellent recording in a tiny booth, there is no guarantee it will be an equally outstanding performance.

A simpler solution is to employ a reflection filter and some carefully positioned gobos. A reflection filter is a semi-circular portable acoustic absorber that is mounted on a stand and sits *behind* the microphone. Although placing an absorber behind a microphone may appear odd, acoustic energy is captured by the absorber and is unable to escape into the room to create acoustic reflections. Plus, as many capacitor microphones will pick up sound from their sides, fewer of the reflections of the room are captured.

While the filter mentioned above will aid in controlling acoustic reflections, they work best when combined with acoustic gobos. These are small acoustic isolation panels that can be purchased or you can build them yourself.

Three panels are placed behind the talent in a widened U shape to prevent reflections from the side and behind being captured by the microphone. Figure 25.9 shows a typical gobo arrangement.

Although many professional gobos are 6 to 7ft tall, reaching from the floor upwards behind the talents head, you can construct much smaller panels that are easier to transport. A simple wooden frame measuring 3ft tall x 2ft wide

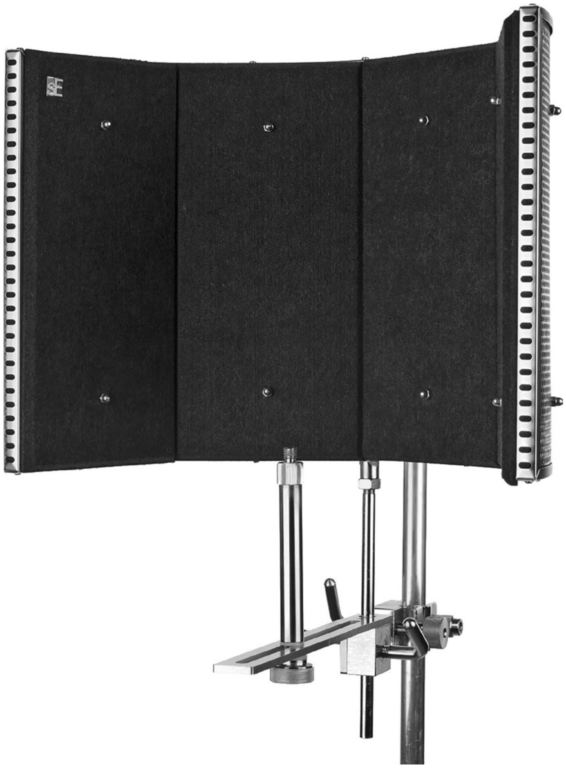

FIGURE 25.8
The SE Reflexion (reflection) filter

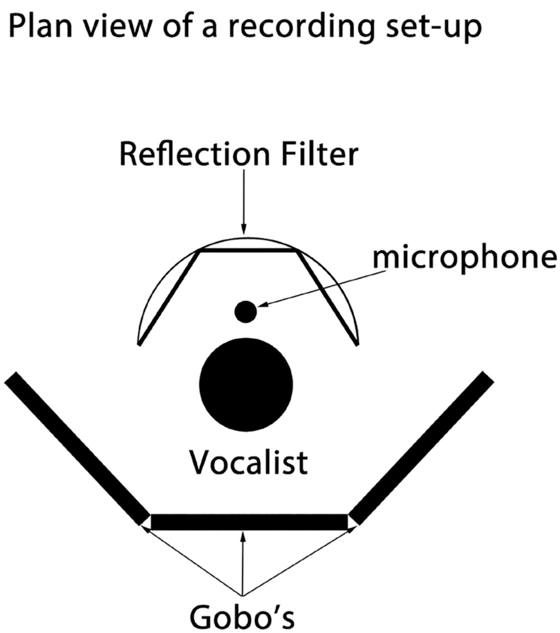

FIGURE 25.9
A gobo arrangement for recording

packed with Rockwool RW3 insulation and covered by a breathable material can act as an excellent acoustic absorber. These are mounted on speaker or microphone stands behind the talent's head to prevent any rear or side reflections from being captured by the microphone.

There are, of course, alternatives to using this arrangement such as isolation units. These are portable sound isolation 'boxes,' similar in some respects to the SE Reflexion filter but they completely surround the vocalist's head. These are placed on a stand, and a microphone is positioned inside the box along with the vocalist's head. While these units isolate sound very well, I've had limited success with them.

A great vocal take is about capturing a performance, a certain moment in time when everything clicks together. Every great vocalist I've worked with performs with the music, they feel the rhythm and groove, and so it's not unusual for them to dance, nod their head (while waiting to perform the next section), and many wave their hands around while singing. None of these is possible if your vocalist has his head stuck in a small box so capturing a *performance* is difficult. Plus, there is the added psychological stress of your vocalist feeling locked in a small box. So while these isolation units are great for narration, they're not always suitable for a vocal performance. Your mileage may vary however.

Whatever isolation methods you choose, the microphone should be mounted on a fixed stand and in a reliable shock mount. These are mechanical elasticated units that are designed to isolate the microphone from any vibrations working their way up the stand.

Almost all capacitor microphones are supplied with a shock mount (aka, a cat's cradle), but the quality varies greatly. Some can perform particularly well while others can intensify the problems of vibration from the stand and cable.

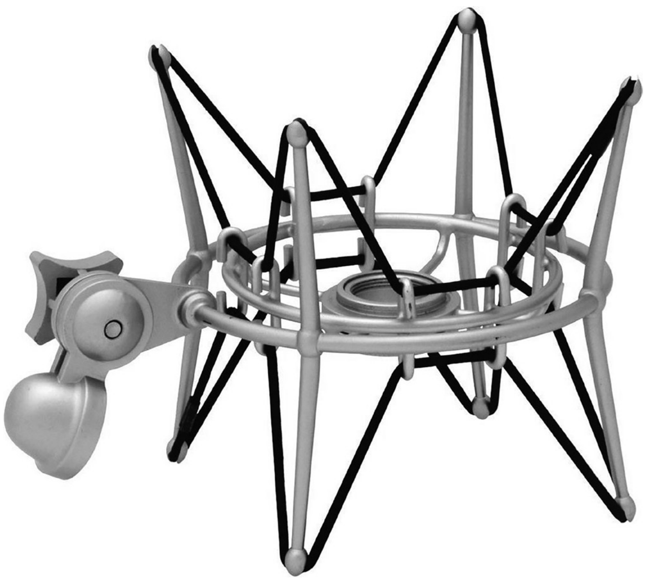

FIGURE 25.10
A shock mount

The microphone should be positioned so that the diaphragm is level with the performer's nose while she's standing with her back straight. This will ensure a good posture while also avoiding plosive popping or breath noises.

Plosive popping is perhaps the most common problem when recording vocals and is a result of the high-velocity bursts of air that come about from words containing the letters P, B, or T. These produce short, sharp bursts of air that force the microphones diaphragm to its extremes of movement and results in a popping sound.

Plosives are challenging to remove afterward requiring either judicious EQ or de-popper software, so they're best controlled during the recording session. This can be done with a pop shield, a circular frame approximately six to eight inches in diameter and covered with a fine nylon mesh. These are available as freestanding units or with a short gooseneck for attaching to the microphone stand. By placing these approximately two inches away from the microphone, any sudden blasts of air are diffused by the pop shield preventing them from reaching the diaphragm.

FIGURE 25.11
The Rycote pop shield (regarded by many as one of the best pop shields)

Although pop shields are available commercially, some choose to make their own from metal coat hangers or embroiders' hoops and nylon stockings. Alternatively, some attach a pencil to the center of the microphone grill so that the plosives strike the pencil and not the diaphragm (this technique can sometimes also work for sibilance).

While these do work in a pinch, I don't recommend it as a technique. Good pop shields are not expensive, and perform better than homemade devices. And a pencil held on a microphone with a rubber band doesn't give off a very professional vibe to your talent. The microphone can be positioned either the correct way up or upside down; both have advantages and disadvantages. If the proper way up, it's held solidly in position, and, if it's a tube unit, the heat can warm the diaphragm that can aid in producing a smoother tone. The downside is that your talent may knock or kick the stand resulting in noise in the recording. Hanging a microphone upside down avoids this problem but it's less sturdy, and if your talent flails her arms wildly while performing, she may knock the microphone.

With the positioning set, the ideal distance between the performer's mouth and the microphone is approximately 12 to 18″. At this distance, the voice tends to sound natural plus any small movements on the talent behalf will not result in any severe changes in the recording level.

If you a recording an amateur or a live performer, he will often edge more closely to the microphone, and this will increase the proximity effect. Some engineers suggest that positioning the microphone above the talent's head and angling it downwards prevents this from occurring. Personal experience has taught me that if your talent raises her chin, it affects her tonality. A better alternative is to increase the distance between pop shield and microphone. If the performer is then informed to brush her lips against the pop shield during the performance, she will maintain an equal distance throughout.

The next consideration is how your music is delivered to the vocal booth. This known as a *cue mix* and is distributed via headphones with good isolation to prevent the music being picked up by the microphone. This means that *open* headphones should be avoided due to their leakage, but equally, you should avoid closed, too. This is because the closed design clamps around your ears that produce a rise in bass frequencies and this makes it difficult to intonate. So, it is best to use semi-closed designs. Although these will likely leak *some* of the mix onto the microphone, you can easily disguise it when mixing.

Some talent may ask to hear his voice in the headphones alongside with the mix. This is called the *foldback* mix, and he may need it to intonate correctly. I usually ask him just to remove one earphone cup but if you choose to send a foldback mix, treat the vocals to some reverb. This will make the vocalist feel more confident about his voice (reverb can make anything sound better than it is). This only needs to be lightly applied and, typically, a small room with a long pre-delay and short tail of 50ms is enough.

RECORDING VOCALS

A compressor should be placed between the microphone and the recording device to prevent any excessive dynamics from clipping the recording. The ideal compressor for many producers is the LA2A. This unit is probably so well used for vocals because it is easy to set up (it only has two parameters) and you don't wind up messing around with multiple parameters for 20 minutes. It should be set so that the compressor allows the majority of the signal to pass through unmolested and only acts on very loud signals.

A good starting point for recording compression is to set a threshold of –8 dB with a 3:1 ratio, a fast attack, and moderately quick release. Once the talent has started to practice, reduce the threshold so that the reduction meters are only active on the most active part of the performance. Varying dynamics in vocals are essential in maintaining the feel and focus of the performer and heavy compression applied during recording cannot be removed afterward.

Usually, talent will require some warm-up time, and this can range from 30 minutes to over an hour. Most professionals will have their routine to warm up, and many prefer to do this in private, so it may be necessary to arrange some space. If the talent is inexperienced, he may insist he does not require a warm-up, so you'll need to exhibit charm and ask him to perform some favorite songs that he enjoys while you *adjust the recording levels*. During this time, listen for any signs of poor intonation or vocal straining. If you hear it, ask him to take a five-minute break. If he needs a drink, fruit juices are better than water, tea, or coffee because these dry your throat but I would also avoid citrus beverages. 7-Up (the soft drink) seems to help a lot of the talent I've worked with but avoid letting them drink too much because the sugar settles at the back of the throat.

Before recording begins, it is *essential* that the talent has practiced and knows the lines off by heart without having to read them off a sheet of paper. A great vocal recording comes from a passionate performance and reading direct from a sheet is going to impact the power and feeling.

Once warmed up, the voice should be at peak performance, and you should attempt to capture the complete performance from beginning to end. The talent should be persuaded to avoid stopping for any mistakes and just to keep going, because you can record more takes later. There is no recorded music today that is completed in only one pass, and even professional artists will take 15 to 20 passes. The first performance is commonly the most energetic and influential so you should aim to capture as much of it as possible with any further takes being little more than edits.

During the recording, *do not wander off mentally*! Do your job and listen intently to the performance, and try to identify problems. Typical problems are poor intonation, the inability to hold high notes, uneasiness and tenseness, sibilance, or too much bass presence.

If the microphone is positioned correctly, there shouldn't be any bass problems. If there is, wait until the performer has finished and then ask she move further away from the microphone.

Severe intonation problems, the inability to hold high notes, uneasiness, or tenseness are challenging to resolve because they reside in the psychological state of your talent. Assuming that he can sing in tune, the most common reason for poor intonation is from the performer's monitoring environment.

If there is too much bass in the music, it will make it difficult to perceive pitch results in a loss of intonation. Watching the talent involuntarily moving further away from the microphone and performing more loudly is a universal sign of this, because it reduces the bass presence of the voice in the foldback mix. If you see this happening, a wide cut of a few dB at 100 to 300 Hz or increasing the volume of a higher pitched instrument may help.

If the problem is an inability to hold high notes or a tense or uneasy voice, then she's not breathing deeply enough because she's nervous. This is something that can only be resolved through tact, plenty of encouragement, and excellent communication skills.

Give them plenty of positive support and compliments after they've completed a take. They should always receive positive reinforcement, 'wrong,' or 'don't' should not be a part of your vocabulary. If they perform something wrong, tell them 'it was great but how about next time we try …'

Once you have successfully recorded the first tonally correct, emotionally charged take, there will undoubtedly be a few mistakes. It isn't unusual to record as many as 20 or 30 takes of different parts to later 'comp' (engineer speak for *compiling*, not compressing) various takes together to produce the final vocal track.

At this point, it is unwise to ask him to perform the whole song again, because it is exhausting; instead have him complete the verses (and perhaps chorus sections) singly, with breaks in between.

DMM resource: To demonstrate the difference the environment can make on a recording and the mix, I recorded an artist (Talson Moon) in a typical environment faced by a home producer; a small home studio. I recorded the vocals through a Neumann U87 microphone into a 500 series Neve 1073. The signal from the pre-amplifier was delivered to my MacBook via the Universal Audio Arrow.

The first recording is taken in the bedroom, and then the result of that recording placed into a mix. The second recording is taken with a gobo and Reflexion set-up, followed by that recording placed into a mix. Note how the second vocal appears to seat in the mix better than the first.

EDITING VOCALS

With all of the vocal parts recorded, you must be willing to spend as much time as necessary to ensure that the vocal track is perfect. This process begins by compiling (comping) all the best vocal takes together to produce one complete take. This involves listening to each different take, choosing the best vocal parts from each, and then pasting them together. This can include anything from comping individual lines through to individual words and even syllables. It's not unusual to spend a day or two comping a vocal track.

During the comping process, you may come across further errors in the recording such as the odd plosive that crept through or small amounts of sibilance. The latter can be removed by placing a thin EQ cut somewhere between 5 and 7 kHz. Alternatively, you can employ a de-esser. These are frequency-dependent compressors that operate at frequencies between 2 and 16 kHz.

If the problem is the odd plosive, these can often be fixed with small amounts of corrective, surgical EQ, but if this approach fails there may be no choice but to recall the performer and record that passage again.

I like to comp the vocals three times to produce three different takes. This way, I can use one as the original vocals for the track, and the two others for doubling during more dynamic parts of the music.

Once the vocals have been compiled into verse and chorus, the next step is to cure any intonation problems with pitch correction software such as Auto-Tune or Melodyne. These unfairly receive a terrible reputation for correcting vocalists who can't hold a note but it's media propaganda. It's somewhat ironic, but the closer your talent is to the correct pitch, the more natural the pitch correction sounds. The software cannot correct a poor performance without sounding like Stephen Hawking trapped in a wind tunnel.

No matter how great the performer, a lack of pitch correction is one the main reason the vocal does not 'sit' with the music. Even vocals that are 20 cents away from the key can prevent them it from sitting correctly, and this often results in heavy-handed EQ and effects to cure the problem.

After intonation, it's common to compress the vocals to restrict their dynamic range. Indeed, the most popular processing used to obtain the upfront sound is compression. Many producers will employ more than one compressor, and the chain will commonly consist of two or three, depending on the vocals. By applying more than one compressor, the signal can be gradually restricted in dynamics resulting in less evident compression.

The first compressor in the chain is solid state to smooth out the overall level. Use a ratio of 4:1, a fast attack and release, and threshold and adjust so that the gain reduction meter reads approximately 4 dB. The second compressor is commonly valve to impart some character onto the vocals. This is less conservatively, with a ratio of 7:1 with a fast attack and release, along with a threshold so that the gain reduction meter reads around –8 dB.

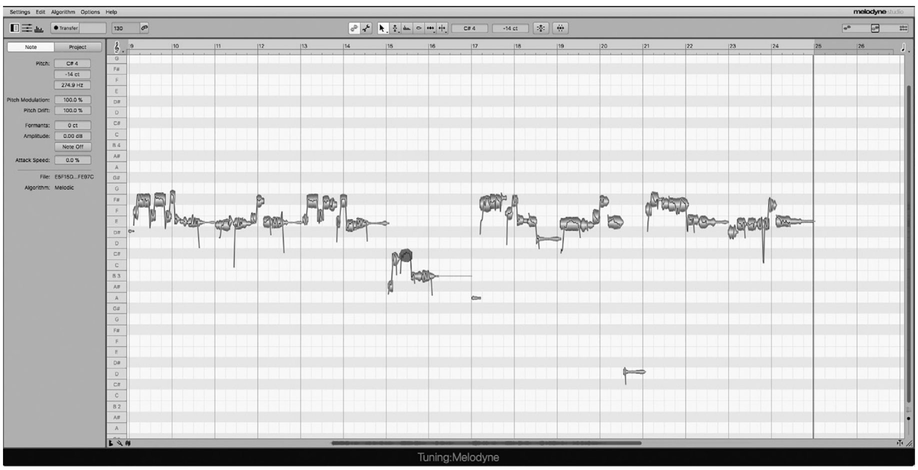

FIGURE 25.12
Correcting intonation with Melodyne

Of course, these are general settings and will need adjusting depending on the vocal character. Note that applying compression may reduce the higher frequency as it squashes down on the transients, but if the top end 'air' seems lacking, it can be restored with small amounts of sonic enhancement or EQ.

Following compression, the next processor is EQ. This is to remove any low-end rumble and fine-tune the midrange so that the vocal sits in the mix. It isn't possible to offer guidance on EQ but a low shelf to remove the lower frequencies with small boosts between 2 and 5 kHz should put you in the ballpark.

After EQ, the signal is commonly sent to a reverb unit to thicken and place the vocals into the mix. The vocals should be sent via a bus to reverb and not inserted so you can keep the majority of the vocal track dry. Typically, the reverb employed is a small room or plate with a short tail and long pre-delay, but like EQ, this depends on the mix.

The basic technique for reverb is to set the effect so that it's only noticeable when it's removed. You can do this by keeping the decay time short for a fast tempo and increasing this for slower speeds. A pre-delay of 40ms with a 70ms decay provides a good starting point, and by then increasing the pre-delay you can separate the effect from the dry vocal to enable the voice to stand out. Following this, it's standard to send the vocal to a couple of delay units. Here, one unit is set to 1/16ths and the second to 1/8th. By adjusting the feedback and send amounts, you can find a balance that works best for the music.

VOCAL EFFECTS

It would be amiss to end a chapter on vocals without first discussing some of the favorite effects that are employed within electronic dance music. In fact, it's not uncommon nowadays to treat a vocal as any other instrument in the mix and EQ or process them through effects chains to create interesting variations.

Naturally, there are no definitive approaches to processing vocals, and it's the producer who pushes the envelope and experiments who reaps the most significant rewards. Nonetheless, some vocal effects are still in extensive use throughout the genres today and knowing how to achieve these may aid towards further experimentation, so let's look at a couple of the famous techniques.

Sample and pitch vocals

Major Lazer is responsible for the resurgence of the sample and pitch vocal melody lines in the chorus with 'Lean On.' Since its release in 2015, there have been countless songs released using the same technique. Although the application of this effect is simple, it is essential to use the right *style* of vocal sample. It should be reasonably short and highly transient for this to work. Consequently, if you want to experiment with this technique, it's probably best to practice using the same sample they did on the original song. The sample is taken from sample content Vengeance Samples Essential House 3 and the file is VEH3 Synths Roots C 238.wav.

You will need to import this file into sample playback software such as Simpler in Ableton, HALion in Cubase, or the EXS24 in Logic Pro X. Once introduced, the sample is stretched across the key range, so it runs an octave above and below the original sample. After this, it's a simple case of programming the lead melody. The MIDI file for this melody is available in the resources pack for this book (available from http://www.dancemusicproduction.com), and it's also shown in Figure 25.13.

It is essential to pay attention to the overlapping notes in the piano roll editor. The charm of this particular lead is that the sampler is set to mono operation (so that the sampler can play only one note at once) and glide is employed so that the notes slide up in pitch to one another.

This technique produces the bandwagon that everyone is jumping onto. Once you have the necessary sound, it's just a case of adding reverb, delay and some stereo spread to the signal.

Telephone vocals

A favorite effect is the telephone vocal, so named because it sounds as though the vocal line is being played over a long-distance phone line or through a small speaker transistor radio.

The effect is accomplished in one of two ways. The first employs a bandpass filter inserted into the vocal track and tuned into the vocal so that frequencies below 1 kHz and above 5 kHz are removed from the signal. If the producer doesn't have access to a bandpass filter or its functions are limited, the same effects can be accomplished by chaining a low-pass and high-pass filter together on the inserts. With the low-pass inserted first, this can be employed to remove all frequencies above 5 kHz. The output of this filter then feeds into a high-pass filter that is adjusted to exclude all the frequencies below 1 kHz. 2-pole (12 dB)

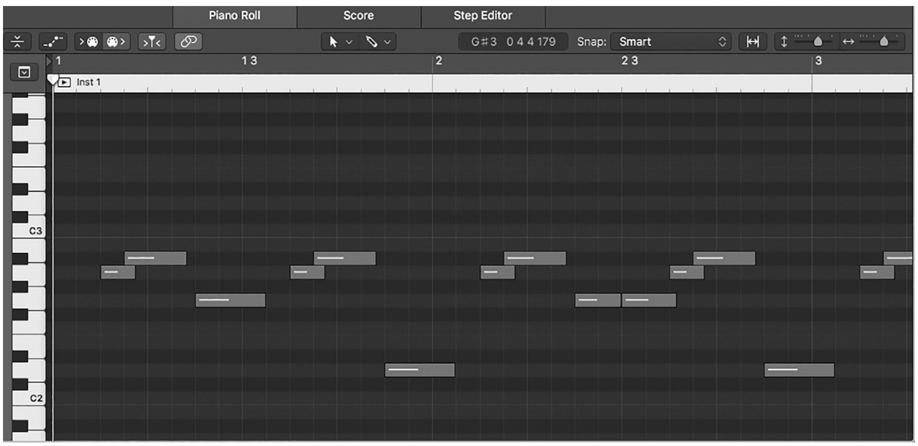

FIGURE 25.13
The lead theme from 'Lean On'

filter slopes will invariably produce better results than 4-pole, but this is open to experimentation.

An alternative approach to employing filters is to insert an EQ unit. Through the use of shelving filters, it is possible to remove frequencies either side of the vocals and then add further presence by boosting with a thin Q at 2 to 4 kHz. Once this effect is accomplished, it is not unusual to automate the filters or the EQ, slowly modulating back and forth by small amounts to add rhythmical modulation and interest to the vocal.

Intonated vocals

The moving pitched vocal effects made famous via Cher's 'Believe' and 'T-Pain' have become commonplace in music, so much so, that they should be applied cautiously rather than zealously, otherwise the results may appear particularly cliché.

The effect is typically accomplished by inserting a pitch auto-tuning plug-in such as Antares Auto-Tune on the vocal track. A scale is chosen for the vocals (most commonly C major is selected, even if the vocals are not in C major) and then the 'retune speed' is set as fast as possible. If there is a mode to select how 'relaxed,' the tuning plug-in is, this is set to relaxed, and the tuning is set to automatic.

When playback occurs and the tuning program detects the voice is not in tune with the current scale, it immediately corrects the intonation dragging the pitch of the vocals upwards and downwards. The effects can be made more noticeable by changing the scale even further away from the original key of the vocals.

Since this has become a very formulaic effect, it is often best employed very occasionally and only lightly on the occasional words so that it doesn't draw too much attention to itself.

The vocoder

The **vo**ice en**coder** is perhaps the most potent vocal effect since it offers a vast array of potential beyond its normal robotic voice. Vocoders are relatively simple in design and permit the producer to use a modulator – commonly the human voice – to modulate a secondary signal known as the carrier, which is usually a sustained synthesizer timbre.

A vocoder operates on the theory that the human voice can be subdivided into many distinct frequency bands. For example, plosive sounds such as 'P' or 'B' consist mostly of low frequencies, 'S' or 'T' sounds consist mostly of high frequencies and vowels consist primarily of midrange frequencies. When the modulator (vocal) signal enters the inputs of a vocoder, a spectrum analyzer measures the properties of the signal and employs some filters to divide the signal into any number of different frequency bands. Once separated, each frequency band is delivered to an envelope follower that produces a series of control voltages based on the frequency content and volume of the vocal part.

This same process is employed on the carrier signal, and these are tuned to the same frequency bands as the input of the modulator. However, rather than generate a series of control voltages, these are connected to a series of voltage-controlled amplifiers. This means that the modulating signal and its subsequent frequencies act on the carrier's voltage controlled amplifiers and therefore attenuate or amplify the carrier signal, in effect, superimposing the modulator onto the carrier (the timbre of the instrument).

What's more, because the vocoder only analyses the spectral content of the modulator and not its pitch, there is no requirement to 'sing' into a vocoder. Instead, the modulating signal will change pitch with the carrier signal. This means the performer could speak into a microphone connected into the vocoder and whatever notes are played on the carrier signal (the synthesizer), the vocals will move to that pitch. This makes vocoding a particularly powerful tool to employ, especially if you're unable to sing and it's an effect used by countless dance artists including Daft Punk.

It should be noted that the quality of the vocoder will have a considerable influence on the results. The more filters contained in the vocoders bank, the more accurately it will be able to analyze and divide the modulating signal, and if this happens to be a voice, it will be much more comprehensible. Typically, a vocoder should have an absolute minimum of six frequency bands, but 32 or more is preferable. However, the number of bands available isn't the only factor to take into account when employing it with vocals.

The intelligibility of natural speech centers between 2.5 kHz and 5 kHz. Higher or lower than this and we find it difficult to determine what's being said. This means that when using a vocoder, the carrier signal must be rich in harmonics around these frequencies, since if it's any higher or lower, then some frequencies of speech may be missed altogether.

To prevent this, employ a couple of shelving filters to remove all frequencies below 2 kHz and above 5 kHz before feeding them into the vocoder. Similarly, for best results, the carrier signals sustain portion should remain reasonably constant to help maintain some intelligibility.

For instance, if the sustain portion is subject to an LFO modulating the pitch or filter the frequency content will be subject to a cyclic change, which may push it in and out of the boundaries of speech resulting in some words being comprehensible while others become unintelligible.

More importantly, vocal tracks should be compressed before they enter the vocoder. If not, the vocals may change in amplitude, and this will create many different control voltages within the vocoder. This can result in the VCA levels that are imposed onto the carrier signal following the changes in amplitude producing an uneven effect that can sound distorted. Additionally, breath noises, rumble from the microphone stand, and any extraneous background noises can also trigger the carrier and therefore, alongside compression, a noise gate should be employed to remove any excessive noise.

Naturally, due to the effect, a vocoder must be employed as an insert effect. With the voice acting as a modulator, the producer then needs to program a suitable carrier wave. It is the tone of this carrier wave that will produce the overall effect and therefore this is the essential element to creating a great vocoding effect. For robotic voice effects similar to Daft Punk's 'Get Lucky,' two sawtooth waves detuned from each other by varying amounts with a short attack, decay, and release but a very long sustain provide a close approximation. If the vocals appear a little too bright, sharp, thin, or 'edgy,' replacing one of the oscillators with a square or sine wave, detuned by an octave, will add more bottom end weight.

Unsurprisingly, much of the experimentation with a vocoder derives from modulating the carrier wave in one way or another, and the most straightforward place to start is by adjusting the pitch in time with the vocals. This is accomplished by programming a series of MIDI notes to play out to the carrier synth, in effect creating a vocal melody.

Similarly, an arpeggio sequence used as a carrier wave can create a strange gated, pitch-shifting effect while an LFO modulating the pitch can create an unusual cyclic pitch-shifted vocal effect. Filter cut-off and resonance can also impart an exciting effect on vocals, and in many sequencers this can be automated so that it slowly opens during the verses, creating a build-up to a chorus section. What's more, the carrier does not necessarily have to be built with saw waves. A sine wave played around C3 or C4 can be used to recreate a more tonally original vocal melody that will have some peculiarity surrounding it.

Vocoders do not have to be used on vocals, and excellent results can be achieved by imposing one instrument onto another. For instance, employing a drum loop as a modulator and a pad as the carrier, the pad will create a gating effect between the kicks of the loop. Alternatively using the pad as a modulator and the drums as the carrier wave the drum loops will turn into a loop created by a pad.

CHAPTER 26

Formal structure in dance music

It sounds a lot more like it does now than it did ten minutes ago.

Sony A&R representative

Beyond the sound design, processing, effects, and composition, another hurdle we must overcome is formal structure. This is the overall plan or structure of the music as it plays from beginning to end. While this may seem insignificant compared to the challenges of learning sound design, processing, effects, composition, and the DAW, forming a great arrangement is surprisingly tricky.

Our main objective is to create a memorable yet straightforward motif for our music. But when you consider that a motif may last 20 seconds or so, and most dance tracks will run anywhere from four to eight minutes, it's not as easy as it may appear. Indeed, merely repeating the same theme incessantly for six minutes, or dropping instruments in and out of a mix in a mechanical fashion, will not maintain a listener's interest. It requires careful planning and a range of finely honed techniques.

Before we can approach an arrangement, we must ensure that our initial idea contains enough musical information to build from. Without this, the arrangement will fall flat because you're trying to build on inadequate foundations. Knowing what constitutes 'enough' musical information is difficult to quantify, but it can be summed up in a motif that features some melodic or rhythmical diversity. This means your motif must contain the binary phrasing and call and response actions examined earlier in this book. If these do not exist in the pattern of the music, then it'll be difficult, if not impossible, to resolve.

Resolve is the ultimate reward in all forms of motion art and is the critical ingredient for us to find it satisfying. We find enjoyment in the accumulation of tension followed by the subsequent release of that tension. This cycle of tension and release forms the basis of all great films, books, computer games, and music. Movie directors, script writers, and authors of fiction elevate the tension levels of the audience so that when the resolve occurs, there is relief and enjoyment.

In motion art, the cycle is accomplished through narrative or dramatic structure. The Greek philosopher, Aristotle, is believed to be the first to have recognized the importance of structure by declaring that all stories must have a beginning, middle, and an end. Gustav Freytag, a German novelist, developed on this theme by stating that any great story should consist of five stages:

- Exposition. This stage introduces the main protagonist and additional characters and sets the scene for the story.
- Incident. A problem will be introduced, or a conflict entered, that will serve to drive the rest of the story to its conclusion.
- Rising action. Events will grow ever more intense, and the problems will increase exponentially. The protagonist will be faced with numerous difficult situations that go towards improving the tension of the story.
- Climax. Here, there will be a significant turning point. Any number of situations will change, and the odds will often stack significantly against the protagonist. This will create the highest suspense in the story.
- Resolution. The protagonist will triumph, and standard service is resumed. Here the reader or viewer receives the most substantial reward, and the story is often typically resolved to a happy ending.

If you consider any great story, you'll be able to identify with this arc. Whether it's a classical play, the best-selling novel, or a box office smash, they are all based on this underlying structure. Without a beginning, middle, and end, there can be no story. While the majority of the tension and final resolve is created through this story arc, all stories will introduce smaller periods of tension and resolve throughout to maintain the audience's attention.

In an action film, this is accomplished through some chase scene part way through the movie. The suspense is built through the thrill of the chase as the protagonist chases the antagonist. At this early development, the antagonist will escape the situation, but this act offers both resolve and increases tension. There

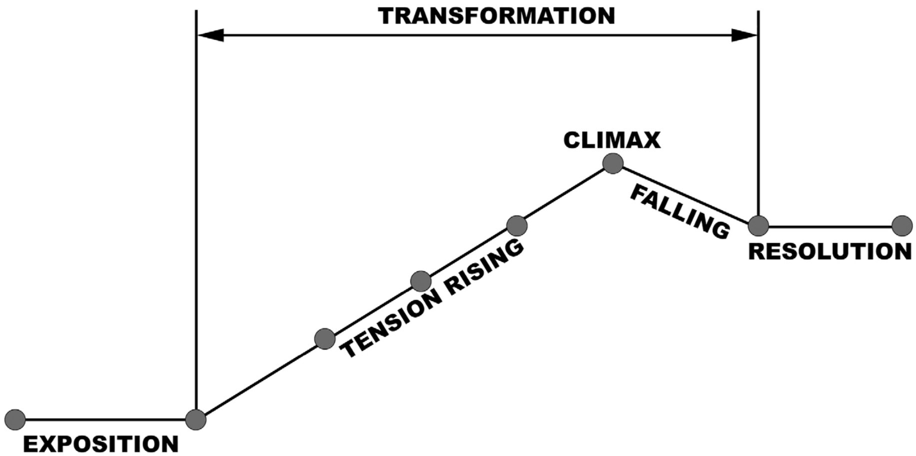

FIGURE 26.0
The dramatic structure

is resolve from the chase as it ends with the protagonist surviving, but background suspense is increased to the main story arc as the antagonist escapes.

This same aspect of development applies to musical arrangements but rather than employ dramatic or narrative structure; it uses a sonata form.

SONATA FORM MUSICAL ARRANGEMENTS

Exposition (sonata form)

Similar to the exposition in a story, within many styles of music this sets the scene. This is accomplished by introducing the central musical theme to the listener. In pop music, the exposition introduces the entire chorus or part of it. In electronic dance music that is destined for radio, this commonly involves the lead motif but lacking in full presence or missing the complete drum section or bass. Examples of this can be heard in countless radio-friendly versions of dance music records.

Of course, not every track destined for the radio will follow this form of exposition; 'Greyhound' by Swedish House Mafia is an excellent example of a different style of exposition, but this should be considered an exception rather than the norm.

Development (sonata form)

During the development stage, the music will move away from its defining motif or melody and develop the song. In pop music, this consists of the verse that tells the story of the music, while in classical music, the composer would take this opportunity to improvise on the melody, through using different instruments to changing key or pitch while maintaining the same melodic rhythm.

Within dance music, however, we place a much higher focus on textural development and motion, and production ethics. So for development, the lead may be thinned with EQ or filters, or it may only play a part of the lead. It may be exchanged for a different timbre or dropped altogether to focus on the groove of the music.

Recapitulation

For the recapitulation, the main melody returns once again but with all instrumentation in place. In pop music, this is the final chorus to drive the message home. In dance music, it is the drop: the complete reveal of the 16 or 32-bar focus of the track that was initially conceived by the producer.

Of course, on casual listening, a six- or eight-minute club based music may not appear to follow sonata form, but more critical listening to its structure will reveal it is fundamentally the same. Although the music may begin with percussive elements, the exposition is the introduction of a groove. Development is the build on that groove and recapitulation is the final reveal of the theme.

Indeed, regardless of whether the music is a three-minute radio mix, a five-minute Beatport mix, or an eight-minute club opus, the purpose is the same as dramatic or narrative structure; to create a series of events that build and release tension. With all music, this is designed by teasing the listener with the prospect of playing the 'best' part of the record, the complete motif.

In pop music, the motif is often revealed at the very early stages, and if the listener likes the theme, they feel a sense of anticipation waiting for it to return. Similarly, with dance music, it is our job to tease listeners with the main motif and employ techniques such as snare rolls of filter sweeps to build the anticipation. We then offer the resolve by exposing the whole motif with all instrumentation.

However, whereas pop music employs a verse and chorus progression, dance music is based on the continuous development on a theme, and this is where many novice producers struggle. Developing on a theme within a DAW is difficult because we're working with nothing more than a series of 'events' or 'regions.'

These are displayed as graphical blocks on the arrange page in a DAW, and therefore it's easy to fall into the trap of treating these as a series of *fixed* musical ideas that should be repeated across the arrange page as and when required. While this technique sometimes works for pop music with a verse and chorus structure, it is not applicable to the development of a theme.

The most common pitfall of many novice producers is to oversimplify an arrangement by repeating the same pattern over and over but with reduced instrumentation. For example, they will duplicate the same event pattern over the length of the arrangement and then systematically unmute channels as the arrangement progresses to its apex. This approach fails to produce an exciting listening environment because it cannot create any anticipation within the music. The only expectation is that a new instrument will be introduced in the next so many bars *ad infinitum* (or, perhaps, *ad nauseam*).

To develop a great arrangement, you must break free of this event based approach and rather than think vertically regarding channels, think *horizontally* regarding time. We must carefully manipulate both the energy and contour of the music.

ENERGY AND CONTOUR

Energy in music is associated with both pitch and frequency. If the pitch of a melody rises or the frequency content of the track or timbre is increased, then we determine this as an increase in the overall energy of the music. Examples of this can be found in all forms of music.

Many pop records will move to a higher pitch for the last chorus, the talent will perform in a higher range and often her vocals, or other instruments, will be layered to increase the frequency content. This helps drive the message home and provides the ultimate release for listeners.

Similarly, many dance music records will employ a low-pass or high-pass filter, removing a significant amount of energy from the frequency range and slowly open the filter to build up energy and anticipation in the listener.

DMM resource: The dance music rising filter used to control energy and anticipation.

The control of a songs energy by employing these techniques form one of the key ingredients to a successful arrangement. The manipulation of different energy levels through an arrangement is called *contour*, and any strong structure will have a carefully manipulated contour. This can be evident by loading any dance music track into a DAW and examining its waveform. Careful examination will reveal the energy contour of the music.

As shown in Figure 26.1, despite the record based on the development of a theme, a relatively small amount of time is spent playing the complete motif. The only area for this example where the entire motif is exposed is close to the end that, comparatively, is an insignificant 32 bars of music. The rest of the music is spent developing towards this finale.

This is where many novice producers fail. After spending days, weeks, or even months creating the motif of the record, they try to hit the listener with their idea at every opportunity when it should be played as little as possible and the majority of the track should be spent teasing the listener with the *prospect* of playing it.

Teasing a listener and maintaining this for the length of an arrangement employs a variety of techniques from filters to syncopation, denial of expectation, and canonic behavior. How these are applied is down to our own creative choices, and I'll discuss these in more detail shortly but a more critical question is *where* these should occur.

If you have listened to electronic dance music for any length of time, you'll instinctively know when to expect a change to occur within the track. This is because instruments or textural changes occur at specific metrical cycles in the music. These positions are called structural downbeats and refer to the first downbeat of the bar where a new instrument or textural development occurs.

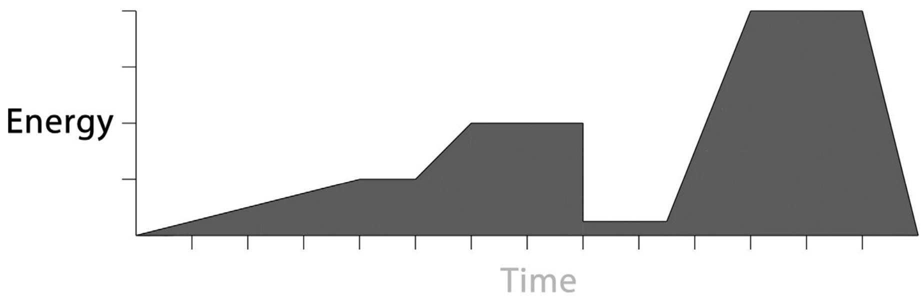

FIGURE 26.1
The energy contour of a dance music track

In almost every instance of electronic dance music, a structural downbeat will occur at multiples of two. So, they will happen on the eighth, 16th, 32nd, or 64th beat of the bar. At one of these positions, it is common to introduce change, whether this is in the form of a new instrument, removing instruments or employing a significant textural difference to the arrangement. Of these, with radio-friendly music, the changes will often occur at every eight or 16 bars while in club based music they will typically happen at every 16th, 32nd, or 64th bar.

DEVELOPMENT

Once you have determined the point to introduce a new instrument, it can be accomplished in any number of ways. One of the most popular techniques is to precede the introduction with a reversed cymbal crash or a white noise riser.

These create a build to the introduction of a new instrument and can be heard in almost every dance music track ever produced. However, if the same process is employed to introduce *every* instrument into the mix, it becomes second nature to the listener and anticipation is lost.

Similarly, consistently using the same approach produces a mechanical feel and reduces any emotional impact. Instead, you must use varied techniques for the introduction of different instruments such as reverse cymbal crashes, white noise, reverse reverb, and slow filtering of an instrument.

The reverse reverb effect consists of taking a single note of audio from the instrument and reversing it in the DAW's sample editor (remember to must first duplicate the event to prevent *all* audio in the project from being reversed!). This new modified event is placed onto another audio channel and reverb is inserted into the channel with a short pre-delay and very long tail, commonly more than three seconds.

The audio is then bounced to an audio file, reimported into a new channel in the DAW, and reversed *again*, so that the audio is the correct way around. This results in the previously stamped reverb now being reversed and sweeping upwards *into* the note.

Similarly, filters can be employed to introduce an instrument over any number of bars slowly. This is accomplished via a high-pass filter, but bandpass and low-pass are also applied depending on the mix and our own creative decision.

In the examples on the resources, I began by employing a crash to introduce both the hi-hats and the bass. I then followed this by using the crash just to add the hi-hats followed by high-pass filtering the bass. Finally, I employed reverse reverb on the bass and a high-pass filter to sweep in the hats. These are all techniques that can be used to create a more exciting introduction to instruments.

DMM resource: Introducing instruments with a crash, reverse reverb, and filters.

While introducing instruments is the most skillful way to build energy within an arrangement, if it occurs every time you feel the mix requires further energy, you will quickly run out of instruments. Therefore, some other techniques must be used to maintain interest.

Syncopation is often used during an arrangement because it produces a strikingly powerful effect. It consists of changing the velocity of different notes in a motif or melody every four or eight bars. This results in a continual shifting emphasis of notes within the patters and can maintain sustained listening.

In the example that follows, I employed velocity syncopation to the bass line, but it could equally be applied to any other melodic or non-melodic instrument. In Tech House, for example, a conventional technique is to employ a pulsing bass but syncopate the velocity of this bass differently for each bar.

DMM resource: The effects of syncopation on a bass line.

Alongside syncopation, for melodic elements, you can reduce the melody of the event but maintain its rhythm. This effect is evident in some House records but is typical of the different forms of Trance. If the bass or lead is heavily melodic, it is reduced to a single pitch. This maintains the original rhythm of the melody but at a single note for some bars before some of the original theme occurs by introducing the original pitch to some notes.

A reverse cymbal crash can be used to 'introduce' a new pitch element to the melody. Alternatively, if the track happens to be rhythmically based, only part of the rhythm is revealed with any silence filled by careful application of delay.

The effect is a sense of anticipation in the listener because they want to hear the whole melodic or rhythmical element. This should be applied cautiously, however, because denying expectation for too long can turn your listeners off. But then, introducing the complete melody too early can have the same effect.

This is where experience, practice, and critical listening of your peers should not be underestimated. In the following example, I maintain a single pitch in the bass but then introduce some pitch after a short reverse cymbal strike ...

DMM resource: Pitch development on a bass.

An alternative to introducing pitch movement early in an arrangement is to employ a *canonic* drone to determine the pitch movement before it occurs. This is popular in House scene but does happen in different styles.

The term 'canonic' is derived from 'musical canon,' used to describe the effects of following one melodic line shortly after with an exact imitation of the same melody. A good example would be someone singing 'Row, Row, Row Your Boat,' whereon a second voice enters, repeating this first line while the original vocal begins to perform the second line of 'gently down the stream.' This overlapping imitation of melodies from numerous performers results in the musical 'canon' effect.

In dance music, the canonic effect involves a single note drone, commonly a bass drone, playing what would primarily be the underlying motions of the bass or lead. For example, let's say that the first note of the bass was at A on the first bar, then E on the second, F on the third, and then D on the fourth bar. In this instance, regardless of the pitch of all other notes in each bar, the canonic would play A for one bar, E in the next, followed by F and then D. Figure 26.2 shows this relationship between the canonic and a bass melody.

This canonic behavior is similar to harmony but on a more simplified scale with no requirement to cadence. What's more, the canonic will usually occur while the bass and melody are at a single pitch. This produces anticipation in the listener because they are made aware of the upcoming pitch movements, but are not yet presented with them in the bass or lead.

This effect can become more interesting if the canonic is syncopated. For instance, in the previous example of A, E, F, and D over four bars, the bass could remain at A throughout but repeated over eight bars. Here, the canonic would play a sustained A for the first five bars, before moving to E, F, and D for a bar each.

This provides a lengthened aural clue as to the upcoming note movements of the bass, but with the canonic expressing this over eight bars rather than four, it creates a syncopated relationship to the forthcoming bass motion. Figure 26.3

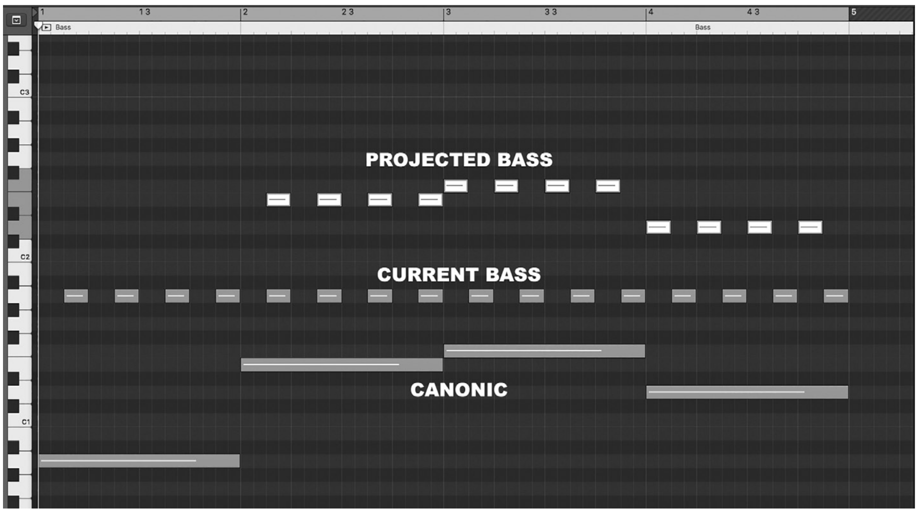

FIGURE 26.2
The canonic underlying the bass melody

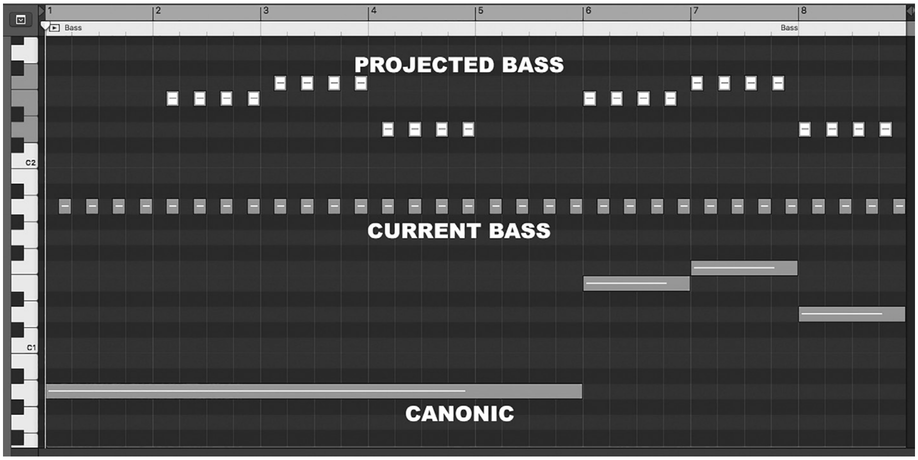

FIGURE 26.3
A syncopated canonic

shows the effect of a syncopated canonic, and the result can be heard in the resource pack.

> DMM resource: A syncopated canonic example.

A further technique is to employ processor or effect development on a timbre. Although all sounds within dance music should exhibit some form of textural development during their period, the same also occurs over the course of an arrangement.

Slow modulated changes in filter, distortion, or any other effect can be used to reveal the timbre of the music slowly. For example, a lead may play the complete melody but be treated to a high pass filter that slowly opens to allow more low frequencies through over a period of 32 or 64 bars.

Alternatively, high feedback on a delay unit or automated reverb may wash over the sound, and these parameters are reduced gradually via automation until the instrument becomes distinct. This effect is common in Progressive House: the timbre is exposed to vast amounts of reverb that is increased or reduced to introduce motion and interest into the arrangement. In the following example, I employed a long reverb tail and high-pass filter to bring attention to the timbre slowly.

> DMM resource: Automated reverb and delay on a lead.

All of these techniques create some anticipation or excitement in the listeners, but perhaps the most powerful is to deny expectation. This can lead to increased tension and a stronger resolution when the music is finally resolved. To accomplish denial, however, the listener must be able to form an expectation of what is about to, or *should*, occur.

There are a variety of ways to achieve denial, but the most common is to introduce the complete binary phrase with limited backing instruments, a different synthesizer voice, or heavily filtered lead. By doing so, the listener is presented with the whole phrase but not in the complete context of the music. With this approach, the tension is increased because they want to hear it with all instrumentation.

This tension can be increased further by slowly introducing the full timbre of the motif but as this occurs, reducing to only its first part and withholding the resolve. Through careful manipulation of effects and filter such as increasing a reverbs tail or increasing the delay time on the lead or bass instrument, it's possible to construct a small build in the music. If this is followed by slowly increasing a filters cut-off and then playing the complete binary phrase again, it will create a form of expectation in listeners; they will expect the music to return in full form. However, rather than return with the complete music, the music returns with additional instrumentation while still not revealing the entire motif of the record. I've employed this denial technique in the example on the resources.

> DMM resource: Creating a denial and increasing tension.

Of course, this is just one instance, and there are many more. A small snare roll, a snare skip, or a riser can make the listener feel the music building, but then the track drops instruments rather than introducing more. Just listening to your favorite artists, music, and identifying these techniques to control your emotion will reveal the most common application for the genre of your choice.

Whatever method is employed, however, it must be applied cautiously because denying a listener his expectations too many times or for too long will break the tension and the music will lose its cohesion and stability.

THE MAIN BREAK

All these techniques combined allow us to construct an arrangement that develops on a theme and leads towards the main drop. The overall contour of the music up to this point is down to your taste and the genre of music, but typically the energy will be controlled to build to its highest level thus far before the breakdown occurs.

The breakdown is the precursor to the drop where the beat drops back in. A breakdown involves removing most of the instrumentation from the mix. The lead-up into this drop can be controlled in any number of ways from snare rolls, snare breaks, synthetic risers, filters, increasing effects parameters, or all of the above.

You can program snare rolls and breaks in any DAW piano roll editor. A roll can begin with snare hits positioned on every beat for the first bar or two, then become more frequent occurring at every beat and 1/8th, then at every 16th and then, for the final bar, at every 32nd. During this time, a low-pass filter may be automated to open, or the velocity may be used to increase the volume of the snare as it progresses. Figure 26.4 shows the development of a snare roll in a MIDI editor.

DMM resource: A basic snare roll.

MIDI of the snare roll.

Further interest can be added to a snare roll through the application of effects such as delay, phasers, flangers, and reverb. These are all automated so that the parameters such as delay time and reverb tail time gradually increase as the snare roll reaches its apex. By doing so when the roll ends, the reverb and delay can echo across the mix as the instruments are removed for the final drop.

DMM resource: A snare roll with automated delay, reverb, and phaser.

This is only one approach, and sometimes a snare roll may not be employed at all. A more relaxed technique is to introduce a snare skip. This involves placing a few snare hits just before the fourth beat of the bar to produce a skip in the rhythm.

Alternatively, in genres such as Progressive House, it's common to automate a filter on the lead so that it opens allowing more frequencies through, while

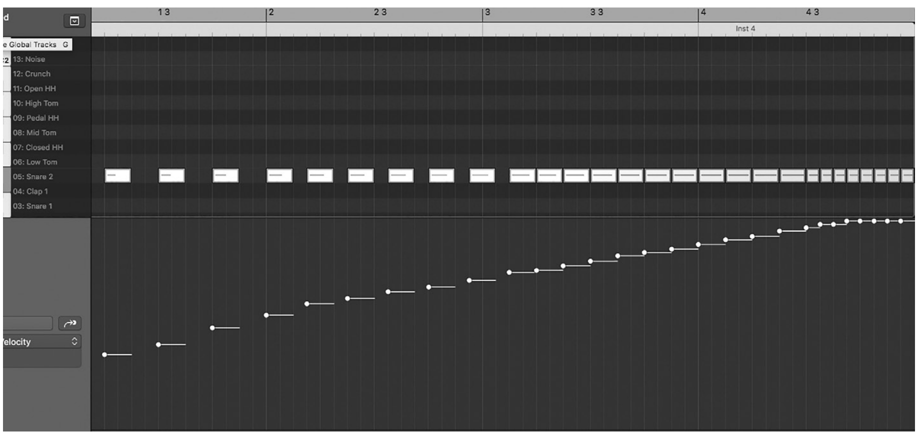

FIGURE 26.4
A snare roll

at the same time, reverb and delay times are increased. This results in a robust building effect so that when the music reaches the main breakdown, the reverb and delay can echo across the mix or stop abruptly behind a falling riser.

When the music reaches the drop, the most commonly applied effect is the low-pass filter. This, above all, offers the most powerful way to control the energy and contour of a build. How the breakdown is handled depends entirely on the genre of music in question, and close listening to your favorite styles will reveal the commonly accepted techniques.

Typically, the breakdown will consist of only a few instruments that are filtered or all the instruments filtered. This filter is automated to open slowly to create the ultimate tension and build and may be backed with another snare roll or a riser until it reaches its apex. When the final drop occurs, the hook is finally introduced, and the resolve is presented to the listener.

DMM resource: A typical build-up to a finale (with a single bar denial).

Of course, everything discussed here is open to experimentation, and creative input. It's a close analysis of your favorite music that will reveal the most common arrangement techniques that are employed in that particular genre. Having said that, if you analytically listen to the above methods in any dance music track, there is no doubt that most – if not all – will be employed in the arrangement.

They will often be mixed with a variety of other applications drawn from your creativity, experimentation, and experience but they will typically include most of the techniques mentioned above.

CHAPTER 27

An overview of House music

House music is what we all need. It lifts our spirits and makes us want to get down.

Anonymous

The development of House music has much of its success accredited to the rise and fall of 1970s' Disco so to investigate House, we should start with Disco. Pinning down the exact point at which Disco first appeared is difficult. A majority of the elements that make Disco had already appeared in earlier records but it first originated in the early 70s and was a natural evolution of funk music.

Influenced by Funk, big-name producers such as Nile Rodgers, Quincy Jones, Tom Moulton, Giorgio Moroder, and Vincent Montana began to hire session musicians so they could produce Funk-inspired hits. They used artists whose only purpose was to supply vocals and become a marketable commodity (sound familiar?). Donna Summer was one of the first success stories with the release of 'Love to Love You Baby' in 1975. This is believed by many to be the first Disco record to be accepted by the mainstream public.

This form of music was still in its infancy, however, and it took the release of the motion picture *Saturday Night Fever* in 1977 for the musical style to become a widespread phenomenon. By the late 70s, over 200,000 people were attending discotheques in the UK alone, and Disco records contributed to over 60% of the UK charts.

As with most genres of music that become popular, record labels jumped on the bandwagon. The style became deluged with countless Disco versions of original songs, and many were rushed and poorly produced to make money as quickly as possible.

In the early 80s, Disco was dealt a further damaging blow by the 'Disco sucks' campaign. Steve Dahl, a Rock DJ, encouraged Rock fans to bring Disco records to a baseball game on 12 July 1979 for a ritual burning. After the game, a huge bonfire was lit, and the 'fans' were asked to throw Disco vinyl onto the fire. For many who attended, it wasn't for the music but instead a racial and homophobic attack.

By 1982, Disco was dead. But during its heyday, it changed the face of club culture, the balance of power between smaller and major labels and prepared the way for a new wave of music. Out of the ashes rose the phoenix of House.

House had been steadily growing into an underground movement before the death of Disco. In fact, it had been in its early stages of evolution *before* Disco hit the mainstream public and its foundations can be traced back to 1970.

Francis Grosso, a resident DJ at a converted church known as the *Sanctuary*, is believed to be the first DJ to mix two early Disco records to produce a continuous groove. He is also alleged to be the first DJ to mix one record over the top of another, a technique that was to form the genesis of dance music culture.

Drawing inspiration from this form of mixing, DJ Nicky Siano set up a New York club known as *The Gallery*. He hired Frankie Knuckles and Larry Levan to prepare the club for the night by spiking the drinks with **L**ysergic **A**cid **D**iethylamide (LSD, or acid). In return, he taught both the basics of this new form of mixing and soon after they earned residency at different clubs.

Levan began at *The Continental Baths* while Knuckles started at *Better Days*. Six months later, Knuckles joined Levan at *The Continental Baths*. The two worked together until 1977 when Levan left to start his own club. During the transition, he was asked to DJ at a new club named the *Warehouse* in Chicago. He refused but recommended Knuckles who accepted the offer and moved to Chicago.

The Warehouse had no music policy, so Knuckles was free to experiment and show off the techniques Nicky Siano taught him. Word spread about this new form of continuous Disco, and the Warehouse became the place to be for the predominantly gay crowd. The term 'House' was used to describe the scene. It represented not only the continual mixing style but also the clothing and the attitude. If *you* were House, you attended all the cool clubs, wore the 'right' clothing, and listened to 'cool' music.

By early 1983, the popularity of the Warehouse began to demise. This was partly due to the reputation of the club spreading to the mainstream audiences but also due to changes in musical style to cater for this new audience. The music was no longer 'House.' The owners began to play commercial music and doubled the admission price. Unhappy with this shift, Knuckles left and started his own club known as the *Powerhouse* and his devoted followers went with him. In retaliation, the Warehouse was renamed the *Muzic Box*, and the owners hired a new DJ named Ron Hardy.

Hardy wasn't a doctor, but he dabbled in numerous pharmaceuticals and was addicted to many of them. However, he was equally a very talented DJ. While Knuckles maintained a clean Disco sound, Hardy pounded out an eclectic mix of beats and grooves. He mixed Euro Disco, Funk, and Soul to produce an endless onslaught to keep the crowd up on the floor. His relentless form of mixing earned him the nickname of *Heart Attack Hardy*. To this day, Ron Hardy is viewed by many as the greatest ever DJ.

Riding on this new wave, WBMX, a local radio station broadcast late night mixes made by the *Hot Mix Five*. The team consisted of Ralphi Rossario, Kenny 'Jammin' Jason, Steve 'Silk' Hurley, Mickey 'Mixin' Oliver, and Farley 'Jackmaster' Funk. These DJs played a non-stop mix of British New Romantic music ranging from Depeche Mode to Yazoo and Gary Numan alongside the latest music from Kraftwerk, Yello, and George Clinton. In fact, so popular was the UK new romantic scene that one-third of the American charts consisted of UK music.

However, it wasn't just the music that the people tuned in for; it was the mixing styles of the five DJs. Using techniques that have never been heard commercially, they would play two of the same records simultaneously to produce phasing effects, perform scratches, and backspins. Due to the show's popularity, it was soon moved to a daytime slot, and kids would skip school just to listen to the latest mixes. It became such a favorite show that Chicago's only dance music store, *Imports Etc.*, began to put a notice board up in the window documenting all the records that had been played the previous day.

By 1983 Frankie Knuckles was suffering from a lack of new material. The 'Disco sucks' campaign had destroyed the industry, and the labels were no longer producing Disco. Instead, he had to turn to playing imports from Italy (the only country left still producing Disco), alongside Dub-influenced music.

He also turned to longtime friend Erasmo Rivieria. Erasmo was studying sound engineering, and Knuckles asked for his help to create reworks of the earlier Disco records to keep his set alive. Using reel-to-reel tape recorders, the duo would record and cut up records, extending the intros and breakbeats, and layering new sounds to create more complex mixes. They began to experiment by placing entirely new rhythms and bass lines underneath familiar tracks.

While this formed the basis of House music as we know it today, no one had yet released a House record. Although artist Byron Walton (aka Jamie Principle) undoubtedly produced the first ever House record with 'Your Love' (a timeless classic even to this day), every record label he sent it to turned him down. It only ended up in the clubs thanks to Frankie Knuckles, who made copies and distributed it to DJs. In the end, the first vinyl House record was Jesse Saunders' release of 'On and On.' In 1984, it was released under his self-financed label Jes Say and distributed through Chicago's Imports Etc.

'On and On' was pressed courtesy of Musical Products, Chicago's only pressing plant, run by Larry Sherman. Taking an interest in this scene, Larry investigated its influence over the crowds and started the first ever House record label Trax. At the same time, another label, DJ International, was started by Rocky Jones and the following years involved a battle between the two to release the best House music.

Many of these consisted of what are regarded as the most influential House records of all time including 'Music is the Key,' 'Move Your Body,' 'Time to Jack,' 'Get Funky,' 'Jack Your Body,' 'Runaway Girl,' 'Promised Land,' 'Washing Machine,' 'House Nation,' and 'Acid Trax.'

By 1985 House was in full swing, while still borrowing heavily from 70s' Disco, the introduction of the Roland TB303 bass synthesizer along with the Roland TR909, TR808, and the Juno 106 had given House a harder edge. It became Disco made by 'amateur' producers. The basses and rhythms were no longer live but recreated and sequenced on machines resulting in a host of 303-driven tracks.

One of these budding early producers was Larry Heard, who, after producing a track entitled 'Washing Machine,' released what was to become one of the most poignant records in the history of House. Under the moniker of *Mr. Fingers*, he released 'Can U Feel It,' the first ever House record that didn't borrow its style from earlier Disco. Instead, it was influenced by Soul, Jazz, and the Techno that was evolving from Chicago. This introduced a whole new idea to the House music scene, as artists began to look elsewhere for influences.

One of these was Todd Terry, a New York Hip-Hop DJ. He began to apply the sampling principles of Rap into House music. Using samples of previous records, he introduced a dominant percussive style to the genre and released '*three massive dance floor House anthems*,' that pushed House music in a whole new direction. His subsequent House releases brought him insurmountable respect from the UK underground scene and has duly been given the title of Todd 'the God' Terry.

Over the following years, House music mutated, multiplied, and diversified into a whole number of different subgenres, each with its own name and production ethics. To date, there a multitude of different subgenres of House, with Progressive House, Hard House, Deep House, Future House, Bass House, Micro House, Euro House, French House, Tech House, Vocal House, Swedish House, Commercial House, and EDM. The last used to stand for **E**lectronic **D**ance **M**usic, but it's since been contorted by journalists and the USA to cover a form of Mainroom House.

MUSICAL ANALYSIS

The divergence of House music has resulted in a genre that has become hugely fragmented and as such cannot be easily identified as featuring any one particular attribute. It can be funky drawing its inspiration from Disco of the seventies; it can be relatively slow and deep drawing inspiration from Techno; it can be vocal; it can be party like; or it can merely be pumping.

In fact, today the word House has become somewhat of a catch-all name for music that is dance (not pop!), yet doesn't fit into any other category. The good news with this is that you can pretty much write what you want and, as long as it has a dance vibe, it could appear somewhere under the House label. The bad news is that it makes it near impossible to analyze the genre in any specific musical sense and it is only possible to make some very rough generalizations.

House music will invariably use a 4/4 time signature and is produced in almost any musical scale depending on the genre of House. For example, Disco House is regularly composed of a major key while the darker Tech House is generally in a minor key.

For this chapter, we will examine the general production ethics in the creation of a Tech House track, but this approach equally applies to many genres of House. As music is an entirely individual, artistic endeavor, the purpose of this analysis is not to lay down a series of 'ground rules' but instead will describe some of the general principles behind the production. In the end, it's up to the individual (i.e., you) to experiment and create music that suits your particular style.

Whatever the House genre, many House records are still produced in A minor. This is most likely because DJs create a proportionate amount of House records, and it is easier to mix them harmonically. Plus, the most influential and powerful area for dance music's bass reproduction on a club speaker system is 50 to 65 Hz, and the root note of A1 sits at 55 Hz. Since the music will modulate or gravitate around the root note of the scale, having the root note of the music at 55 Hz will help to maintain the bass 'groove' energy of the record at a particularly useful frequency. Regarding tempo, it can range from a somewhat slow 110BPM to a more substantial 140BPM.

House is produced in one of three ways: everything is sampled and rearranged; only some elements are sampled, and the rest is programmed; or the entire track is programmed via virtual instruments. The approach taken depends entirely on what style of House is composed.

For example, the Disco House produced by the likes of Daft Punk (previously, Random Access Memories) relied on sampling significant parts from previous Disco hits and dropping their vocoded vocals over the top. If you write this style of music, then this is much easier to analyze because it's based on the Disco vibe. It consists of a four to the floor rhythm with a heavily syncopated bass line and the characteristic electric guitar.

In contrast, Deep House uses much darker timbres (deep bass lines mixed with atmospheric jazzy chords) that don't exhibit a happy vibe but are still danceable. Tech House leans more towards the Techno side of production with an emphasis on intricate beats and sub basses but with House stabs or chords.

Because there are so many different genres of House, to cover them all would take the rest of this book. So, what follows is merely a guide to producing the essential elements of all types of House, and since you know how your choice of the genre already sounds, the rest of the genre chapters can be adapted to suit what you hear.

HOUSE RHYTHMS

House relies heavily on the strict four to the floor rhythm with a kick drum laid on every beat of the bar. Typically, this is augmented with an open hi-hat positioned on every 1/8th (the offbeat) and a closed hat on alternating 16th

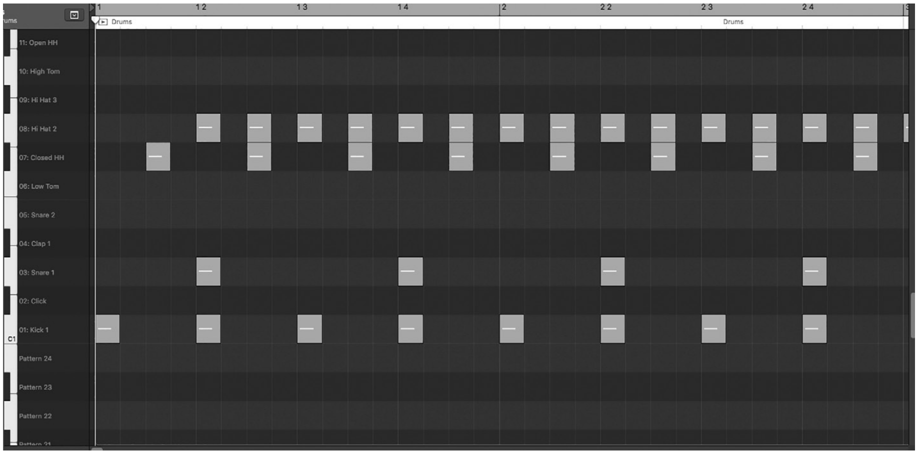

FIGURE 27.0
The fundamental cornerstone of a House drum loop

notes. Snares (or claps) are also often employed on the second and fourth beat underneath the kick. This produces the primary loop for many genres of dance music, including House, and was discussed in detail in Chapter 16 on constructing loops.

Notably, House music, perhaps more than any other dance genre, relies on plenty of drive. This can be applied in a variety of ways, but the most commonly used technique is to introduce a slight timing variance in the rhythm to create a push in the beat. This involves keeping the kick dead on all the beats but moving the snares or claps forward in time by a few ticks. This produces a small surge in the rhythm that is typical of many House genres.

For House kicks, the original synthesizer of choice was the somewhat ubiquitous Roland TR-909, but this is no longer considered to have enough low-frequency energy or power. Over recent years there appears to be an unwitting competition between producers for who can produce the most powerful kick drum.

The somewhat atypical powerful kick drum in almost all forms of House music is created by layering kick samples or drum synthesis. These techniques were discussed in detail in an earlier chapter, so I won't repeat it all here.

The House kick is almost always treated to small amounts of reverb. Here a room style reverb is a popular choice with a long pre-delay that skips the transient of the kick so that it can cut through the mix. If the layered hi-hat or square wave is layered to create a transient, then these are left unaffected. This lets you employ a shorter pre-delay on the main kick. Diffusion should be kept low, and the tail should remain short so as not to wash over further instruments.

One of the critical elements of many House loops is that the individual sounds remain short and transient as this provides a more dynamic feel to the music and can even differentiate between different genres of House. Indeed, styles such as Funky House, French House, and Disco House employ rhythms that flow together whereas genres such as Minimal and Tech House will have each instrument cut short to produce a snappy, tight, controlled feel to the music.

For this, a noise gate can be inserted after the reverb and on each instrument channel to control the decay of each instrument. Or you can bounce the instruments into audio and just cut the audio tail off in the DAW.

House kicks rely heavily on compression to produce the archetypal hardened sound. Although the settings to use will depend on the sound you wish to achieve, a good starting point is to use a low ratio combined with a high threshold and a fast attack with a medium release. The attack will crush the initial transient producing a more substantial 'thump' while experimenting with the release can further change the character of the kick. If this is set quite long, the compressor will not have recovered by the time the next kick begins and, depending on the compressor, can often add a pleasing distortion to the sound.

Similar to the kicks, the snare or claps were often sourced from the TR909 but could also be derived from the E-Mu Drumulator due to its warmer, rounded sound. However, to attain the typical sound of many House records that are prevalent today, the snare (or clap) is usually born from samples of the 909 or layered samples.

If you're creating the more experimental forms of House, you may wish to synthesize your snares. You can do this using either a triangle or square wave mixed with some pink noise or saturation (distortion of the oscillator). These are both modulated with an amp envelope using a fast attack and decay.

Pink noise is preferred over white, but if this proves to be too heavy for the loop, then white noise may provide the better option. Much of this noise is removed with a high-pass filter to produce the snare, but in some cases, a bandpass filter may be preferable because this allows you to modify the low and high content of the snare more accurately.

If at all possible, it's worthwhile using a different envelope on the main oscillator, and the noise since this will allow you to keep the body of the snare (i.e., the central oscillator) quite short and sharp but will enable you to lengthen the decay of the noise waveform. By doing so, the snare can be modified so that it rings out after each hit so you can individually adjust the decay's shape of the noise waveform.

Typically, the snares or claps in House are treated to a room style reverb but the length of the reverbs tail will depend on the genre of music. The reverb's tail is longer than a kick in most types but the pre-delay changes. In genres such as Tech House, it's not unusual to use a very short pre-delay so that the transient is molested. In some styles of House, reverse reverb is often used on the snare, but this is typically limited to the second beat or will only occur at structural downbeats.

Compression is sometimes used on the snares/claps, but this depends on the genre of House music. Here compression is employed to modify the tonal character of the snare. If applied excessively with a fast attack, the compressor would clamp down on its attack stage diminishing the transient, reducing the dynamic range between the transient and body. This results in a more powerful, prominent snare or clap that exhibits a 'thwack' style sound.

Additionally, microtonal adjustments or filter modulation is requisite on the snares or claps to produce the motion between the two beats in the bar. For 'Hot Chocolate' (the track used as an example in this chapter), I used *Soundtoys' Filter-Freak* on the snare to control a low pass filters cut-off. The modulation was applied, so that first snare of the bar passes through a low-pass filter unmolested but the second strike occurring on beat four is subjected to a very light low-pass filtering that removes a tiny portion of the higher harmonics of the timbre. This introduced enough of a textural difference between the two beats to maintain interest in this part of the loop.

In many examples of House music, the hi-hats are taken from sample content. There are plenty of these to choose from on the market. They will often be treated to numerous processing effects from distortion to reverb to transient designers and compression. All these processors and effects have been discussed in detail and experimentation with all these effects will yield the best sonic results for the genre.

A common effect for hi-hats in House is a delay. The settings for the delay are up to you and your creativity, but it should be kept relatively short to prevent too much delay wash that may cloud the rhythmic interplay. Typically a 1/16th or 1/8th setting with a short delay time is suitable, but some employ a noise gate after the delay unit or sidechain compression. Applying a noise gate directly after the delay allows you to use larger settings but also prevents them from 'stacking' up.

An alternative is to employ the delay on a bus and place a compressor directly after the delay unit. The ratio of the compressor is set to 2:1 with a low output gain and a low threshold. The original hi-hats are then fed into the side-chain of the compressor. This way, each time a hi-hat occurs, the compressor activates ducking out any delayed hi-hats, and when no hat occurs, the compressor isn't activated, permitting the delayed hats to continue.

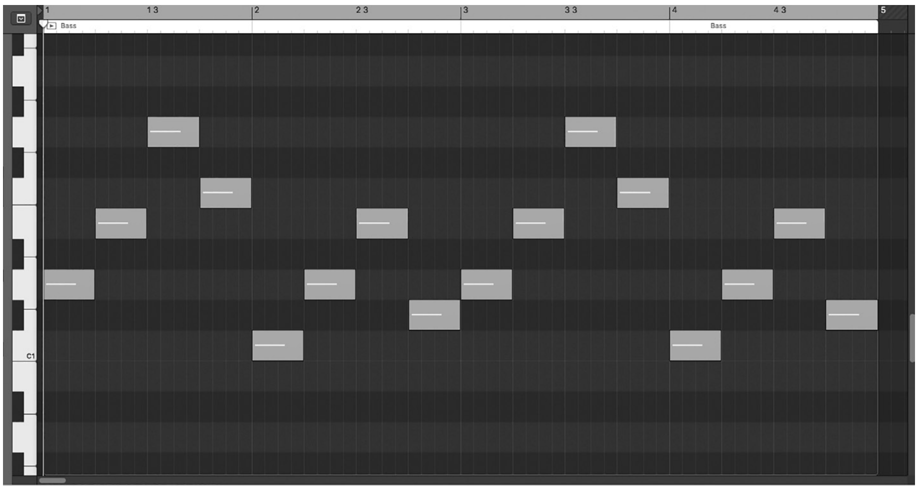

FIGURE 27.1
The secondary hi-hat pattern used in the books track. Note the use of velocity to create a moving rhythm

Both open and closed hi-hats will also be treated to velocity filter modulation cyclic microtonal pitch or filter modulation. Typically, the open hats are treated to a low-pass filter that is modulated via a sine or sample and hold waveform over a period of two or four bars. Closed hats are treated to microtonal pitch movement over a period of three bars. Three or six bars are chosen because this works against the standard structural downbeat of dance music and results in a cross-syncopation of modulation.

This approach produces the atypical basic House loop, but depending on the genre, further instrumentation will be applied. Congas, toms, bongos, tambourines, and shakers are often added to create a distinct feel to the rhythms. These are all commonly positioned on syncopated grid positions in the loop or will employ hemiola or/and polymeter depending on the genre. Many artists won't program these but will import sampled loops and cut them up to produce new and exciting percussive rhythms.

Genres such as Disco House, Euro House, and French House will maintain a reasonably basic rhythm because the emphasis is on the funky bass lines and chorded leads. But, in genres such as Tech House, there is a far stronger emphasis on the rhythm, and therefore they will employ more rhythmical techniques to maintain interest.

Hemiola has already been covered elsewhere in this book, but it consists of creating an unequal subdivision of beats within the bar and placing a hi-hat or percussive rhythmical pattern onto these different subdivisions. This effect can then be further augmented with the use of polymeter on further percussive elements.

A typical application of polymeter here is to employ a 5/4, 6/4, and 7/4 on synthesized percussive hits. These and, in fact, all percussive instrumentation within House music (except for the kick) will be subjected to groove or swing quantize. The amount applied ranges from genre to genre, but typically they all remain within 60–70% on the 1/16th.

Of course, it would be naive to say that all House beats are created through MIDI, or sample content and for some producers the rhythms are acquired from sampling previous House, Disco or Funk records. In the case of House records, these will have probably been sampled from other records, which have been sampled from previous House records, which have been sampled from other records, which will likely have been …well, there is no need to continue.

For obvious reasons, I couldn't condone infringing another artist's copyright, but that's not to say that some artists don't do it and so, purely in the interests of theory, I'll cover some of the techniques they use.

Almost every House record every pressed to vinyl starts with just a drum loop, so sourcing one isn't particularly difficult. The real skill comes from what you do with it in the audio editor. It's unwise just to sample a loop and use it as is. Even though there's a very high chance that what you're sampling is already a sample of a sample of a sample of a sample, you might be unlucky, and they may have programmed it themselves. What's more, it isn't artistically challenging.

Therefore, after sampling a loop, place it into a sequencer, and move the parts around to create your variation.

This is the most simplistic approach, and many artists will take this a step further by first employing a transient designer to alter the transients of the loop before placing it into a sample-slicing program or applying the workstations own sample-slicing facilities. Using this technique, the loop does not have to be sourced from a House track as any drum break can be manipulated to create the snappy sounds used in House.

Along with working with individual loops, many House producers also stack up a series of drum loops to create heavier rhythms. Todd Terry and Armand Van Helden are well known for dropping kicks and snares over pre-sampled loops to add more of their own influence.

If you take this approach, however, you need to exercise care that the kick and snare occur in the right place otherwise you'll experience a phasing effect. Possibly the best way to circumvent this is to cut the bar into four segments where the kick occurs and then trigger the kick or kick and snare alongside the originals at the beginning of each section to produce a continual loop that stays in time. This also permits you to swap the quarter bars around to create variations.

BASS

Almost every genre of House relies a great deal on the groove created from the interaction between the bass and the drum rhythm. Although the style has diversified dramatically from its original roots of Disco, except for a few House genres such as Tech and Minimal, the bass rhythm still has its roots firmly embedded in Disco. French House, Disco House, and Commercial House will all borrow heavily from the funky groove of the Disco-style bass. The simplest of these is the walking bass line that was used on countless Disco records.

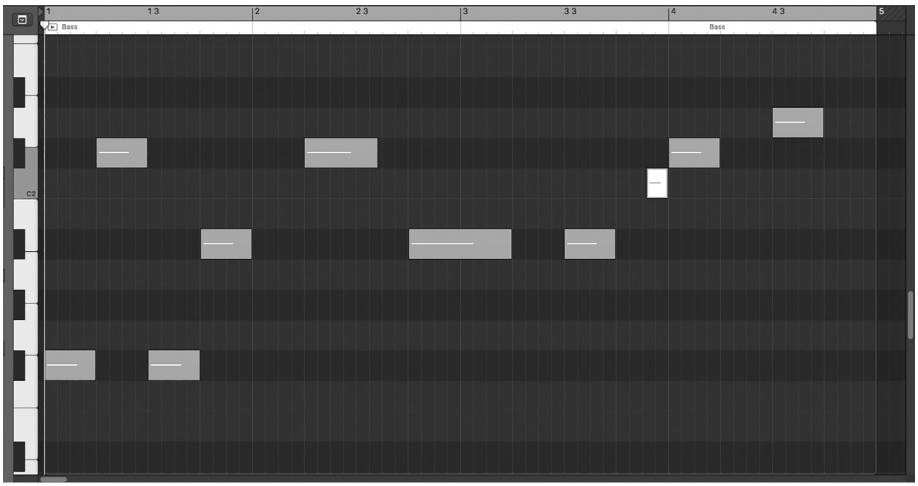

FIGURE 27.2
Disco's infamous walking bass line

The above bass line is a simple example that has been used in some dance tracks, but many Disco-inspired House records use a bass line written in the minor pentatonic scale (i.e., using only the black notes on the keyboard). Both major and minor pentatonic scales are shown below:

Minor: E♭ – G♭ – A♭ – B♭ – D♭ – E♭
Major: G♭ – A♭ – B♭ – D♭ – E♭ – G♭

In these scales, only five notes are available, but a further sixth and seventh are introduced as leading tones. These are notes that do not exist in the pentatonic scale but can occasionally be used to pass through to the next note and help to add to the movement and groove of the music. In the example below, the notes G and D, while not in the pentatonic scale, are used as 'stepping stones' that 'lead' into the next note in the pentatonic scale.

Figure 27.3 shows a typical Disco or Funky House bass line that exhibits a very similar groove to that of their original Disco roots. But the bass line of further House genres will follow a similar groove, just stripped back dependent on the style.

In some forms of House, the bass line is stripped back through the use of notes with a longer duration and, rather than resolving in the second bar (binary phrase) as it would in Disco House, it resolves over a period of 16 bars. As the genre changes and the bass line becomes stripped back further, it will begin to lean more towards Tech House. In fact, this genre shares more similarities with Techno than it does with House. For Skreener, I stripped back the bass line to just four notes to create a Tech House feel. To help further accentuate the Tech feel, I used some asynchronous positioning.

Whatever the genre of House, it's important to note that the complexity of the bass often depends on the lead instruments that will play over the top. Although some House music relies on a particularly funky bass groove, the overlaying

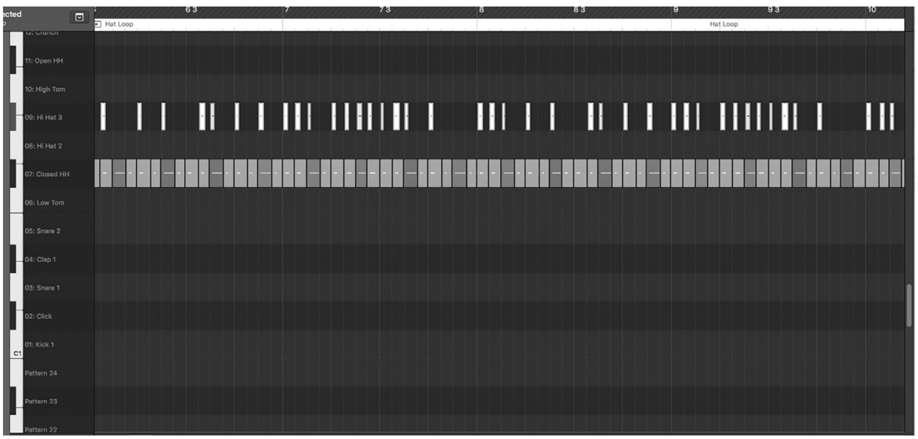

FIGURE 27.3
A pentatonic bass line with a leading tone (the leading tone is the highlighted note)

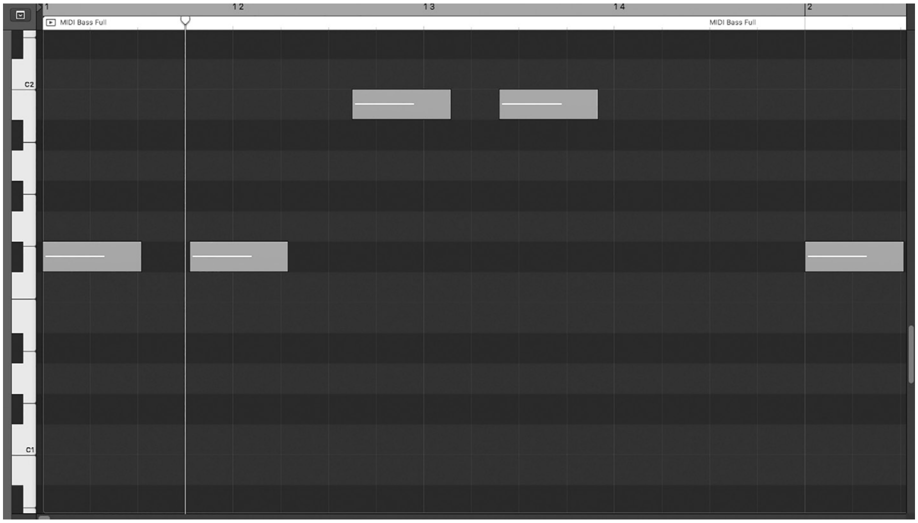

FIGURE 27.4
The bass used in 'Hot Chocolate.' The note positions are asynchronous to the beat to help create the tech groove

instruments are kept relatively simple by comparison. Alternatively, if the overlying lead elements are more complicated, then, in contrast, the bass line remains relatively simple. This is an important concept to grasp because if both the bass and lead instruments are melodically involved, it will result in a jumble of melodies that sounds too complex. So, when producing House, you need to decide as to whether the bass or the lead elements create the centerpiece of the music.

For example, the piano in the somewhat classic House track 'Love Story' (Laylo and Bushwacka) produces the centerpiece of the music while the bass acts as an underpinning and is kept simple. Conversely, tracks such as Kid Crème's 'Down and Under' and Stylophonic's 'Soul Reply' employ a funky bass that is complemented with simple melodies.

As with the rhythms, the timbre used for the basses in House can vary wildly from one track to the next. On the one hand, it can be from a real bass guitar, while, on the other, it could consist of nothing more than a low-frequency sine wave pulsing away in the background to anything in between. Like much of House music, this makes it incredibly difficult to pin down any particular timbre so what follows are a few synthesis tips to create the fundamental structure of most House basses. After you have this, experimentation is the key, and it's worth trying out new waveforms, envelope settings, and modulation options to create the bass you need.

The foundation of most synthetic House basses can be constructed with a sine wave and saw or pulse oscillator. The primary waveform (the sine) provides a broad body to the sound, while the secondary oscillator can either offer raspy (saw) or woody (square) overtones. If the sound is to be cutting and evident in the mix, then a saw is the best choice while, if you want to be more rounded and merely lay down a groove for a lead to sit over, then a square is the better option. Listening carefully to the sound they produce together, begin to detune the secondary oscillator against the first until the timbre exhibits the 'fatness' and harmonic content you want. Typically, this can be attained by detuning the waves from one another but as always let your ears decide what's best.

Nearly all bass timbres will start immediately on the key press, so set the amp's attack to zero and then follow this with a fast decay, no sustain and a short release. This produces a timbre that starts immediately and creates a little pluck before entering the release stage. If the sound is too short for the melody being played, increase the sustain and experiment with the release until it flows together to produce the rhythm you require.

If the bass appears to be 'running away,' try moving it a few ticks forward or back and play with the amp's attack stage. To add some movement and character to the sound, set the filter cut-off to low pass and set both the filter cut-off and resonance to midway, and then adjust the envelope to a fast attack and decay with a short release and no sustain.

This envelope is applied positively, but experiments with negative settings as this may produce the character you need. It's also worthwhile experimenting with convex and concave settings on the decay slope of the filter envelope as this will severely affect the nature of the sound. In most House tracks, the filter follows the pitch (opens as the pitch increases), so you could employ positive filter key follow.

This creates the basic timbre but experiment with LFOs, effects, and layering. Typical uses of an LFO will be to lightly modulate the pitch of the primary or secondary oscillator, the filter's cut-off or the pulse width if a pulse waveform is used.

In regards to effects, distortion is particularly useful and a commonly used effect in House alongside phasing/flanging. If these are applied, ensure that only frequencies above 150 Hz are processed. The bass should always sit in the center of the stereo spectrum so not only do both speakers share the energy but also that any panning applied during mixing is evident. With heavy flanging or phasing, the bass will be smeared across the stereo image, and the mix may lack any bottom end cohesion. Of course, if the bass is too thin or doesn't have enough body, then effects will not rescue it, so you could always layer it with a sine wave transposed down by a few octaves.

We have no expectations of how a synthesized bass should sound so you shouldn't be afraid of stacking up as many sounds as you need to build the required sound. EQ can always be used to remove harmonics from a bass that is too heavy, but it cannot be used to introduce harmonics that are not already present.

Alongside the synthetic basses, some House tracks will employ a real bass. These are sampled from other records or are occasionally taken from sample content rather than programmed in MIDI. However, it is possible to program a realistic bass provided that a reasonable sample engine is used (such as Spectrasonics' Omnisphere or a multisampled Kontakt Instrument).

The key to programming and emulating a real bass instrument is to take note of how they're played and then mimic this action with MIDI, and a series of control change commands. Most bass guitars use the first four strings of a standard guitar E – A – D – G that are tuned an octave lower. This means that the E is three octaves below middle C. Also they are monophonic, not polyphonic, so the only time notes will overlap is when the resonance of the previous string is still dying away as the next note is plucked.

This effect can be emulated by leaving the previous note playing for a few ticks while the next note in the sequence has started. The strings can either be plucked or struck, and the two techniques produce different results. If the string is plucked, the sound is much brighter and has a more extended resonance than if it were merely struck. For this, the velocity will need to be mapped to the filter cut-off so that higher values open the filter more.

Not all notes will be struck at the same velocity, however, and if the bassist is playing a fast rhythm, the following notes will commonly have less velocity since he has to move his hand and pluck the next string quickly.

Depending on the 'bassist,' they may also use a technique known as 'hammer on,' where they play a string and then hit a different pitch on the fret. This results in the pitch changing without actually being accompanied with another pluck of the string. To emulate this, you'll need to make use of pitch bend, so this will first require setting to a maximum bend limit of two semitones since guitars don't 'bend' any further than this.

Begin by programming two notes, for instance, an E0 follow by an A0 and leave the E0 playing underneath the successive A0. Just before where the second note occurs, program a pitch bend message to bend the tone up to A0. If this is programmed correctly, on playback, you'll notice that as the E0 ends the pitch will bend upwards to A0 simulating the effect.

Although this could be left as is, you could also drop in a CC11 message (expression) directly after the pitch bend, as this will reduce the overall volume of the second note so that it doesn't sound as if it's been plucked. In addition to this, it's also worth employing some fret noise and finger slides. Most good sample-based plug-in instruments will include fret noise that can be dropped in between the notes to emulate the bassist's fingers sliding along the fretboard.

Whether the producer chooses to employ real bass or a synthesized one, after it's written, it is commonly compressed against the drum loop. This will pump the bass to reproduce the classic bottom end groove of most House records.

You can do this by feeding both bass and drum loop (preferably with the hi-hats muted) into a compressor with a threshold so that each kick registers approximately –6 dB on the gain reduction meter. With a ratio of around 6:1 and a fast attack, the gain make-up is adjusted so that it's at the same volume level when the compressor is bypassed.

Finally, the release parameter is set to 200ms and gradually reduced while the music is playing. The shorter the release becomes, the more the kick will begin to pump the bass, becoming progressively heavier the more that it's shortened.

MELODIES, MOTIFS, AND CHORDS

The lead instruments melody will depend entirely on the type of House being produced. Funky bass lines will require less active melodies to sit over the top while less active basses will need more melodic elements. Unfortunately, since this genre has become so hopelessly fragmented, it's difficult to offer any guidelines apart from to listen to the latest House records to see where the current trend lies.

Synthetic House leads will often employ saw, triangle, and noise waveforms to produce a harmonically rich sound that will cut through the mix and can be filtered if required. Depending on how many are employed in the timbre, these can be detuned from one another to produce more complex exciting sounds.

If the timbre requires more body, then adding a sine or pulse wave will help to widen and give it more presence. To maintain the dynamic edge of the music, the amplifier's attack is predominantly set to zero so that it starts on key press but the decay, sustain, and release settings will depend entirely on what type of sound you require.

Generally speaking, it's unwise to use an extended release setting since this may blur the lead notes together and the music could lose its dynamic edge, but it is worth experimenting with the decay and sustain while the melody is playing to the synth to see the effect it has the rhythm.

As lead sounds need to remain attractive to the ear, employ LFOs or a filter envelope to augment the sound as it plays. A good starting point for the filter EG is to set the filter cut-off to low-pass and set both the filter cut-off and resonance to midway, and then adjust the envelope to a fast attack and decay with a short release and no sustain. Once this is set, experiment by applying it to the filter by positive and negative amounts. On top of this, LFOs configured to modulate the pitch, filter cut-off, resonance, and pulse width can also be used to add interest. Once the basic timbre is down, the key is, as always, to experiment.

Many House tracks will also employ chords and in particular genres such as *Deep House* and *Tech House* rely on them to create the atmosphere of the track. Here, it is the progression and style of chords that can often dictate the genre of the music. For example, in many of the uplifting forms of House, the chords will consist mostly of triads as these are considered light and breezy chords.

Here, if the music is written in a minor key (of which most uplifting style House tracks are), they will often employ a 'seventh' as the penultimate chord in the

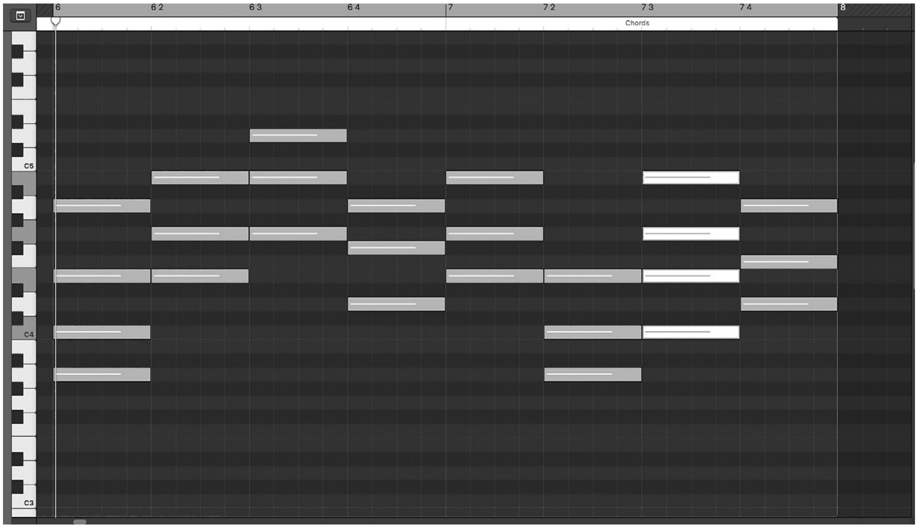

FIGURE 27.5
A typical 'epic' chord progression in an uplifting House genre

progression. This is because seventh chords add anticipation to the music and contribute to the epic feel of this style of House.

In the above example, when the chord progression reaches the 'seventh,' the track is lifted and receives its uplifting feel. Conversely, genres such as Tech House, Deep House, and Minimal will employ seventh chords throughout the progression and, rather than move up in pitch with each consecutive chord, it will move down to produce a more serious, almost depressing feel to the music.

There are many different styles of chord, and much of the synthesis choice is down to what you feel the track needs, but as a starting point, three saw waveforms are required with two detuned from one another. Apply a small amount of vibrato using an LFO to these two detuned saws with an amplifier envelope with a medium attack and release and a full sustain (there is no decay since sustain is fully open and it would have nothing to fall to).

The final saw is then pitched upwards as far as possible without becoming an individual sound (i.e., less than 20 Hz). If possible use two filters – one set as a low pass to remove the low-end frequencies from the two detuned saws and a high pass to remove some of the high-frequency content from the recently pitched saw.

This will produce a basic string timbre that can be further augmented with effects and processors such as flangers, phasers, delay, reverb, and compression. Extensive chorus effects, rotary speaker simulations, flangers, phasers, and reverb can all help to add a sense of movement to help fill out any holes in the mix.

For genres such as Tech House, Minimal and Deep House, you'll typically want to plane chords. This involves composing a triad or seventh chord in the root note of the music and then sampling this chord. The chord is played back using a single note on the piano keyboard, and, as you change notes, the chord changes in tonally interesting ways.

I used this technique on the House track. I sampled a seventh chord from an Emu Orbit (a popular hardware synthesizer in the 1990s) and then played that chord through Logic's EXs24 sampler.

To create further interest as the track developed, I sampled a second triad chord from Sylenth and used this as the response to the previous calling chord from Figure 30.5.

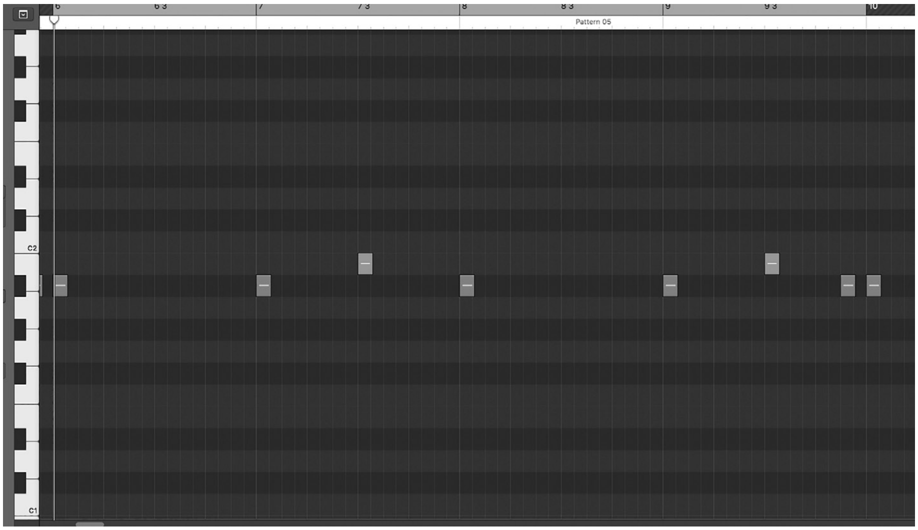

FIGURE 27.6
The Tech House chord arrangement from the track

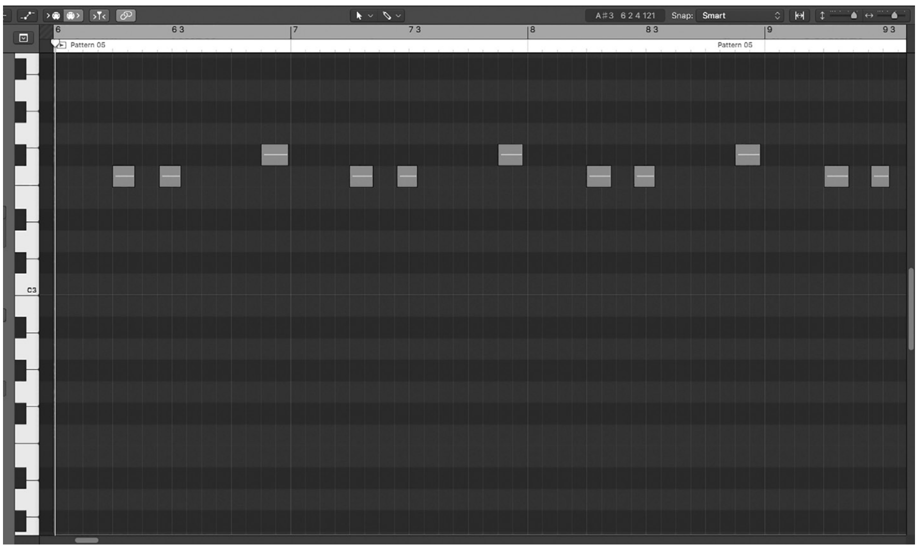

FIGURE 27.7
The Tech House chord 'response' from the track

In order to create a motif for the track (to make it more memorable), I asked 5Kulptor to contribute some vocals for the track. Finally, I simply added some vocal chants and placed them to occur asymmetrically in the bar. By doing so, it becomes difficult for the listener to determine the start and end of a bar, and so maintains repeated listening. Chants and the various sound effects have already been discussed in previous chapters, so I won't reiterate it all again here.

Ultimately, however, as with all the chapters in this section, the purpose here is to discuss some of the techniques and show how the theory and technology explained earlier in the book can be combined to produce a track. However, it should be viewed as offering a few basic starting ideas that *you* can evolve from.

There is no one definitive way to produce this genre and the best way to learn new techniques and production ethics is to actively listen to the current market leaders and experiment with the tools at your disposal. There are no right or wrong ways to approach any piece of music, and if it sounds right to you, then it usually is. New genres do not evolve from following step-by-step guides or from emulating peers; producers who experiment and push boundaries create them.

DMM resource: The companion files contain an excerpt of 'Hot Chocolate,' a House track released under the name Skreener, using the techniques described above.

Track 1 is the excerpt played through GM instruments.

Track 2 is the excerpt with full production.

'Hot Chocolate' © Skreener (R. Snoman) ft. 5Kulptor.

Lyrics by Tom Larkin.

CHAPTER 28

Techno

I find it amazing when people say that electronic music has no soul. You can't blame the computer. If there is no soul in music it's because nobody put it there.

Bjork

To the uninitiated, Techno can describe any form of upbeat electronic dance music. Although this was formerly the case, over the years, it evolved to become a genre in its own right. Initially, *Kraftwerk* coined the term to describe how they mixed electronic instruments and technology to produce 'pop' music. But, as more artists began to introduce technology into their productions, the real foundations of where Techno developed from is difficult to pinpoint.

To some, the roots of this genre can be traced back to as early as 1981 with the release of 'Shari Vari,' by *A Number of Names*, Donna Summers' (and *Giorgio Moroder)* 'I Feel Love,' and *Cybotron's* 'Techno City.' To others, it emerged in the mid-1980s when the 'Belleville Three' collaborated in Detroit. These three high school friends – Juan Atkins, Kevin Saunderson, and Derrick May – used to swap mixtapes with one another.

They religiously listened to the Midnight Funk Generation on WJLB-FM hosted by DJ Charles 'Electrifying Mojo' Johnson. It was a five-hour mix of electronic music from numerous artists including Kraftwerk, *Tangerine Dream*, and *George Clinton*. Motivated, they began to produce music using cheap second-hand synthesizers including the Roland TR909, TR808, and TB303.

The music they produced was labeled 'House music,' and both May and Saunderson admit to gaining much of their inspiration from the Chicago clubs (particularly the Warehouse and Frankie Knuckles). Indeed, Derrick May's 1987 hit 'Strings of Life' is still viewed by many as House music. Although to Derrick and many other aficionados, it was an early form of Detroit Techno.

It wasn't until 1988 that Techno became a genre in its own right. Neil Rushton produced a compilation album entitled *Techno – The New Dance Sound of Detroit* for Virgin records. Following this release, Techno no longer described any form of electronic music but instead described minimalist, mechanical House music. It was characterized by a mix of dark pounding rhythms with a soulful feel and a stripped back vibe.

This minimalist feel was not deliberate but a result of the limited technology available at the time. Because the Roland TB303, TR909, and TR808 were some of the few electronic instruments available to those without an extensive budget, most tracks were written using just these and recorded to two-track tape cassettes.

Similar to most genres of dance, Techno mutated as artists embraced these ideals. By 1992 and the rave generation Techno bore no relationship to the funky beats and rhythms of House music. Instead, it adopted drug-influenced hypnotic Tribal beats. As technology evolved and MIDI instruments, samplers, and digital audio manipulation techniques became more accessible, Techno began to grow increasingly complex.

While it still bore a resemblance to the stripped-down feel of Detroit Techno consisting solely of rhythms and bass, the rhythmic interplay became more complicated. More and more rhythms were laid atop one another, and the entire studio became one instrument with which to experiment.

Of course, Detroit Techno still exists today, but it has been vastly overshadowed by the Tribal beats of 'pure' Techno developed by numerous artists including *Adam Beyer*, *DJ Umek*, *Henrik B*, *Jeff Mills*, *Carl Craig*, *Kenny Larkin*, and *Richie Hawtin*. Each of these artists has injected his own style into the music while keeping with some of the original form set by his contemporaries.

MUSICAL ANALYSIS

Techno is electronic dance music in its most primitive form. It's chiefly formed around the cohesion and adaptation of various rhythms. Although synthetic sounds are also employed, they are often atonal, as it's the abundance of percussive elements that remain the centerpiece of the music. In fact, the genre defines itself on a collection of carefully programmed and manipulated textures rather than motifs.

The majority of Techno is produced by DJs for DJs. It's composed to allow him (or her) to seamlessly mix all the different compositions to provide one continuous rhythmical mix to last through the night. Techno follows the four to the floor time signature, and the tempo can range from 128BPM to 150BPM depending on the subgenre.

For this chapter, we will examine the general production ethics in the creation of a 'Mainroom' Techno track, but this approach equally applies to many genres of Techno. As music is an entirely individual, artistic endeavor, the purpose of this analysis is not to lay down a series of 'ground rules,' but, instead, will describe some of the general principles behind the production. In the end, it's up to the individual (i.e., you) to experiment and create music that suits your particular style.

In the late 1990s, Techno was different from every other genre of music because it didn't rely on programming and mixing in the 'conventional' manner. It was based around employing the entire studio as one interconnected tool. A hardware sequencer was used to trigger many drum rhythms contained in connected samplers and drum machines. Each of these rhythms ran through effects units that were manipulated produce new variations. These were layered with others or dropped in and out of the mix to create the final arrangement.

Today, the approach may have changed, but the theory remains the same. Numerous sampled rhythms are taken from sample content and dragged into the DAW. These are all different tempos from different genres of music but the DAW (typically, Ableton Live) auto time-compresses or stretches the loops to the same tempo. These loops are then edited in audio. Playing the loops together, the artist will remove individual sounds from each loop until he creates a new cohesive rhythm.

This is performed so that the different rhythmical elements all interact not only regarding syncopation but also in polyrhythm. Judicious application of EQ is used to enhance any interesting harmonic relationships created from this layering. The style is produced in any number of scales, but C major, A major, D minor, E minor, A minor, and C minor are all popular.

A Techno loop will often begin with the design of a kick drum. This can either be synthesized or created by layering numerous samples together. The kick is placed on every beat of the bar, along with snares or claps on the second and fourth beat to add expression to these two beats. To complement this basic pattern, closed hi-hats are commonly placed on every 16th division or variation of 16ths, while to introduce some basic syncopation open hi-hats are often employed and put on every 1/8th division of the bar.

The style of the kick timbre depends on its subgenre, but typically the kick is particularly deep with a harsh, hard transient. Most artists will build their kick using

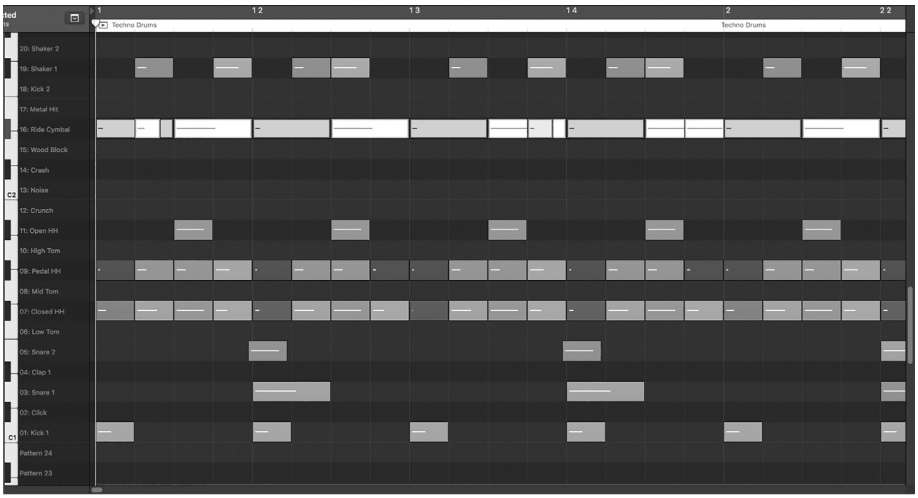

FIGURE 28.0
A Techno loop (used in the track)

a kick synthesizer employing a sine wave. Usually, a 60, 80, or 100 Hz sine wave produces the best starting point. The kick synth will apply a positive pitch EG to the sine wave, and it requires a fast amplitude attack with slower decay. This more gradual decay provides the thick meat of the kick to coincide with the deep bass.

Reverb is fundamental to the creation of any Techno kick but how it is applied depends on the style. A 'standard' Techno kick is often treated to a small amount of hall reverb with a long pre-delay to bypass the transient of the kick, but the tail is kept longer than most genres so that the reverb is evident. This is followed by a noise gate to ensure the reverb tail does not decay away as in a typical situation but is deliberately and evidently cut short.

A kick typical of the genre can be created by sending the kick drum to a bus channel featuring a hall reverb unit with full diffusion and a decay of five seconds. A compressor is placed directly after this reverb unit with a fast attack and an extended release (approx. 500ms) with a ratio of 5:1. The same bus channel is then used as the side-chain input for this compressor while the ratio is lowered until the reverb effect begins to pump.

The kick is then sent to a secondary bus that contains a room reverb unit with full diffusion but only a short decay of one second or less. A noise gate follows this reverb with a fast attack and release and the same second bus used as a side-chain for the gate. As the threshold of the gate is reduced, the reverb will begin to gate, and this should be set to personal preference. Finally, both reverbs are mixed in with the kick to produce the 'hiss kick' that appears in some Techno tracks.

The kick will often benefit from hard compression, but this must be applied cautiously so as not to destroy the bright, hard transient stage. Here, the attack is set so that it bypasses the initial transient but captures the decay stage. A ratio of 8:1 followed by reducing the threshold should provide the typical characteristics of a Techno kick.

CLAPS AND SNARES

Claps and snares are commonly part of the sampled loops that are dragged into the DAW. The artist may cut out every instrument from the drum loop leaving just the snare (or clap). These may then be layered with snares and claps from further edited loops.

Alternatively, the snare may be sourced from sample content. There is a multitude of Techno and Tech House-inspired samples now available on the market. The only problem is that other artists may easily spot these, so it is not uncommon to layer or modify them. A snare could be layered with another synthesized snare to produce a more exciting tone.

For this, you could employ a triangle wave for the first oscillator and noise for the second. The amplifier envelope generator requires a zero attack, sustain and release with the decay used to set the 'length' of the snare. If possible, employ a different amp EG for both the noise and triangle wave. By doing so, the triangle wave can be kept quite short and swift with a fast decay while the noise can be made to ring a little further by increasing its decay parameter.

If the snare exhibits too many low frequencies, then employ a high-pass, bandpass filter or notch filter depending on the type of sound you require. Notching out the middle frequencies will result in a cleaner sound that's common in this genre.

Claps can be created with a filter and amplifier envelope on a white noise oscillator. Both envelopes use a fast attack with no sustain or release and the decay will dictate the length of the clap. Finally, a sawtooth LFO modulating the filter frequency and pitch of the timbre with style the clap. Increasing or decreasing the LFO's frequency will then change the sonic character of the clap significantly.

Snares or claps are always treated to large amounts of reverb by inserting a reverb onto the channel. This is set to a large room or small hall and employs an extended pre-delay with a long delay and largest diffusion settings. After this, a noise gate removes the tail in an evident way. Compression follows the noise gate to compress the whole timbre heavily.

Pitch modulation can add further interest to these instruments, but they are often treated over a period of three or six bars rather than the period of a single bar. Each snare may also be processed to a different pitch than previous while all are remaining within 30 cents of the original.

FIGURE 28.1
Modulating the snares

The hi-hats are almost always a result of mixing various loops and taking the best samples from each. Some of the hats will be layered with other hat loops in the bar, while others are not. This creates a constantly changing landscape of different hats and rhythms occurring through the bar.

A favorite technique is to employ white noise and insert a compressor onto the same track that is side-chained to a secondary rhythmical channel. This secondary channel is usually another hi-hat set to no output so that it is used as nothing more than a rhythmical control for the compressor. Alternatively, some producers will merely side-chain the kick drum to the white noise to create lengthy sounding hats that pump with the kick.

Hats are almost always treated to delay and modulation. The hi-hats should be kept to short delay settings to prevent too much delay wash that will cloud the signal. A 1/16th or 1/8th setting with a very short delay time would be enough.

The open hats are often treated to a low-pass filter that is modulated via an offset sine wave or a sample and hold waveform. This is applied over a period of three bars while further cyclic modulation will be applied to the closed hats over a different period of bars.

This prevents the listeners from explicitly identifying a fixed cyclic pattern in the different rhythms. Typically, odd numbered bars are chosen for the cyclic modulations since this works against the standard structural downbeat of dance music and result in a cross-syncopation of modulation.

ANCILLARY INSTRUMENTS

Once the primary loop is formed, further percussive instruments are layered onto the rhythm. The initial approach here is to 'fill' the gaps between the kick, snare, and open hi-hat with any number of different synthesized percussive elements. These again may be the result of sliced drum loops or may be single percussive hits sourced from sample content.

The purpose is to create a syncopated groove employing techniques such as hemiola, polymeter, and call and response to create binary phrasing. Hemiola consists of creating an unequal subdivision of beats within the bar and placing a hi-hat or percussive rhythmical pattern onto these unequal subdivisions.

This effect is further augmented with the use of polymeter on further hi-hats, snares, or percussive elements. A typical application of polymeter here is to employ a 5/4 on a rhythmic hi-hat and 6/4 and 7/4 on additional rhythmical elements such as congas or synthesized percussive hits.

As Techno relies heavily on its percussive elements, call and response form a fundamental cornerstone of the music. Many of the percussive instruments will employ some form of binary phrasing to keep the listener interested. This is not applied singularly, however, and most of the ancillary instrumentation will create a call and response motion over a different period of bars. One percussive element may make the call over three bars to be answered in the fourth while another percussive part may make a call over half a bar and another makes a call over six bars.

A second important principle in the design of Techno is to maintain short transient hits on all of the percussive elements. It is not unusual for a Techno arrangement to feature a noise gate on every channel. The gates on each are then carefully adjusted to ensure that each percussive hits end before the next begins. This ensures that all of the percussion hits are kept short and sharp and helps to maintain the rhythmical yet minimal flow of the music.

BASS

The bass is fundamental to modern Techno. It not only solidifies the drum rhythms but also creates the requisite groove. In many examples of the genre, the bass is simple so as not to detract from the drum rhythms. Indeed, the bass commonly consists of little more than a series of quarter or half notes with little movement in pitch. Techno does not employ exotic bass lines. In numerous examples of the genre, it consists of little more than two quarter notes, played at two different pitches that repeat over and over again. This produces a ½ bar loop in the bass that plays off against the one-bar loop of the drum rhythms creating a form of polyrhythm. It is rare for Techno to employ anything more complicated, and it is unusual for a bass line to go further than a ½ bar.

Artists will sometimes employ the Roland TR808 kick drum as a bass. You can use the original instrument (rare and expensive), the hardware re-release (meh!), or a plug-in instrument. The kick will first need tuning to the scale of your music using the tone control. The best way to do this is to play an initialized tone from

FIGURE 28.2
A typical Techno bass line

a synthesizer at the key of your track and match it by ear while reducing the tone control on the TR808.

Once the kick is in key, you need to lengthen the decay parameter. As you do so, you'll notice the weight of the kick (and bass) increase. Finally, employ a compressor with a slow attack, slow release, a low threshold, and high ratio. This will allow the transient to pass unmolested but capture the rear of the kick so that it can be increased in gain. This will even out the dynamics of the sound to create the bass. If pitch is required, then the note can be sampled and played at various pitches in the sampler.

Another classic Techno bass sound is the nervous sine bass. This is nothing more than a sine waveform playing the bass notes polyphonically at a low frequency. One note may play at C1 while the next note plays at D1. This creates a phase interrelationship between the low frequencies and produces a slight warble in the bass tone.

For those who are a little more adventurous, four saw waves, or a couple of saws and a sine wave to add some bottom end can produce a powerful Techno bass. If a sine wave is used, detune this by an octave below the other oscillators and then proceed to detune each saw from one another by differing amounts.

The amp envelope for all the waveforms is commonly set to a fast attack with a medium to long decay and no sustain or release. If a pitch envelope is available in the synth, you could positively or negatively modulate the pitch of the oscillators to add some small amounts of movement. Once this basic timbre is down, you can experiment with the attack, decay, and release of the amps EG to help the bass sit comfortably in with the drum loop. In fact, it's essential to

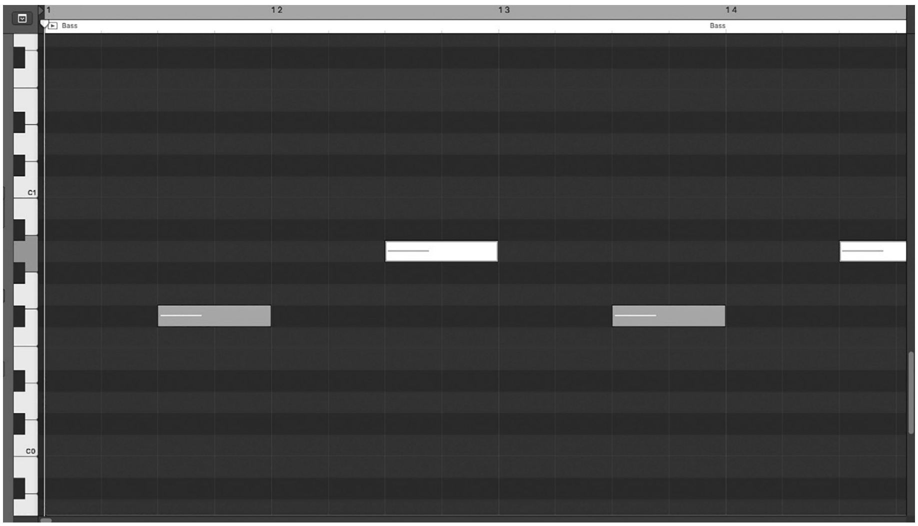

FIGURE 28.3
The bass used in the track (the kick fills in the silence)

accomplish this rhythmic and tonal interaction between the drum loop and the bass before moving on.

It's also worth experimenting with effects to make it more vibrant sounding. It is not unusual to apply reverb to the bass. This is implemented on a bus and followed with a compressor, noise gate, and EQ. The compressor is used to reduce the dynamic range of the reverb so that it remains constant while the noise gate removes the tail as soon as the bass stops. The EQ can then be used to sculpt the frequencies in the reverb to create the final sound.

Small amounts of controlled distortion can also help to pull the bass out of the mix or give it a much stronger presence. Similarly, a conservatively applied delay effect can be used to create more complex sounding rhythms.

CHORDS, STABS, AND SOUND EFFECTS

One final aspect of Techno is the addition of sound effects, chorded stabs, and vocal shouts. The sound effects are generated by whatever means necessary, from sampling and contorting any available sound and treating it with effects, processors, and EQ. Typical effects here are to time-stretch timbres well beyond the normal range and into obscurity, to then slice, rearrange and further process them with effects and processors.

Many Techno producers will own a field recorder. This is a small digital recording device that they'll use to record real-world sounds. These could be trains passing by, traffic, even bathroom extractor fans. These sounds are imported into the DAW, edited, and manipulated to produce interesting background textures for the music.

Another common effect is to time-stretch pads massively to produce a drawn-out almost digitally corrupt timbre and then apply heavy reverb followed by a gate. Creativity on behalf of the producer is the key here, and it's not unusual for a producer to spend a good few weeks merely experimenting with sampled sources and sounds to create unique sounds.

Chorded stabs can be created in this same way, although rather than time-stretching a pad, it is time-compressed. Taking a pad sample that is usually four bars and time-compressing it into a 1/16th or 1/8th note is a conventional technique for producing Techno stabs. Similarly, sampling chords such as guitars or horns and time compressing them into shorter notes is a favorite technique. For my Techno track, I simply used a 'donk' bass and placed this asymmetrically on the bar. I've already covered how to create this timbre in an earlier chapter.

To complement the rhythm, you can place further chords into the arrangement. For the track featured in the resource pack, I played a triad chord through an organ preset on my Korg M1 (hardware) and sampled the results into a Yamaha SU700 sampler. I used the SU700 because this can sample in 12-bit, which adds a pleasant distortion to sounds, making it more suited to Techno production.

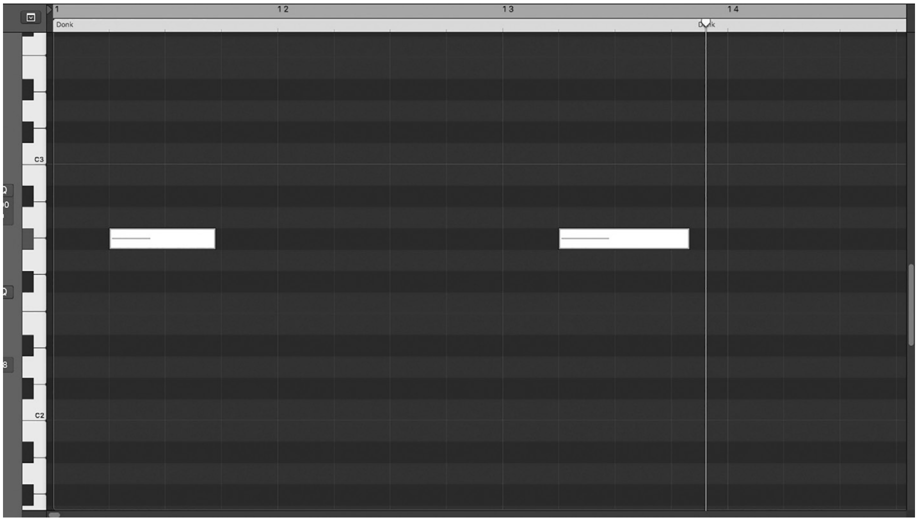

FIGURE 28.4
Asymmetrical positioning of the donk sound

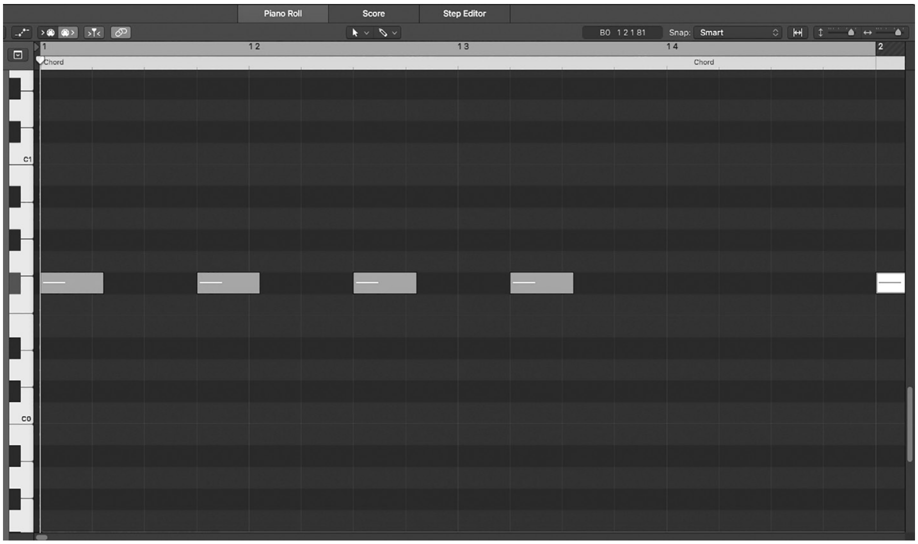

FIGURE 28.5
Positioning of the M1 organ sound

The vocals in this genre usually consist of little more than a very short sample. The verse and chorus structure is avoided, and in many cases, only little phrases are used. These can be sampled from old records, because they're so short they're rarely recognized so you don't always need permission. In some Techno, vocal samples are taken from the old 'speak and spell' machines of the early 80s. This particular machine isn't a requirement (with its increased use in dance music, second-hand prices of these units have risen). The same effect can be obtained from most vocoders, so long as the carrier consists of a saw wave and the modulator is robotically spoken.

Ultimately, the fundamental technique for Techno is experimentation with effects and processing. Heavy compression, bit-crushers, distortion, saturation, phasers, delay, reverb, and automation of all these parameters over any number of bars provide the foundation. Techno should be considered a DJ's tool rather than a record in its own right and the music should be produced with this in mind.

There is no one definitive way to produce this genre and the best way to learn new techniques and production ethics is to actively listen to the current market leaders and experiment with the tools at your disposal. There are no right or wrong ways to approach any piece of music, and if it sounds right then, it usually is. New genres do not evolve from following step-by-step guides or from emulating peers; producers who experiment and push boundaries create them.

DMM resource: The companion files contain an excerpt of 'Tempest,' a Techno track released under the name Ascii, using the techniques described above.

Track 1 is the excerpt played through GM instruments.

Track 2 is the excerpt with full production.

'Tempest' © Ascii (R. Snoman).

CHAPTER 29

Uplifting Trance

Good music doesn't have a shelf life.

Anonymous

Trance is an ambiguous genre because, like House, it manifests itself in many forms. It can be Goa, Psychedelic, Uplifting, Euphoric, Acid or Progressive. It is, however, the only form of dance music constructed around what you might consider a melody. Whereas most genres of electronic dance music employ motifs, many types of Trance will feature energetic melodies.

This genre of music has its roots embedded in Germany. During the 1990s a joint project between DJ Dag and Jam El Mar resulted in Dance2Trance. The first track, entitled 'We Came in Peace,' is considered by many to be the first recording of the genre. Although by today's standards it was mainly raw, consisting solely of repetitive patterns (as Techno is today), it laid the essential foundations for the genre. Its purpose was to place clubbers in a trance-like state.

The ideas behind this are nothing new: tribal shamans use natural hallucinogenic herbs and rhythms pounded out on log drums to induce the tribe's people into trance-like states. The only difference with Dance2Trance was that the pharmaceuticals were man made and machines rather than skin produced the pounding rhythms stretched across a log. Indeed, to many Trance aficionados, the purpose of placing clubbers into a trance state is believed to have formed the basis of both Goa and Psychedelic Trance.

Although both these genres are still produced to this day, it was the growing use of 3,4-**M**ethylene**D**ioxy-N-**M**ethyl**A**mphetamine (MDMA, or ecstasy) that resulted in new forms of Trance. Because MDMA stimulates serotonin levels in the brain, it's challenging to place clubbers into states of trance with Tribal rhythms. So instead, exotic melodies took precedence. It was no longer created to induce a trance-like state but imitate (and stimulate) the highs and lows of

MDMA. This basic premise remains today. Chemicals still play a role in causing a state, albeit a euphoric one, and the name Euphoric Trance was given to such tracks.

This form of music, employing long breakdowns and substantial melodic reprises, became a popularized style that dominates many clubs, so much so, many call it *Mainroom* Trance. As this *still* remains as the most popular form of the genre, it will be the focus of the chapter.

MUSICAL ANALYSIS

The best way to begin composing any genre is to source the most popular tracks and break them down into their essential elements. We can examine the similarities between each and determine what it is that differentiates it from other musical styles.

In fact, all music that can be placed into a genre-specific category will share similarities regarding the arrangement, groove, and sonic elements.

In the case of Mainroom Trance, it exhibits an up tempo, uplifting feel that's accessible to most mainstream clubbers. It's best illustrated as consisting of melodic leads, often with vocal hooks dropped over a drum pattern and *running* bass line. The drums commonly feature snare rolls for builds to the reprise and breakdowns, alongside small motifs or chord progressions that work around the main melody. Uplifting Trance can be written in any scale, but a significant proportion of trance tracks tend to be written in C major, D major, F major, A# major, G# major, A minor, or F minor. All employ a 4/4 time signature, and the tempo typically runs from 136 to 140BPM, with 138BPM considered the standard.

THE RHYTHM SECTION

The rhythm section is invariably kept simple and relies on the *four to the floor* pattern. A kick sits on all four beats of the bar with snares or claps resting on the second and fourth beat. This pattern is augmented with a 1/8th offbeat open hi-hat and a couple of closed hi-hat patterns and drum loops.

Trance kicks are typically tight rather than 'boomy' and exhibit a bright transient stage to help them cut through. The kicks, as with all dance music, can be created through sample layering, taken from sample content, or more commonly programmed in a kick synthesizer.

A good starting point for the main kick is to employ a sine wave between E2 and G2. It requires an immediate attack stage to maintain the transient and a short release. The genre features an offbeat bass so the kicks release can be adjusted to end before the offbeat bass occurs. An exponential fall in the amplitude and pitch is preferred over a logarithmic because the latter tends to create a kick with more bounce.

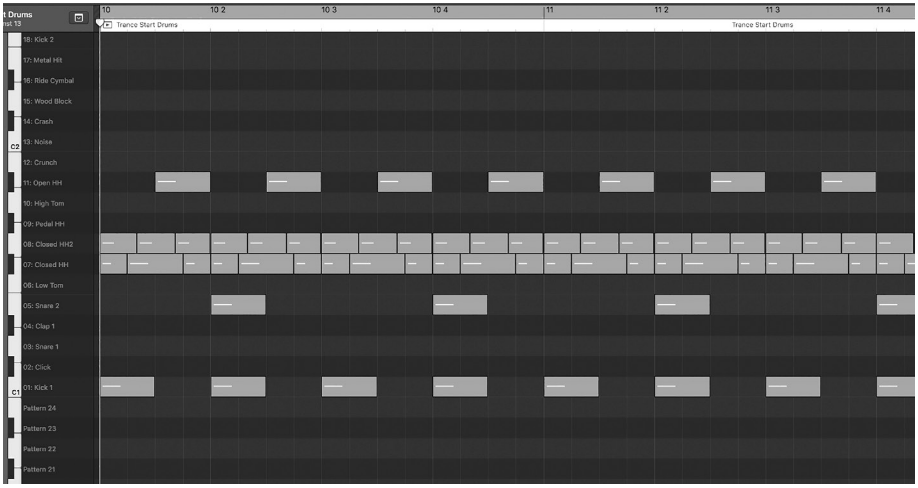

FIGURE 29.0
Typical Trance drum loop

Once the basis of the kick is down, it should be played with the bass (discussed later). If the two don't sit properly together, use a multi-band graphic EQ to boost and cut parts of the kick to help it gel with the running bass. The kick is sometimes treated to small amounts of reverb. Here, you can use a room reverb with a long pre-delay so that the transient passes unmolested. A good starting point is a pre-delay of 90ms with a tail of approximately 40ms.

Compression is typical because it plays a significant role in achieving the right sound for the genre. Here, the attack of the compressor should be configured, so that is skips the initial transient but captures the body and tail. The ratio is set at approximately 3:1 to 5:1 and, while listening to the kick, reduce the threshold until the required sound is achieved. A side effect of this compression may be a reduction of the transient, but if this occurs, a hi-hat layered over the kick's transient can help. Some artists will apply small amounts of controlled distortion to the kick, following this distortion with EQ so that it can be further sculpted.

Trance will employ either snares or claps to augment the second and fourth beats. As with the kicks these can be produced through sample layering, from sample content, they can be synthesized or more usually a mix of all. This is because Trance claps and snares often feature an elongated tail section that slurs. It's possible to create this sound by layering claps (or snares) with some white or pink noise from a synth. The noise can be bounced with the clap and then reimported into the DAW to be edited with the fadeout tools to create the sound you need.

Typically, snares or claps are treated to a room style reverb with an extended release although allowing it to tail off gently. The reverb release should remain reasonably short, however, so as not to wash over the loop entirely. Some artists will employ reverse reverb to some of the snare/clap hits but this is typically limited to either one hit per bar (commonly the second beat) or will only occur at structural downbeats.

If samples are used for the snares or claps, then either velocity modulation or filter modulation should be used to produce some change between the beats. In the example track, I used *Sound Toys' FilterFreak* on the snare with an offset sine wave modulation to control a low pass filters cut-off. The modulation was applied, so that first snare of the bar passes through a low-pass filter unmolested but the second (occurring on beat four) is subjected to a very light low-pass filter. This removes a small portion of the higher harmonics of the timbre and introduces some textural variance into the two beats.

The hi-hats are often a mix of both composition and sampled loops. In many Trance tracks, the open hats are sourced from the Roland TR909 because of the distinctive character. These can be samples from the machine, recorded from the TR09 rerelease, from a software emulation or the original unit. For the closed hi-hats, artists will import some closed hat loops and drum loops and extract the parts of the loops that catch their ears. This is performed with the editing tools available in the DAW.

For example, in a drum or percussive loop, the artist may take just a few hat or tom samples and remove the rest. While, in another loop, they may take further hats or snares to layer over the previous to create interesting rhythms. A conventional technique is to take a complete 4/4 drum loop and – using an EQ – high pass all frequencies below 1 kHz. This effectively removes the kick and most of the snare but leaves the upper energy of the loop to fill the higher registers and create a groove.

This approach can be augmented with some synthesis to create interesting textures. A good example here is to employ ring modulation on a high-pitched triangle wave with a lower pitched triangle. This produces a high-frequency noise that can be modified with an amplifier envelope set to a zero attack, sustain, and release with a short to medium decay. If there isn't enough noise present, it can be augmented with a white noise waveform using the same envelope. Once you have this basic timbre, shortening the decay creates a closed hi-hat style sound. It's also worth experimenting with changing the decay slope to linear or exponential.

Like the snare/clap, the open and closed hi-hats can be treated to a similar form of cyclic filter modulation. Typically, the open hats will be treated to a low-pass filter that is modulated via an offset sine wave or via a sample and hold waveform. When this effect is applied *very lightly* over the length of two bars, very slight harmonic changes occur over a period of two bars, and since the modulation source is offset or random, it prevents it from appearing as a cyclic repeated motion.

Occasional flourishes in the drums at the end of every fourth or eighth bar can help to add some interest to the rhythms. Here, hemiola on closed hats at the end of every fourth bar can add some variation or the snare/clap may be placed a 1/16th before the kick occurs. Another standard technique is to employ a noise gate on the closed hi-hat channel and then trigger the gate with a rhythmical programmed pattern from another channel via the side-chain function. This pattern is changed every so often on a structural downbeat to inject more interest into the drum rhythms.

FIGURE 29.1
Modulating the drum rhythm

Finally, swing quantize is almost always applied to the snares and open hi-hats. This is implemented gently so as not to interrupt the timing too heavily and applications of between 51 and 53% on a 1/16th grid will be enough to maintain some rhythmical interest. After this, the loop will often be treated to some parallel compression to help the loop gel and introduce more energy.

BASS GROOVE

The bass rhythm in Trance almost always consists of the *running* bass. This is the term used to describe a fast, surging motion delivered from the bass line. This consists of three or four bass lines playing off against one another.

The process starts with a low-frequency sub bass that sits on every 1/8th offbeat (underneath the open hi-hats). The purpose of this positioning it to ensure that the sub-frequencies of the bass don't collide with the sub-frequencies of the bass. Trance is commonly based on an eight-bar loop with the pitch of the bass changing every two bars, following the chord change.

The standard choice for the creation of sub bass is a triangle and sine wave from an analog subtractive synthesizer. The sine wave produces the depth of the sound while the triangle adds the upper harmonics to allow it to cut through.

The amplitude envelope should feature a fast attack, medium decay, sustain, and a quick release. If employing a square or saw, a 12 dB low-pass filter should be used to remove all the higher harmonics, leaving only the lower frequencies intact. The sub rarely requires a filter 'pluck,' because its sole purpose is to pulse along as part of the rhythm and create the groove. Figure 29.2 shows a typical Trance sub bass arrangement in Logic Pro's Piano Roll editor. Note how each transient of the bass not only remains off the beat so as not to clash with the kick.

After the sub bass, there is the mid-bass. This consists of a series of 1/16th notes playing in succession with a single 1/16th rest at the beginning of each beat. This rest creates space for the kick. These will follow the same notes as the sub bass. In the example track, the sub bass plays A for two bars before moving to F for two bars, then D for two bars, and finally E for 2 bars. The mid-bass follows this same motion in the bass.

For this instrument, the tonality should be higher than the sub bass. For this, the mid-bass can be played an octave higher than the sub bass or – if for sub bass the oscillators were dropped by an octave in the synth – you can leave the octave of the oscillators at their standard settings. The choice of waveform depends on the sound you want, but this mid-bass should feature a pluck. Consequently, you will need to employ waveforms with a high harmonic content and a popular choice are mixes of saws or pulse waves.

Employ detuning of the waveforms alongside unison to create a more significant sound and use an amplitude envelope with a fast attack, medium decay, no sustain and a short release. The release should be set so that the patch stops before the next note begins. To create the pluck, set the cut-off to quarter ways open and then modulate the filter with an envelope set to a zero attack, quarter decay, no sustain,

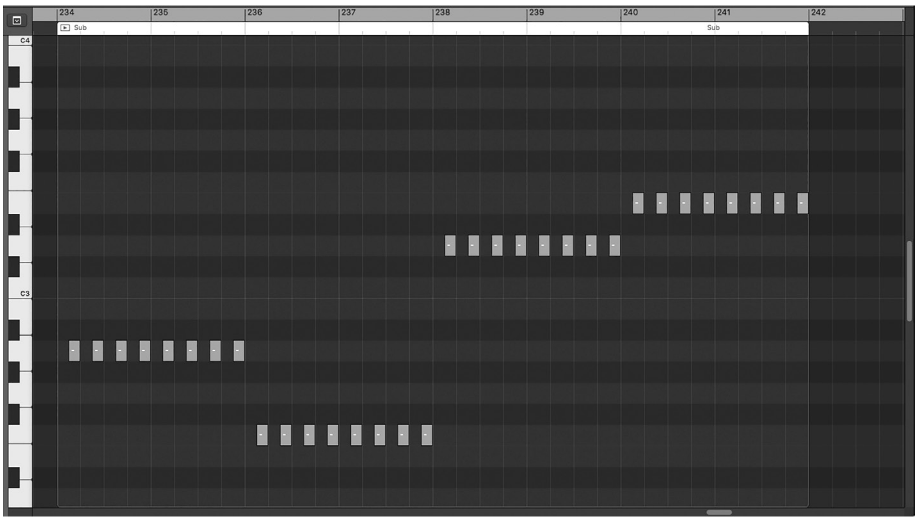

FIGURE 29.2
The Trance sub bass

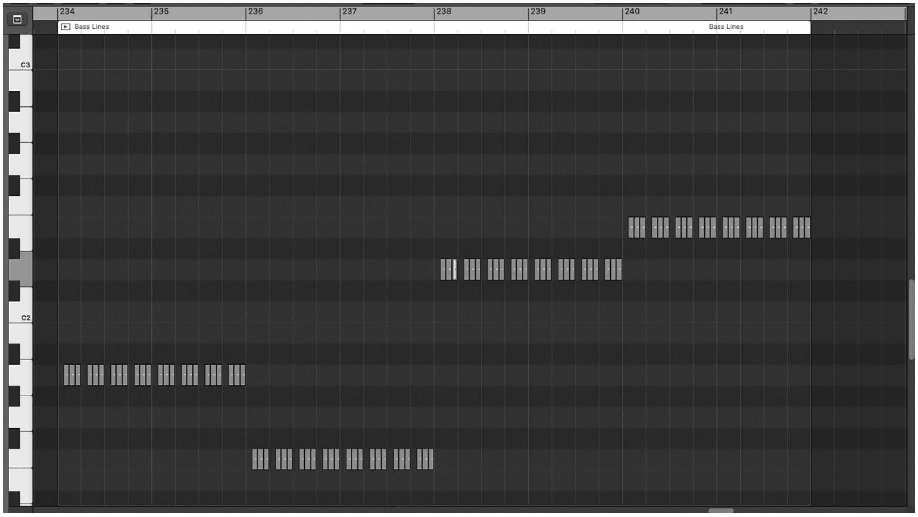

FIGURE 29.3
The Trance mid-bass (1)

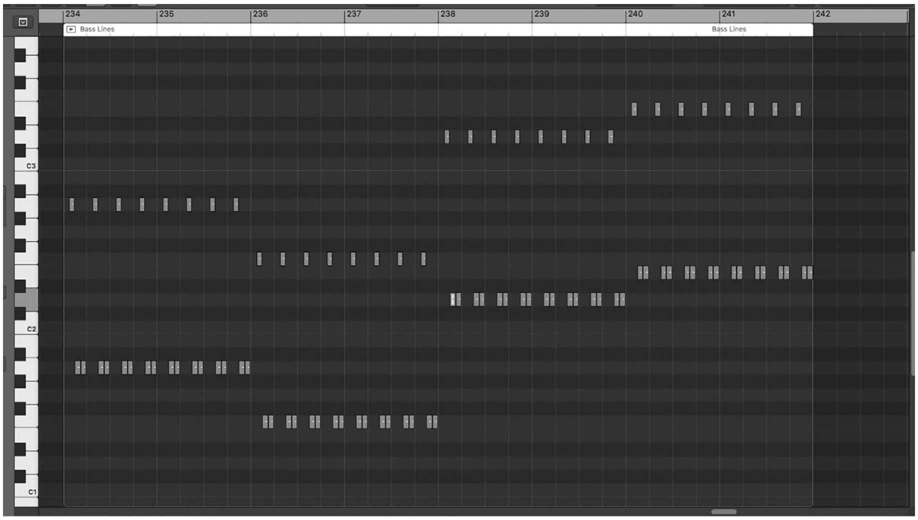

FIGURE 29.4
The Trance mid-bass (2)

and no release. Adjust the amount the envelope modifies the filter, the cut-off, and the modifier's decay until it produces a pleasant filter pluck.

After the mid-bass, we can produce the next bass. This is again a form of mid-bass but different in both composition and synthesis. For the structure, it should follow the previous mid-bass rhythmically, but feature a change in pitch.

The previous mid-bass was composed as a series of 1/16th notes with a rest on every beat. The second mid-bass follows that same composition but rather than remain at one note for the two bars, it will octave shift on a note. The note to octave shift is down to your choice, but for the example track, I moved the first note after each beat up by an octave. This is shown in Figure 29.4.

The approach for synthesis is the same as the previous mid-bass. It should employ a pluck, but it should also exhibit a different sonic character than the last bass. So, if you used a couple of saws on the previous bass then for this, it may be worth employing a saw and a pulse, or a couple of pulse waves. Once you have this bass down, you can then introduce another mid-bass. As before the rhythm remains the same, but the pitch is different. For the third mid-bass in the example track, I moved the middle notes up by an octave as shown in Figure 29.5.

More basses can be added to the above, and, in many Trance tracks, it's not unusual for it to feature four or five bass lines all working off one another. Once they are all sitting together, some may be treated to delay, distortion and various other effects.

If employing a delay, it should be placed on a bus and followed with a compressor set to a fast attack and release with a high ratio and low threshold. The compressor is then fed with the original bass signal into its side chain input, and the basses are *sent* to this delay bus. This way, every time an original bass hit occurs, it triggers the compressor through its side chain input. As the compressor is triggered, it restricts the dynamics and lowers the volume of the delayed bass. However, during the silence between the original bass hits, no compression applied and therefore the delay occurs at full volume.

Distortion and compression are also often used on basses. Here, small amounts of controlled distortion can help to pull the bass out of the mix or give it a stronger presence while side-chain pumping compression is applied to both basses. To accomplish this, you can duplicate the original kick track but set this one to

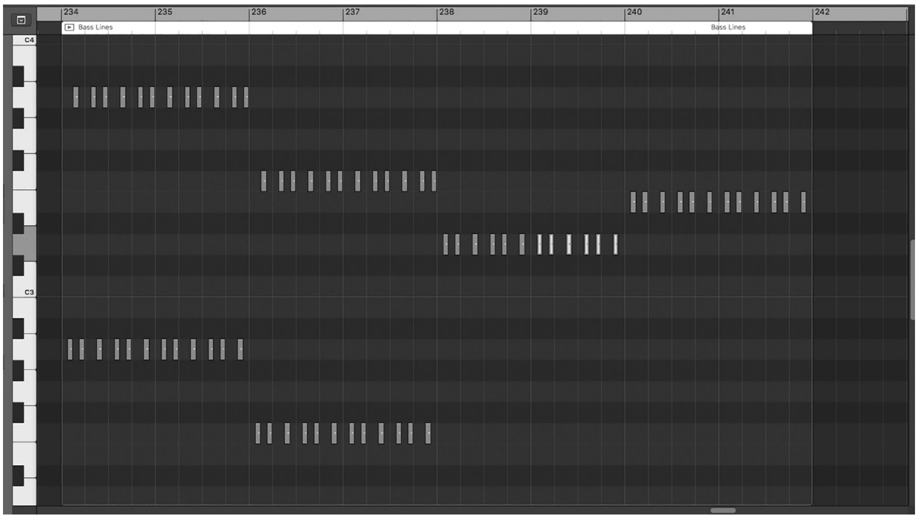

FIGURE 29.5
Another mid-bass

no output in the mixer so that it cannot be heard. This track can then be used as a side-chain input channel on the basses.

All the mid-basses should be grouped into a group channel, and then a single compressor can be applied to that group channel. The newly created kick channel is side-chained into the compressor and adjusted so that the basses pump rhythmically with the kick. By creating a secondary kick track to accomplish this side-chain rather than using the original kick channel, if the original audible kick is removed from the track during a drop, the 'pulsing' in any side-chained instruments will still be present.

For compression, start by setting the ratio to 9:1, with an attack of 5ms and a medium release of 200ms. Slowly decrease the threshold until every kick registers on the gain reduction meter by at least 3 dB. To avoid the volume anomaly (i.e., louder invariably sounds better!) adjust the make-up gain so that the loop is at the same volume as when the compressor is bypassed and then start experimenting with the release settings. By lengthening the release, the kicks will begin to pump the basses, and the pumping will become progressively heavier.

UPLIFTING TRANCE MELODIES

The lead melody is the essential component to Trance music. Although time and care must be taken producing every compositional part, with Uplifting Trance, the theme is the selling point of the music. It's also the most difficult to build as a great lead is a mix of both melody and synthesis. Scrutiny of the current scene and the market leaders is essential for acquiring the right feel for the dance floor.

Unfortunately, concerning composition, Trance leads follow very few 'rules,' but there are some basic guidelines that we can apply. The most popular method is to extract the lead melody from the chord structure of the record. The melody is commonly a series of 1/16th notes bouncing between the different notes that comprise the chord. Of course, this means you need a chord progression. How to compose these has already been covered in detail in a previous chapter. In the example track, the chord was composed of the notes A – E – G/E – G – A/F – A – B/D – F – A and therefore, the melody jumps between these notes as shown in Figure 29.6.

An alternative approach to producing a trance melody is to employ an arpeggiator. Here, a chord is held down, and you can tweak the arpeggiator's pattern until inspiration strikes, and you can develop on it further. Or you could insert a noise gate onto the chords channel and then program a rhythmical hi-hat pattern. This can be used to control the noise gate to produce a rhythmical lead. This gated effect, once suitable, can be applied to the notes (i.e., cut them up to create the effect) so that each note retriggers the synth. You could then offset the top notes of a chord against the lower ones to produce an alternating note pattern.

FIGURE 29.6
The Trance melody

MELODIC TIMBRES

Just as the melody for the lead is essential, so is the timbre. The lead is the most prominent part of a track and should sit in the midrange/upper midrange. It should be rich in harmonics that occupy this area and should also exhibit modulation to remain interesting.

The secret (*if there is one*) to the lead lies with saws, unison, effects, and layering. Begin with three saw oscillators, or if the synth has them available, stacked saw oscillators (aka multi-saw oscillators), and detune them from one another by as much as possible but not too far that the oscillators separate from one another. Once detuned, unison should be introduced until the sound becomes very thick. The amplifier envelope should be set to a fast attack with a rapid decay, medium release, and average sustain value while a low-pass filter is modulated with a second envelope with the same settings as the amplifier. The low-pass filter should be completely open, allowing all frequencies through.

This alone is rarely enough, and it should be layered with further synthesized leads. Like the basses, the leads can be programmed in the same way but employ pulse waves in place of saws and detune them differently.

Once a few leads have been synthesized, the amplifier attack, decay, and release can be modified to create an exciting sound. Lengthening the attack of one layer and keeping the other two short will produce an interesting textural variation as the third layers enters the scene. Similarly, different decay and release times will add further interest to the timbre. Once the layers are together, they can be grouped into a single channel and any processing or effects applied to that one channel (and all leads).

EQ is a popular process for leads: a 6 dB boost at 650 Hz with a high shelf boost of 10 dB at 5 kHz will provide the characteristic bright sound. This EQ is commonly followed by a distortion unit and chorus and then a secondary EQ unit. The settings for these are dependent on the style of timbre required, so experimentation with all three is the key. The distortion is usually applied gently to add further harmonic characteristics to the higher frequencies while chorus can be used to thicken the sound further, and the EQ can be used to remove any problematic frequencies.

A large hall setting can be applied to the lead with a pre-delay set so that the initial reflections bypass the transient. The larger the reverberant field, the more 'epic' the lead will become, but the reverb should be followed with a noise gate, so that tail of the reverb is cut the moment the timbre ends to prevent reverb from washing over the mix. Finally, the lead is processed with a short delay. Both 1/16th and 1/8th settings are typical, but experimentation will yield the best results. Although not always necessary, parallel compression can be employed on the lead to bring up its presence in the mix.

If vocals are employed on the track, then there may be little need for any effects, as the lead should sit under them. However, if you want to keep a full harmonically rich sound *and* vocals, it's wise to employ a compressor on the lead timbre and feed the vocals into a side-chain so that the lead drops when the vocals are present.

MOTIFS

With both main melody and the all essential groove down, the final stage is to add motifs. These countermelodies are the little ad lib riffs best referred to as the icing used to decorate the musical cake. They play an essential role adding much needed variation to what otherwise would be a repetitive track.

There are various techniques employed to create motifs, but one of the quickest ways is to make a copy of the MIDI lead and 'simplify' it by removing notes to form a pattern. This pattern can be offset from the central lead by shifting them forward or later by a 1/16th, or it can occur at the same time as the lead, but the attack and release of the timbre are lengthened.

Above all, using any of these techniques, it's sensible to avoid getting too carried away. Many dance tracks just use a single pitch of notes, playing every eight, sixteenth or quarter, or composed to interweave with the bass rather than the lead. This is because not only do simple motifs have a dramatic effect on the music than complex arpeggios, but you don't want to detract too much from the primary melodic element.

Additionally, the timbre used for any motif should be different from the lead melody, and it's at this point that you'll need to consider the frequencies that are used in the mix thus far. As touched on, in many Trance tracks the lead is harmonically rich, and this reduces the available frequencies for a motif, so it's common practice to use a low-pass filter cut-off to minimize the harmonic content while the lead is playing. And then set this filter to open wider while there is no lead playing.

Ultimately as with all the chapters in this part of the book, the purpose here is to reveal some of the techniques and show how the theory and technology discussed earlier in this book combine to produce a track. There is no definitive way to create this (or any) genre, and the best way to learn new techniques and production ethics is to actively listen to the current market leaders and experiment with the tools at your disposal. There are no right or wrong ways to approach music, and if it sounds right, then it usually is.

New genres do not evolve from following step-by-step guides or from emulating peers; producers who experiment and push boundaries create them.

DMM resource: The companion files contain two tracks of a Trance track, created using the techniques described above.

Track 1 is the excerpt played through GM instruments.

Track 2 is the excerpt with full production.

CHAPTER 30

Dubstep

Dubstep didn't invent bass, it just zoned in on it. Bass, to varying depths, is the foundation to most dance music.

Kode9

Although Dubstep is considered a relatively new genre, its history goes back further than many believe, but like most genres of dance music, its history is also amorphous. That said, it can mostly be traced back to 2-Step in 1998.

A development from UK Garage, 2-Step consisted of syncopated rhythms that veered away from the generic four to the floor of dance music. However, it was the B sides of these 2-Step records that formed the beginnings of Dubstep. They contained more experimental remixes of 2-Step, mixing the deep bass philosophy of 2-Step with the funky syncopated rhythmic undertones of Drum & Bass and Grime.

DJ Hatcha, an employee at Big Apple Records in Croydon, South London, is credited as being one of the first DJs to play these B sides to clubbers. During his sets at *Forward*, in the Velvet Rooms in Soho, London, they were used as a showcase for experimental 2-Step music that many clubbers termed 'forward music.' Ammunition Promotions held the Forward club and promoted the night as 'b-lines to make your chest cavity shudder.'

Forward also ran a pirate radio station called *Rinse FM*. This featured artists such as DJ Hatcha, Kode9, Zed Bias, and J Da Flex who would play out the latest 'forward' tracks. It's believed that during this time, Ammunition Promotions coined the term 'Dubstep' in an XLR8R magazine in 2002.

Throughout 2003 DJ Hatcha pioneered the direction of Dubstep. Working in a record store that was frequented by artists such as Skream (who later worked in the same store), Benga, and Loefah, these artists would pass their dubplates to him to broadcast on both *Rinse FM* and club Forward.

As Dubstep gained momentum on the underground scene, in 2003, a new event named *Filthy Dub*, promoted by Plastician (formerly PlasticMan) and David Carlisle, was established to promote Dubstep further. This included records from N-Type, Benga, Skream, and Digital Mystikz.

While Radio 1 DJ John Peel had been a staunch supporter of Dubstep since 2003, it was Digital Mystikz (aka artists Mala and Coki), along with Loefah and Sgt. Pokes, who managed to bring Dubstep to the mainstream market.

After creating the Dubstep label DMZ, they held a bimonthly DMZ nightclub at the *Mass* club complex in Brixton. Radio 1 DJ, Mary Anne Hobbs, heard this new genre at DMZ in 2006 and gathered all of the top Dubstep producers together for a Radio 1 show entitled 'Dubstep Warz.' This proved to be the turning point of what, so far, was mainly an underground form of music.

By 2007, the sound of Dubstep began to influence the commercial mainstream market. Britney Spears employed the deep bass wobble that was typical of the genre in some of her music (in particular 'Freakshow'). Then, in 2009, La Roux asked Skream to remix their single 'In For the Kill,' alongside handing other tracks to artists such as Nero and Zinc to remix.

In 2010 Dubstep had become an unstoppable force. Magnetic Man reached number 10 in the UK chart with his track 'I Need Air' (many inexplicably consider this Dubstep) and 'Katy On a Mission,' produced by Benga debuted at number 5 in the UK singles chart and remained in the top 10 for another five weeks. Skrillex – an artist often wrongly attributed as the father of Dubstep – released 'Scary Monsters and Nice Sprites' in 2010 but it was DJ Fresh who scored the first Dubstep UK number 1. 'Louder' conquered the UK charts in 2011, followed shortly after by Nero's 'Promises,' a second Dubstep track that reached the number 1 spot.

MUSICAL ANALYSIS

Dubstep can be characterized as featuring heavy bass lines (often as low as 30 Hz) mixed with 2-Step-*style* drum rhythms and dark reverberated chords. It is typically composed of a 4/4 signature and a tempo of 140BPM. However, due to the nature of 2-Step, it often results in the music appearing to be half of that to the listener. The music is primarily composed of minor keys and, for many examples, F minor and E minor appear to be a popular choice.

For this chapter, we will examine the general production ethics in the creation of a typical Dubstep track. As music is an entirely individual, artistic endeavor, the purpose of this analysis is not to lay down a series of 'ground rules,' but, instead, will describe some of the general principles behind the production. In the end, it's up to the individual (i.e., you) to experiment and create music that suits your particular style.

Perhaps the best place to start in the production of Dubstep is with the drum rhythms. Initially, the beats followed the 2-Step drum rhythm with a kick drum occurring on the first and third beat of the bar and the snare dancing around this, but this has changed over time. Artists now place a snare on the second, third or fourth (or all) beat of the bar and the kick dances around this snare position. Since the snare remains constant on specific beats of each bar, it reinforces the slower half-tempo paced feel of the music.

In many examples, the kick does not occur at the same time as the snare and rarely happens *on* the beat. Instead, the kick will shift around this beat to introduce a skippy feel to the rhythm. Moreover, this kick pattern will sometimes run in a polymeter fashion with the kick pattern only completing after a period of three or five bars rather than the usual one or two.

In Dubstep, the kicks will often remain 'tight' and light with almost a wood-style character and a bright transient stage for it to cut through the sweeping subsonic bass instruments. These styles of kick are available on the multitude of Dubstep sample content that has appeared, but many producers will build their own in kick in synthesizers.

A good starting point for a Dubstep kick is an 80 Hz to a 120 Hz sine wave with a positive pitch envelope with no attack and a medium release. The pitch envelope is typically logarithmic with a short decay. To help the kick remain above the sub bass, it's sometimes layered with a bright sample such as a hi-hat or you could use square wave with a fast amplifier attack and decay setting. If pitched correctly, this produces a short, sharp click that, when layered above the kick, helps it pull to the front of the mix.

The kick is usually kept dry but can be treated to small amounts of reverb. Here, a little room style reverb with a long pre-delay so that it bypasses the transient combined with a short tail is a good starting point. Many producers will also EQ the low-mid of the kick to bring out its presence and energy. Typically, small boosts of 1 to 3 dB at 100 Hz alongside a low shelf to remove all of the kicks frequencies below 70 Hz. This increases presence and makes more frequency room for the bass to prevent conflicts.

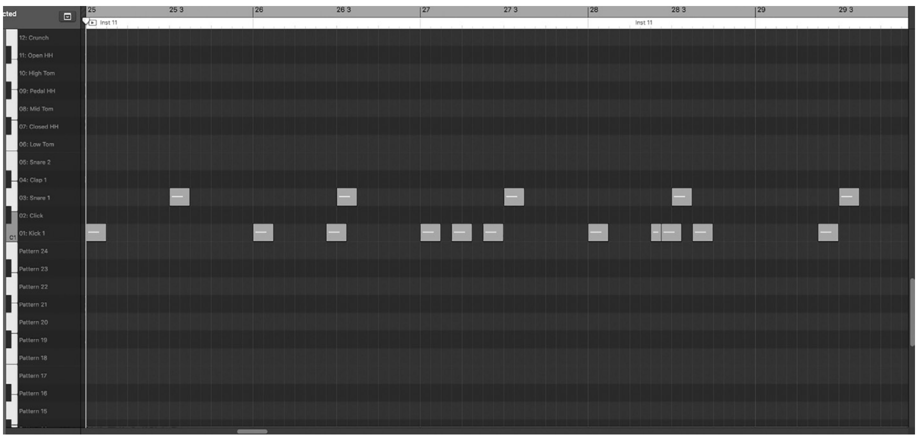

FIGURE 30.0
A typical Dubstep kick and snare arrangement (note how the snare is 'fixed' occurring on the third beat of every bar and the kick pattern changes through a period of five bars before repeating)

In addition to reverb and EQ, compression can play a role in achieving the correct tonal character. For this, the attack of the compressor should be configured so that it bypasses the transient but captures both body and tail. Set the ratio to 2:1 and while listening to the kick, reduce the threshold until the kick begins to punch.

The snares in Dubstep are often short and snappy and sometimes layered to create a hard transient. The transient snare is usually from a TR909 or TR808 and treated to processing and effects. Typically, this involves compression with a fast attack so that the compressor captures the transient and a 4:1 ratio to pull down the transient and produce a harder timbre. This is often followed by EQ providing 2 dB to 4 dB boosts at around 200 Hz to further pull out the transient of the snare.

The snare may be layered, or a secondary snare may be used, depending on the style. Typically, it's a snare mixed with noise or bitcrusher to give it a harder edge. A noise snare can be programmed in most synthesizers with a square oscillator and some white noise with a fast attack and medium decay on the amplifier envelope. A high-pass filter will remove the low end, or you could employ a bandpass to capture more of the central energy. If possible, apply a different amp EG for the noise and square wave. The square wave is best with a fast decay while the noise can ring a further by increasing its decay parameter. The further this rings out, the more slurred it will appear. EQ can then bring out its noise characteristics. In many Dubstep tracks, the prominent frequencies of the snare tend to be around 3.5 kHz.

Dubstep snares are treated to a room style reverb although often the reverb's tail is longer than a kick, allowing it to tail off gently. The reverb settings to employ here depend on the sound you want but typically it should remain relatively short so as not to wash over the loop.

Some tracks in Dubstep will feature a reverse snare that crawls up to the original snare. You can replicate this by duplicating the noisy snare and reversing it in the DAW's editor. You can then apply reverb with a short pre-delay and a tail of approximately 100 to 300ms. This effect is printed onto the audio and the audio file and it's reversed again, so it now plays the correct way around. This effect is typically applied over a period of two bars. The first reverse snare will build up to the first snare while in the second bar it may be time-stretched, so it occurs much more quickly than in the previous bar. This produces a syncopated feel over two bars that will contribute to the half-tempo feeling.

As with almost all genres of dance music, the snares are often treated with micro-tonal adjustments or filter modulation. This isn't as strict as in many other styles, however, since the snares occur over two bars and not one but it can add to the motion of the loop. For the Dubstep track, I produced in the resources file, I employed pitch modulation between the two snares. Duplicating the snare event and pitching it up in Logics editor by +7 cents. I then bounced the file and reinserted it into the project so that the second bars contained the pitch-shifted snares.

The closed hi-hats often exhibit a triplet feel or in some examples will remain more spacious, only occurring every second and third bar out of a four-bar loop.

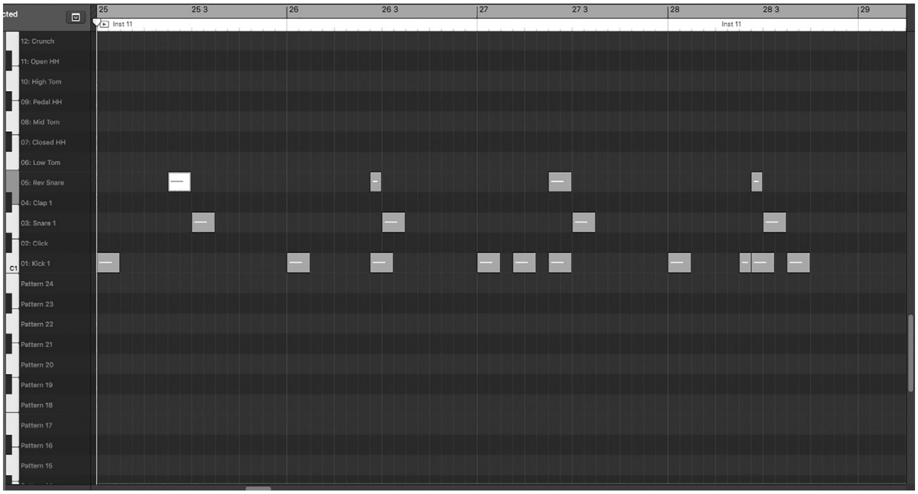

FIGURE 30.1
A reverse snare arrangement (note how the second reverse snare cuts short of its goal – this is shown in MIDI for clarification)

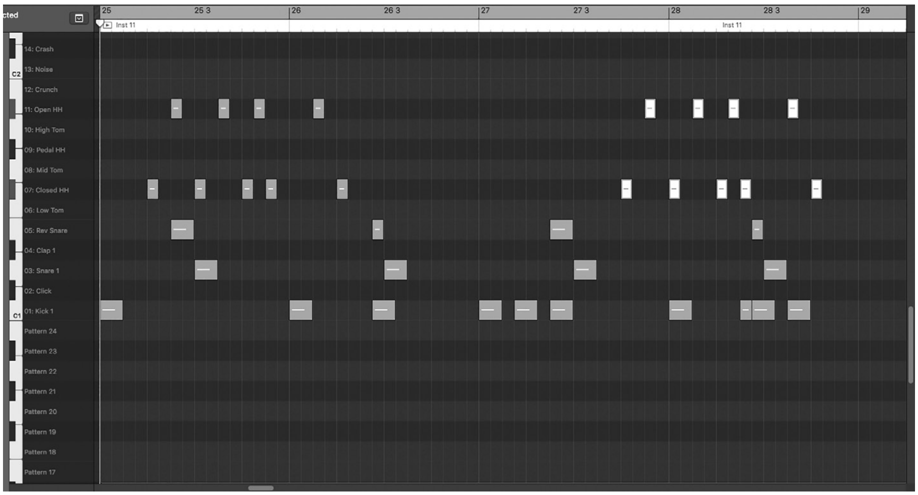

FIGURE 30.2
A typical sparse hi-hat arrangement

Here, they may just happen six times in one bar positioned close together off-beat and then in the next bar may appear only three times off the beat.

These rhythmical patterns of the hi-hats are entirely up to your creativity, but they do typically play off the beat to help towards creating the syncopated feel. Constant 1/16th patterns are avoided because these ‘give away’ the much faster underlying tempo and destroy the illusion of a slower pace. Open hi-hats are employed, but they only appear occasionally, sometimes spread out to occur once in four bars or sometimes they will occur on each beat at the end of a structural cycle of the music.

The closed hi-hat timbres very often exhibit a dirty, bit-reduced tone. They usually alternate between two different timbres or have a strong form of pitch or filter cyclic modulation applied so that it becomes evident. Typically, hi-hats will be taken from sample content and treated with numerous effects and processors such as bitrate reduction and light distortion.

Synthesized hats may give you more of an individual sound, and these can easily be created by ring modulating a high-pitched triangle wave with a lower pitched triangle. This produces a high-frequency noise, so it only requires a fast attack sustain and release on the amplitude to create the final sound. If the hats sound bland, you can augment them with some white noise using the same envelope.

Once you have the timbre, shortening the decay will change between an open hat and a closed hat. It's also worth experimenting with changing the decay slope to logarithmic or exponential to produce fatter or thinner sounding hats.

Beyond these instruments, further ancillary percussive instrumentation is rare in Dubstep with many artists preferring to leave the beats sparse to maintain the slower paced feel of the music. Occasional instruments may be used, but these will be spaced at every four, eight, or 16 bars and often only consists of a single percussive strike that will ring out for the duration of a bar.

THE BASS

Perhaps the key feature of Dubstep is the 'wub.' This is a low-frequency bass that exhibits an almost distorted, dirty sound. It will cyclically modulate or 'wobbles' in frequency or pitch during its duration. Indeed, the 'wub' in many tracks will remain at the same melodic pitch in the workstation's piano roll, and it's the cyclic movement of the filter that creates the interest.

There is a multitude of different styles of 'wub,' but for the filter to sweep the sound, they must be rich in harmonics. Many Dubstep producers will use wavetable instruments such as Xfer's Serum or Native Instruments' Massive. Here, you can choose any interesting wavetable that is rich in harmonics and can be swept via an LFO on the filter.

Typically, the amplifier envelope should employ a fast attack with a short decay, high sustain, and no release. A second envelope set to a fast attack with a medium decay, no sustain, and an extended release can be used to modulate a low-pass filter. Here, the cut-off should be set about half way and the resonance should be set around a quarter.

This will produce the necessary bright harmonically rich timbre, but these harmonics should be swept with a filter, or alternatively, the wavetable should be modulated via an envelope. If using a filter, bandpass with a low cut-off and a resonance set mid-way is a good starting point. The filter can be tempo synced and modulated via a triangle LFO at a 1/4 triplet of the tempo of the sequencer.

DMM resource: An audio example of programming a typical 'wub' sound.

While this 'wub' forms the foundation of many Dubstep records, it is unusual for any track to feature only one bass and most will contain a combination of three or maybe four bass lines continually interacting with one another at different positions in the bars to create a call and response system within the music.

For example, one bass line will make a call while a secondary bass – employing a different sonic texture and 'wub' – will reply to it. This can be followed by a call made by yet another bass line, followed by a reply in yet another texturally different bass line. Generally speaking, many of these single note bass lines will be a quarter, half, or full notes and often straddle over the bars to maintain the half tempo feel. But a brighter bass in response to this call may consist of a series of short 1/16th hits to push the tempo of the music.

More importantly, almost all examples of Dubstep will feature a low-frequency sub bass line that underpins the record and supplies the trouser flapping sub-undertones that push the club's speakers to their limits.

These characteristic sub basses can be created in most synthesizers with a sine or triangle wave. The amplifier attack should be immediate and the decay setting is then increased until it fills the lowest frequencies of the music. It may also be sensible to modulate the pitch by a couple of cents using an envelope set to a slow attack and medium decay. This will create a bass timbre where the note bends slightly as it's played.

Alternatively, you can use an LFO set to a sine wave with a slow rate and set it to start at the beginning of every note. Experimentation is the key, changing the attack, decay, and release of the amp or/and filter EG from linear to convex or concave will also create new variations. For example, changing the decay to a curved slope setting will produce a more rounded bass timbre. Similarly, small amounts of controlled distortion or very light flanging can also add movement.

Typically, these styles of sub bass will employ more extensive notes that are treated to pitch modification. This movement is commonly recorded 'live' into

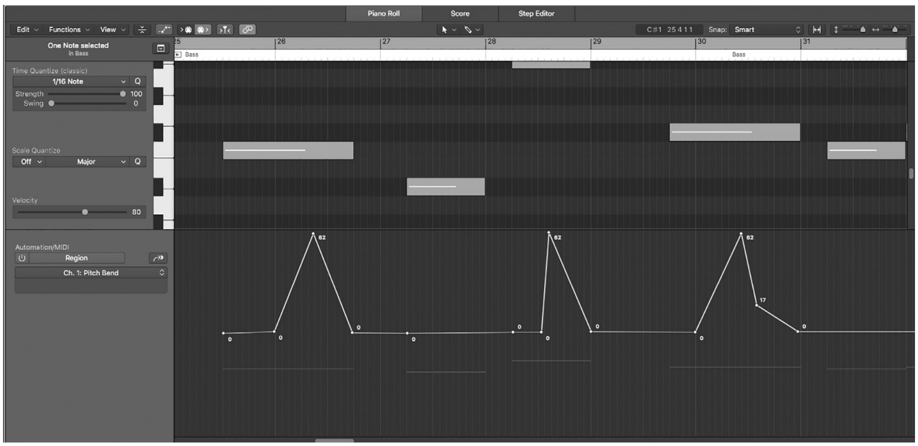

FIGURE 30.3
An example of a sub bass rhythm in Dubstep (note the pitch bend)

the sequencer via the pitch wheel and then further editing in the sequencer to create bass lines that warp in pitch and can act as either a call or a response.

The rhythmic movement and interaction with the bass and rhythms provide the basis for this genre, and therefore it's also worth experimenting by applying effects to this bass timbre. While some effects should be avoided since they will spread the sound across the image, in this genre, the bass is one of the most critical parts of the music so small amounts of automated distortion can create interesting effects.

As with the drum rhythms, creative compression can also help in attaining an interesting bass timbre. As before, try accessing the compressor as a send effect with a medium threshold setting and a high ratio. The returned signal can be added to the uncompressed signal; you can then experiment with the attack and release parameters to produce an interesting bass tone. A conventional technique here is to send the transient snare and kick to a bus (0 dB) and then use this bus as a side-chain for a compressor placed onto the bass and lead instruments.

The settings to use for the compressor depend on the effect you wish to accomplish, but a good starting point is a fast attack with a 50ms release and a ratio of 2:1. Once applied, lower the threshold and increase the make-up gain until the effect is evident.

CHORDS, MELODIES, AND EFFECTS

In many examples of Dubstep, there are no melodies to speak of, and much of the instrumentation is in the form of very, very simple chords and progressions. Indeed, it is rare to hear a chord progression move beyond two chords, and it's only in a few examples where the chords will move beyond three.

Unlike *traditional* music theory, the chords do not resolve in cadence, but are typically constructed from a random selection, most commonly the minor chords of that particular minor key. For example, the harmonies of III, VI, and VII (in the natural minor) or III, V, and VI (in harmonic minor) are often avoided.

Effects can play a vital role in the creation of new strings and pads for the genre, although these should be used conservatively so as not to detract from the bass rhythm. Often, significant chorus effects, rotary speaker simulations, flangers, and phasers can all help to add a sense of interest, but reverb is *the* leading contender. Cavernous reverb is often employed to fill out the spectrum and bring energy and space to the mix while the underlying bass punches and shudders the listeners' chest.

In addition to the chords, some Dubstep will feature short vocal snippets and even though there have been some more commercial mixes that have featured a lengthy vocal takes (such as Nero's 'Promises'), for many aficionados, this is considered commercial and outside the bounds of where Dubstep has originated from. The snippets are commonly sampled from TV, old vinyl records, or are gleaned from sample CDs and chopped/edited to suit the music.

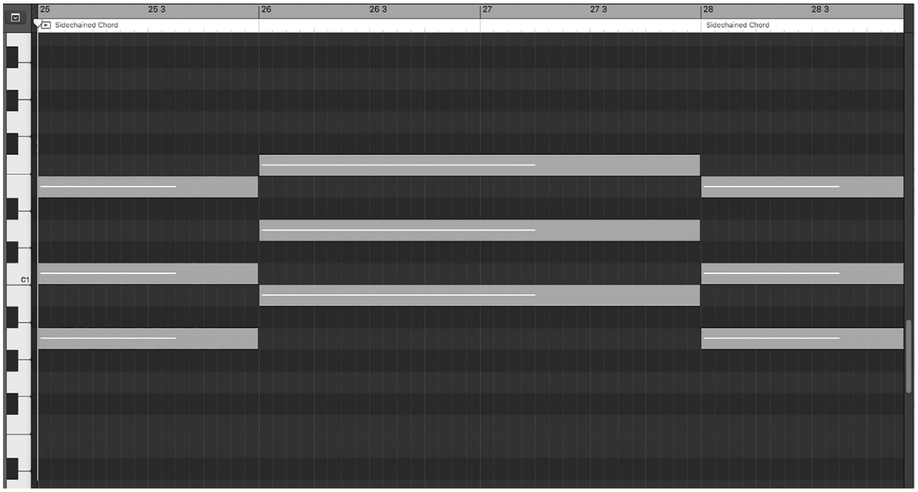

FIGURE 30.4
A typical Dubstep chord (i.e., harmonically simple)

Further sound effects can be generated by whatever means necessary, from sampling and contorting sounds or samples with effects and EQ. For twisting audio, the Sherman Filterbank 2, the Camelspace range of plug-ins, Glitch, Sugarbytes' Effectrix, or Steinberg's GRM tools are almost a requisite for creating strange evolving timbres.

That said, the effects and processing applied are, of course, entirely open to artistic license as the idea is to create anything that sounds good and fits within the mix. Transient designers can be especially useful in this genre, as they permit you to remove the transients of the percussive rhythms that can evolve throughout the track with some thoughtful automation. Similarly, substantial compression can be used to squash the transient of the sounds, and with the aid of a spectral analyzer, you can identify the frequencies that contribute to the sound while removing those surrounding it. Alternatively, pitch-shift individual notes up and by extreme amounts or apply heavy chorus or flangers/phasers to singular hi-hats or snares or try time stretching followed by time compression to add some digital clutter and then mix this with the other loops.

Ultimately, however, as with all the chapters in this section, the purpose here is to merely reveal some of the techniques used by producers and show how the theory and technology discussed earlier in the book combine to produce a track. However, it should be viewed as offering a few basic starting ideas that *you* can evolve from.

There is no one definitive way to produce this genre, and many artists have mixed the four to the floor kicks of dance music with Dubstep's low bass, to then slip in and out of the 2-Step vibe part way through the music. This is a relatively new genre in many respects and, to some artists one, that is still finding its feet. Indeed, it will not be long before the style begins to diversify like every other genre of EDM and new Dubstep forms such as Commercial Dubstep, Funky Dubstep, Hardcore DubStep start to flourish as artists expand the music further.

As ever, however, the best way to learn new techniques and production ethics is to actively listen to the current market leaders and experiment with the tools at your disposal. There are no right or wrong ways to approach any piece of music, and if it sounds right to the producer, then it usually is. New genres do not evolve from following step-by-step guides or from emulating peers; producers who experiment and push boundaries create them.

Nevertheless, with just these essential elements, it is possible to create the primary focus of the music, and from here the producer can then look towards building an arrangement. The theory behind arranging has already been discussed in an earlier chapter but merely listening to the current market leaders mixed among the method discussed in previous chapters will reveal the current trends in both sounds and arrangement.

DMM resource: The companion files contain a short excerpt of a Dubstep track I created using the techniques described above.

Track 1 is the excerpt played through GM instruments.

Track 2 is the excerpt with full production.

CHAPTER 31
Ambient and Chill Out

If there's loads of material going by you don't notice the individual things quite so much. Also, it foregrounds the sonic dimensions like electronic ambient music, it pushes all of that color to the foreground, so you hear every little atom of sound.

Max Richter (quoted in The Quietus, 2015)

Ambient music has enjoyed a long if somewhat diverse history and its subsequent offshoots have formed an important role in dance music since 1989. However, it re-established itself as part of the electronic dance music scene when beats again took second place to atmospheric content, and it became relabeled as 'Chill Out' by record labels and the media.

The roots of ambient music are nothing if not unusual. Apparently, Brian Eno was hit by a taxi in the 1970s, and, while recovering, he listened to a tape recording of harp music. These tapes were old, resulting in the volume of the music fluctuating and on occasion dropping so low in volume that it mixed with the rain hitting the hospital windows.

This second *accident* formed the beginnings of ambient as Eno began to experiment by mixing real-world sounds such as whale song and wind chimes with continually changing synthesized textures. Described by Eno himself as music that didn't draw attention to itself, it enjoyed some success but was soon relabeled as 'muzak.' Subsequently, poor imitations became background music for elevators, and, to many, the genre was quickly written off as 'music suitable only for tree huggers.'

When the rave generation emerged in the late 80s and early 90s, Alex Patterson, a rave DJ, began to experiment with Eno's previous works. He would play the music to clubbers in small siderooms who were resting from the faster paced hard hitting beats of the main room. These side-rooms were often called as 'chill-out' rooms, a place you could go to to take a break from the manic up-tempo beats.

As the slower paced music grew in popularity for those looking to relax, Patterson teamed up with fellow musician Jimmy Cauty to form 'The Orb' and released their album *A Huge Ever Growing Pulsating Brain That Rules from the Centre of the Ultraworld*. This is believed to be the first ever ambient music aimed at chill-out rooms.

Soon after its release, Patterson and Cauty went their separate ways. Jimmy Cauty teamed up with Bill Drummond to form The KLF while Patterson continued to write under the moniker of The Orb and continued to DJ in the chill-out rooms.

As chill-out rooms grew to form a fundamental part of the rave scene, 'Ambient House' developed into its genre. DJs moved from the main rooms and became VJs (**V**ideo **J**ockeys). They were mixing real-world sounds with drawn-out drum loops while simultaneously projecting and combining images onto large screens to accompany the music.

By 1992, the genre reached the mainstream. Different artists adopted the scene, each putting her twist on the music and diversifying it into genres such as *Ambient Dub*, *Conventional*, *Beatless*, and *Soundscape*.

By 1995, record companies adopted the sound and saturated the market with ambient compilations. Even artists who had ignored the genre jumped on the bandwagon in the hopes of making a quick buck out of the new fad. Eventually, as with most musical styles, Ambient House became a victim of its success. The public grew tired of the sound and, to the joy of many a clubber, it was no longer the new 'fashion' and returned to its humble beginnings – the chill-out rooms.

Fast forward to the year 2000, and a small Balearic island in the middle of the Mediterranean began to revive the public's interest in Ambient House. DJs in Ibiza's Café Del Mar started to tailor music to suit the beautiful sunsets by mixing Jazz, Classical, Hispanic, and New Age together to produce laid-back beats for clubbers to once again chill out to. Now repackaged and relabeled 'Chill-Out music,' it's enjoying renewed interest and has become a genre in its own right.

However, while Chill Out certainly has its roots deeply embedded in ambient music, they have over time become two very different genres. Thus, as this book is dedicated to dance music, for this chapter, we'll concentrate on Chill Out rather than ambient …

MUSICAL ANALYSIS

Chill Out is music that incorporates elements from many different styles such as Electronica, New Age, Classical, Hispanic, and Jazz. However, it's this very mix of different styles that makes an exact definition impossible. Indeed, provided that the tempo remains below 125BPM and it employs a laid-back groove, it could be considered Chill Out. In fact, the only way to analyze the music is to take Brian Eno's advice in that it shouldn't draw too much attention to itself.

Defining precisely what this means is difficult, but it could be explained as exhibiting a slow rhythmic pace or trance-like nature combining both electronic and real instruments. These are often backed up with dropout beats and smooth, sometimes haunting, vocals. The sound is continually evolving, however, and music labeled as Chill Out could equally be considered gentle pop music.

Chill Out can employ almost any time signature from the four to the floor, to a more swing oriented 3/4 but, for many examples, the signature remains at 4/4. Instead, it's the positioning of the beats that sometimes undermine the 4/4 feel. Regarding tempo, it ranges from 80BPM and can move towards the upper limits of 126BPM. Although it can be written in any scale, popular scales appear to be C major, D major, E♭ major, A minor, E minor, C# minor and G minor.

For this chapter, we will examine the general production ethics in the creation of a typical Chill out track. But, music is an entirely individual, artistic endeavor and the purpose of this analysis is not to lay down a series of 'ground rules' on how it should be created. Instead, it intends to describe some of the general principles behind how to accomplish the appropriate arrangements, sounds, and processing. Indeed, it's up to the individual (i.e., you) to experiment and create music that suits your particular style.

THE RHYTHMS

Unlike many other genres of dance music, there are no particular distinctions as to what makes a good Chill-Out drum rhythm and the only guide that can be offered is that the loop should remain relatively simple and exhibit a laid-back feel.

The kick drum can occur on the beat, every beat, or it can be less frequent, appearing on the second and fourth, or first and third beat, or any 16th division thereof. Indeed, in many Chill-Out tracks where the kick drum occurs on all four beats, there is sometimes an additional kick drum that will happen outside of this standard beat on a 1/8th or 1/16th position. This moves the music away from the rigidity of the usual four to the floor and can help to increase the relaxed feel of the music.

Like the kick, the snare drum can appear anywhere in a Chill-Out loop but generally speaking if the kick occurs *off* the beat, the snare will happen *on* the beat to maintain rhythmical positioning with the listener. This same theory can work in opposite too, if the kick lands on the beat, the snare can occur off the beat. It is important to note that this relationship should not be solely attributed to the laid-back feel of the loop, however; this relaxed motion is a result the positioning of the hi-hats and any ancillary instruments.

As shown in Figure 31.0 multiple rhythms are employed for the hi-hats. Unlike many other genres, the closed hats do not occur on the standard 16th, and instead, the patterns are more complex, often consisting of both hats and shaker rhythms that syncopate with the regular beat. It's this syncopation of hats,

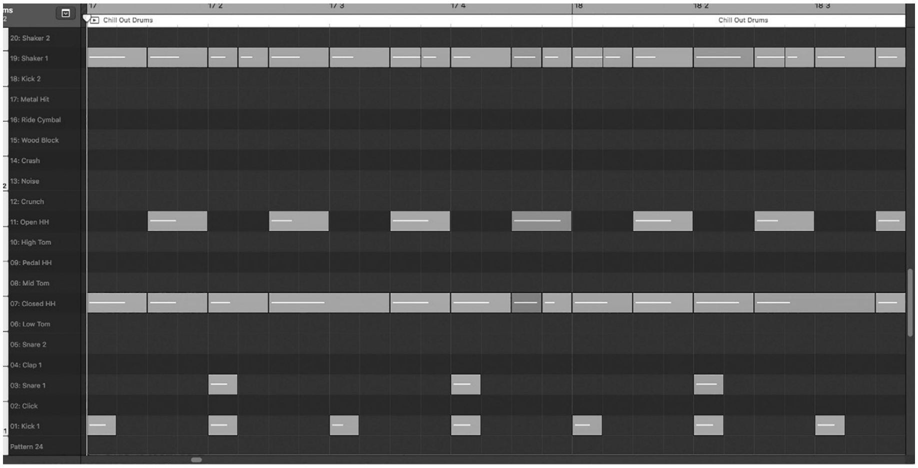

FIGURE 31.0
The basic foundations of a Chill-Out loop (as in the example supplied with the book)

moving around the snare and open hats that aids in creating the chilled and relaxed feel to the music.

Alongside syncopation, groove quantization is of paramount importance. This shouldn't be applied heavily, small amounts of swing to the closed hats and sometimes the kick (or snare) will further increase the laid-back groove. This form of quantizing should be applied to either the kick *or* the snare, not both. If both kick and snare are treated to groove quantization, the music will appear sloppy and out of time. A good starting point for groove quantize is 1/16th with 60% swing.

This, of course, is only a general guideline to the drum patterns and it's open to artistic license. The key is not to produce a loop that sounds rigid or programmed. You should experiment by moving the kicks concerning the snares to adjust the interplay between the two. If the tempo appears too fast, reducing the number of snares, hi-hats or auxiliary instruments will slow it down. This is preferable to slowing down the pace as this will affect the rest of the instrumentation.

Auxiliary instruments such as congas bongos, toms, cabassas, triangles, shakers, and tambourines also often make an appearance. These offer further syncopation from the main rhythm and use both hemiola and polymeter.

RHYTHMIC TEXTURES

The kicks in Chill Out often remain 'loose,' exhibiting more of a boom texture than a tight, controlled one. Kick layering is rarely necessary with Chill Out because the purpose is not to punch the listener in the chest. Consequently, a single kick is often sourced directly from sample content or programmed by the producer on any capable kick synthesizer.

A good starting point is a 90 Hz sine wave with positive pitch modulation using a fast attack and a medium to longish decay period. There is little need to employ a 'click' at the beginning of the waveform because this can make the kick appear tighter but experimentation is essential. If possible, experimenting with the kick's decay transition is critical to the creation of creating a suitable kick drum. Here, exponential is commonly the favored choice so that the pitch curves outwards as the pitch and amplitude falls as this produces a thicker sound.

If a kick sample appears tight or exhibits a bright attack stage, a compressor with a low threshold, medium ratio and fast attack so that it clamps down on the transient may help towards the creation of a looser sounding kick. Alternatively, a transient designer can be used to remove the initial attack.

The kick is typically treated to small amounts of reverb, and a room reverb is a preferred choice with a long pre-delay so that the initial reflections bypass the transient of the beat. The reverb tail can be longer than most other genres of music, ranging from 60ms to 100ms, but it should not be allowed to wash over other percussive instruments otherwise it may affect the groove. If high reverb settings are used to make the kick appear 'loose,' then a noise gate should follow the reverb to ensure that tail doesn't reduce the impact of any other instruments.

In direct contrast to the kicks, the snares or claps often employ a sharp transient stage as these commonly remain apparent in the loop, more so if it's a particularly busy loop packed with auxiliary instruments. As with kick, snares (or claps) can be sourced from sample content and processed with EQ to make them appear sharp and snappy, or snares can be programmed in most synthesizers with a triangle oscillator and some pink noise.

The amplifier's attack is set to immediate, and the decay can be set anywhere from medium to long depending on how slurry you want the snare to appear. This is treated with a high-pass filter to remove the low end, but alternatively, a bandpass filter can be used if the timbre exhibits too much high-end energy. If possible, employ a different amp EG for both the noise and triangle wave. The triangle wave can be kept short with a fast decay while the noise can ring a little further by increasing its decay parameter. The further this rings out, the more slurred and 'ambient' it will appear. If the snares are too bright, it can be removed with some EQ or preferably, try replacing the triangle wave with a pulse and experiment with the pulse width.

The snares benefit from small amounts of reverb by sending, not inserting, them to a reverb unit. A typical snare room preset that's available on nearly every reverb unit around will usually suffice. But depending on the snare positioning and tempo, the decay should be set carefully so that it does not wash over the transients of further sounds.

The hi-hats in this genre are often sourced from the multitude of sample content on the market and are treated to a variety of effects to the producer's taste. This can include noise gates to shorten the hat, transient designers, EQ, distortion, and reverb. They can, however, equally be synthesized through ring modulation.

By modulating a high-pitched triangle wave with a lower pitched triangle, you can produce a high-frequency noise. As a starting point, set the amp envelope to a zero attack, sustain, and release with a short to medium decay. If there isn't enough of a noise present, it can be augmented with a white noise waveform using the same envelope.

Once this basic timbre is constructed, shortening the decay creates a closed hi-hat while lengthening it will produce an open hat. Both open and closed hi-hats will commonly be treated to small amounts of delay, but this must be applied with caution so as not to upset the syncopated feel of the hats.

Ancillary instrumentation can also be used for the creation of a syncopated groove. These sounds can be sourced again from sample content or synthesized in a variety of ways. The various synthesis techniques have been discussed in detail in previous chapters. However, one favorite ancillary instrument is the Jazz brush. This can be produced by introducing a small amount of pink noise over the top of the snare. This gives the impression of a brush stick being used to play a snare instrument, and if treated with small amounts of reverb, distortion, and EQ it can produce a convincing effect.

The snares, hats, and any ancillary percussive instruments are often treated to a form of cyclic pitch or filter modulation. There is no defining method for any of these instruments, but the effect is lightly applied to each instrument over a different period of bars. For example, the snares may be pitched to change by a few cents every fourth beat of the bar, whereas the closed hi-hats may be cyclically modulated with a filter every two bars, one open hi-hat every three bars, another open hat every four bars and ancillary instruments every six bars. This pattern is changed every so often on a structural downbeat to inject more interest into the drum rhythms.

MELODIC LEADS

Although it isn't unusual for a Chill-Out track to be based entirely on synthetic instruments, real instruments are increasingly common. These can range from acoustic guitars playing Hispanic-style rhythms or chords through to Classical strings, pianos, wind instruments, or a combination thereof.

These are often laid down *before* the bass since they're very commonly sourced from sample content or other records, and it's easier to form a bass line around their progression than it is to edit the pitch of the audio. This is not to suggest it does not occur, however, and many artists will source a lead instrument from a record. They then spend a good day adjusting the rhythms and pitch of the audio using software such as *Melodyne*.

Of course, samples of real instruments do not necessarily have to be used (although in the majority, they do sound better) and provided that you have access to either a sample-based instrument such as the Native Instruments' Kontakt instrument, it is possible to program expressive and realistic instruments.

When programming a real instrument, you must pay attention to the scale and range of the song. This is because every acoustic instrument has a limit to how high and low it can play. For instance, an acoustic guitar has a standard tuning of E, A, D, G, B, E with the latter E a major third above middle C. This is important because many sample-based instruments will stretch a sample beyond its normal range to fill the entire octave range. If you exceed the limitations of an acoustic instrument, the human ear will instinctively know that it's been programmed rather than played.

Creating a realistic instrument consists of more than just striking a few notes and regardless of how good the original samples may be. There are Kontakt instruments that consist of more than 20 Gb of samples, but if you do not emulate the natural flow of an instrumentalist, you will be unable to recreate a realistic interpretation. For example, when programming an acoustic guitar, you should first take into account the natural movements of a guitarist.

If the style is Hispanic, the strings are commonly plucked rather than strummed so the velocity of each note will be relatively high throughout. This should be mapped within the instrument to control the filters low-pass cut-off parameter (i.e., the harder a string is plucked, the brighter the timbre becomes).

Not all of these velocity plucks will be the same, however, because the performer will play in time with the track and if a note occurs directly after the previous, it takes a finite amount of time to move to the next string. This often results in the string's not being accentuated as much as the earlier tone. Conversely, if there is silence or distance between the two notes, then there is higher likelihood that the string will be plucked at a higher velocity.

In many instances, if the string has been plucked hard, the resonance may still be dying away as the next note starts and should be taken into account. Similarly, an acoustic guitar will not bend more than an octave, so it's prudent to set the sampler to an octave so you can use the pitch bend wheel to create slides. Moreover, in between notes that are close together, you should add the occasional fret squeal for realism.

If the guitar is strummed, then you need to take the action of the hand into account. Commonly, a guitarist will begin by playing *downwards* rather than upwards. If the rhythm is quite fast on an upward stroke, it's rare that all the strings will be hit. Indeed, it's quite unusual for the bottom string to be struck, as this tends to make the guitar sound too 'thick.' Additionally, all the strings will not be struck at the same time due to the strumming action. This means that each note on message will occur a few ticks later than the previous, which will depend on whether it's strummed upwards or downwards and the speed of the rhythm.

Interestingly, from recording guitarists, I've noticed that if they start on a downward stroke, they come in a little *earlier* than the beat. But, if they start on an upward stroke, they tend to start *on* the beat. I have no idea why they do this, but it's a widespread occurrence. Answers on a postcard, please.

FIGURE 31.1
Native Instruments' Kontakt Running Evolution steel strings

WIND INSTRUMENTS

Fundamentally there are two types of wind instrument: brass and reed. However, for programming realism into both these instruments, they both follow very similar guidelines.

If you want to replicate a wind instrument, it is wise to invest in a breath controller. These connect to the computer via a USB input or sometimes an available MIDI port. They measure changes in air pressure as you blow into them and convert them into a CC2 message (breath controller). This can then be used to control the sample-based instruments. They are expensive, however, so if the use of wind instruments is only occasional, you can program them.

With any wind instrument, the volume and brightness of the notes are proportional to the amount of air pressure in the instrument. Nearly all sample-based instruments will employ the velocity to control the filter cut-off so the brightness can be controlled with the judicious use of MIDI velocity. The volume of the note is a little more complicated because many instrumentalists will deliberately by blowing softly before increasing the pressure to force the instrument to become louder.

This means that the notes begin softly before rising in volume and occasionally pitch. Perhaps the best way to emulate this is to carefully program a series of breath controller (CC2) messages while using a mix of expression controllers

(CC11) to control the volume adjustments. Alternatively, brass instruments will often start below the required pitch and slide up to it so this is best emulated by recording live pitch bend movements into a sequencer and editing them to suit later in the piano roll editor.

Many wind instruments also introduce vibrato if the note is held for a prolonged period due to the variations in air pressure through the instrument. While it is possible to emulate this response with expression (CC11) controllers, introducing small pitch spikes at the later stages of sustain often produces better results. These pitched 'spikes' only appear in the later stages of the notes sustain, however, and should not be introduced at the beginning of a note.

Possibly the best way to determine where these spikes should appear is to pretend you're playing a wind instrument. Blow at the start of the MIDI note and when you begin to draw short of breath, insert short pitch bend messages. Alternatively, if the instrument allows you to fade in an LFO, you could use this to modulate the volume and pitch. Using a sine wave at a slow rate, the LFO modulation amount could be controlled via an envelope with a prolonged attack. This way as the attack of the modulation envelope increases, the LFO's augmentation grows stronger.

More importantly for realistic interpretations of wind instruments, remember that musicians need to breathe. This means that you should avoid playing a single note for an excessive period. Similarly, if a series of notes are performed consecutively, the musician needs enough time to take a deep breath for the next note (wind instruments are not polyphonic!). If there isn't enough time, the following note will be softer due to less air velocity from a short breath.

Finally, consider how notes end. Neither reed nor brass instruments stop at the end of the note. Instead, they will fade down in volume while also lowering in pitch as the air velocity reduces (an effect known as *diminuendo*). This can be emulated with a series of expression (CC11) messages and some pitch bend.

SYNTHETIC LEADS

Of course, not all Chill-Out tracks employ real instruments, and some rely solely on synthetic. If you use synthesized leads, avoid aggressive synthesizer patches such as distorted saw-based leads. These give the impression of a 'cutting' track whereas softer sounds appear laid back. This is an important characteristic to keep in mind, especially when mixing and mastering. Slow, relaxed songs will invariably have the mid-range cut to emphasize the low and high end while more aggressive songs have the bass and hi-hats cut slightly to enhance the midrange.

Notably, many Chill-Out tracks refrain from composing a melody of notes less than a 1/8th because shorter notes make the record appear faster. Additionally, with more extended notes, many of the tones can utilize a prolonged attack that will invariably create a more relaxed perception of the music.

In fact, extended attack times form an essential aspect of the music, and it's worth experimenting with all the sounds. By lengthening the amplifier or filter's attack and release times, it can have a significant effect on the feel of the music. Notably, vocal chants are sometimes used in the place of leads because these can be used as instruments in themselves. In many tracks, these are sourced from other records or sample content, but you can always record your own.

Condenser microphones are the preferred choice over dynamic because these produce more accurate results, but the diaphragm size depends on the effect you wish to achieve. Many records will use a large diaphragm, but if you're after 'ghostly' chants, a small diaphragm microphone will produce better results due to the more precise frequency response. Of course, this is the conventional approach and, if, at all possible, it's worthwhile experimenting with different microphones to see what produces the best results.

Vocals will also benefit from compression during the recording stage, but this must be applied lightly. Unlike most other genres where the vocals are often squashed excessively to suit the compressed nature of the music, Chill Out relies on a live feel with the high frequencies intact. A good starting point is to set a threshold of –9 dB with a 2:1 ratio and a fast attack and moderately quick release. Once the vocalist has started to practice, reduce the threshold so that the reduction meters only illuminate on the most active parts of the performance.

BASS

The bass often remains minimal consisting mostly of notes over a 1/8th and up to a bar in length. This is to prevent the underlying groove from appearing too fast. These styles of bass lines follow the same principles encapsulated in the Dub scene – by remaining deep and minimal they slow the pace of the music while also permitting plenty of space for the lead/MC toast.

Indeed, it's important that the bass is not too busy rhythmically or melodically. Many Chill-Out tracks borrow heavily from the pentatonic scale used in both Dub and R&B. That is, they use a five-note scale consisting of only the black notes on the keyboard, and restrict the pitch movements from note to note by five semitones. This limited pitch movement was employed in the example track (although I composed the track in C# minor rather than the pentatonic scale).

Another conventional technique in Chill Out is for the bass (or chords) to occasionally change pitch in a syncopated fashion. Rather than change pitch on every beat or bar, one note may be shorter and syncopate against the rest of the music by changing pitch mid-beat. This technique was used for the fourth bar of the bass in the example track and is shown in Figure 31.2.

Either real or synthesized bass tones work for most Chill Out so you could use Kontakt or program your own in a synthesizer. A good starting point is to employ a sine or triangle wave detuned from a saw or square. Unlike most of the timbres used throughout the production of Chill Out, the attack should be defined to prevent the bass becoming indistinct. The decay is rarely used and

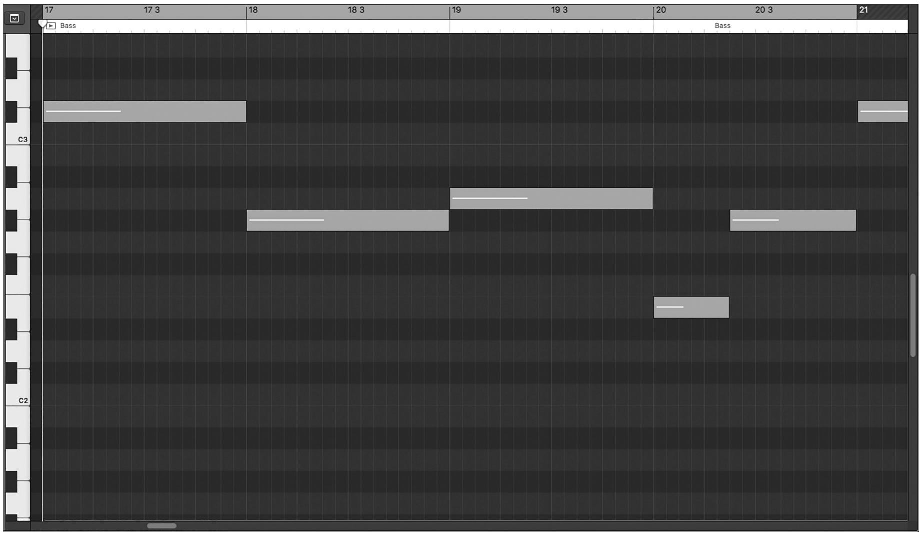

FIGURE 31.2
The syncopated bass used in the example track

instead it will use a long sustain and short release setting. This forces the sound to enter directly into the sustain stage that produces a constant bass tone for as long as the key is depressed. This technique was used for the bass in the example track. I mixed a triangle wave and pulse wave, pitching the triangle wave an octave down from the pulse wave. A low-pass filter was used to remove the upper harmonics of the pulse wave to produce a smoother sounding, deep bass tone.

If you produce bass with little sonic definition, a small pluck can be added by employing a low-pass filter with a low cut-off and high resonance. This can be modulated via an envelope with no attack, sustain and release, but a medium decay. By adjusting the depth of the envelope modulating the filters or increasing/decreasing the decay stage, more or less of a pluck can be applied.

It's also worth experimenting with the release parameters on the filter and amp envelope to help the bass sit comfortably in with the drum loop. In fact, it's essential to accomplish this rhythmic and tonal interaction between the drum loop and the bass by experimenting with both the amp and filters envelopes.

The only effects that are typically applied to the bass are small amounts of reverb, delay, and distortion. As always, these must be applied carefully, and a preferred approach is first to use an M/S processor so that the effects are only applied above 160 Hz to ensure the bottom end of the mix maintains its solidity. For the example track, I applied a small amount of room reverb to the bass, followed by heavy compression.

CHORDS/PADS

Evolving pads and chords are often a crucial element in this genre. Because many of the tracks are thematically pure, a pad can be used to fill the space between the groove of the record and the lead or vocals. What's more, slow evolving strings often add to the overall atmosphere and can be used to dictate the drive behind the track.

Chords and progressions have already been discussed in earlier chapters, but as a refresher, they act as harmony to the bass. This means that they should fit in between the rhythmic interplay of the instruments without drawing too much attention. For this, they should be closely related to the key of the track and not use a progression that is particularly dissonant. Forming a chord progression from notes used in the bass and experimenting with the progression can often work.

For instance, in the example track, the bass consists of five notes: C# > G# > A > E and G#. These same notes were used as the root notes of the chord progression, and this employed some inversions to create chords that added a further dimension to the music.

Although many Chill-Out tracks will meander along like a Sunday stroll, if you place a single chord that plays consistently over a large number of bars, you can lose the feel. You can avoid this by moving the chords back in time by a small amount so that they occur little later than the bar. Alternatively, if they follow a faster progression, they can be moved forward in time to add some push to the music.

If the pads employ a long attack stage, they might not become evident until much later in the bar, and this can change the feel of the music. In this instance, you will need to counter the effect by moving the chords so that they occur earlier.

The instruments used to create chords are typically analog due to the constant phase discrepancies of the oscillators that produce additional movement. Thus analog or analog emulations will invariably produce much better results than an FM synth or one based on sample and synthesis.

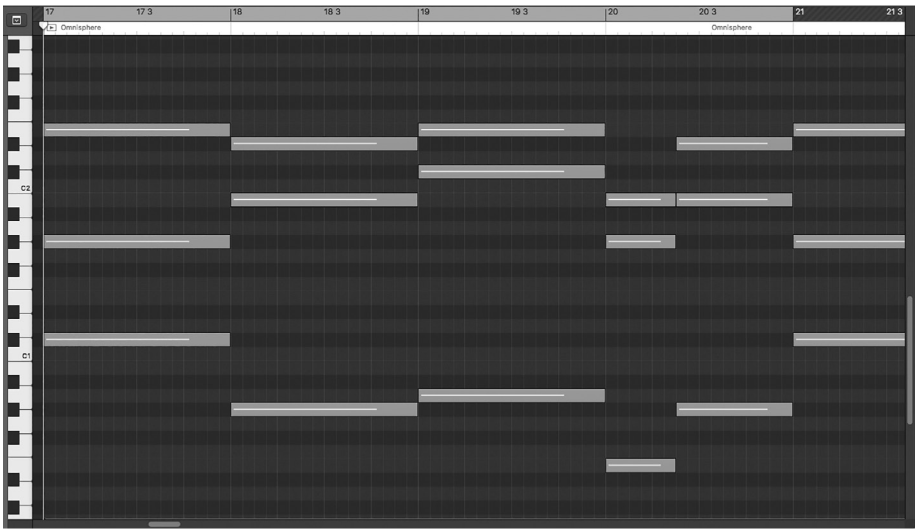

FIGURE 31.3
Chords used in the example track

With the drums, bass, lead, and possibly vocals, there will be a limited frequency range for the chords to sit. If the tone of these chords is rich, it may be difficult to fit them into the mix without resorting to aggressive EQ. This can affect the sonic character, and therefore it is better to *design* the chords in the synthesizer than attempt to carve space for them later with EQ.

If the mix is already busy regarding frequencies, you can build a relatively thin pad that can then be widened if required with effects such as phasers, flangers, reverb, or chorus effects. For this, pulse waves are a good option because they have less harmonic content than saws.

To start, only one pulse wave is required with the amp envelope set to a medium attack, sustain, and release but a fast decay. If this tone is to sit in the upper mid-range of the music, then employ a 12 dB high-pass filter to remove the lower frequencies or use a low-pass to remove the higher frequencies. Then experiment with the resonance until it produces a static tone that sits well on the track. To add some interest and movement to the sound, you can use an LFO with a slow rate to modulate the pulse width of the oscillator and the filter.

If the timbre still appears static, use a different rate and waveform for the oscillator and filter so that the two beat against each other. Alternatively, if the tone seems too thin, you can add a second pulse detuned from the first with the same amp envelope but use a different LFO waveform to modulate the pulse width. Alternatively, you can play lower notes to add further low-end frequencies and energy.

Effects play an essential role in creating new and evolving pads. Broad chorus effects, rotary speaker simulations, flangers, phasers, and reverb can all help to add a sense of movement to help fill out any holes in the mix.

A noise gate can also creatively model the pad. For instance, another channel with a hi-hat rhythm could be side-chained into the gate to produce an impressive gated pad. Or by carefully adjusting the threshold so that it lies just on the edge of the pads volume with an immediate attack and release, the gate will chatter starting and stopping abruptly producing a 'sampled' effect that cannot be replicated through synthesis.

Ultimately, however, as with all the chapters in this section, the purpose here is to merely reveal some of the techniques used by producers and show how the theory and technology discussed earlier in the book combine to produce a track. However, it should be viewed as offering a few basic starting ideas that *you* can evolve from.

There is no one definitive way to produce this genre and the best way to learn new techniques and production ethics is to actively listen to the current market leaders and experiment with the tools at your disposal. There are no right or wrong ways to approach any piece of music, and if it sounds right to the producer, then it usually is. New genres do not evolve from following step-by-step guides or from emulating peers; producers who experiment and push boundaries create them.

Nevertheless, with just these essential elements, it is possible to create the primary focus of the music, and from here the producer can then look towards building an arrangement. The theory behind arranging has already been discussed in an earlier chapter but merely listening to the current market leaders mixed among the theory discussed in the previous chapter will very quickly reveal the current trends in both sounds and arrangement.

DMM resource: The companion files contain a short excerpt of ‘Blue Skies,’ a Chill-Out track released under the name Aeon Soul, using the techniques described above.

Track 1 is the excerpt played through GM instruments.

Track 2 is the excerpt with full production.

‘Blue Skies’ © Aeon Soul (R. Snoman) ft. Talson Moon.

Lyrics by Tom Larkin.

CHAPTER 32

Drum & Bass

It's a ubiquitous piece of the pop culture soundscape…it's been used so much I might argue it's now entered the collective audio unconscious.

Nate Harrison

Pinpointing the foundations of where Drum & Bass originated is difficult. In a quick summary, it could be traced back to a natural development of Jungle, which was itself an elaborate infusion of Breakbeat, Reggae, Dub, Hardcore, and Artcore.

Alternatively, we could look further back to 1969 and suggest that the very beginnings lay with a little known record by the Winstons. The B side of their record 'Color Him Father' featured a six-second drum break. This became affectionately known by many dance musicians as the 'Amen Break,' borrowed from the title of the record *Amen Brother*. The loop has been sampled and rearranged countless times over the years and became the staple sound for many of the rhythms of Jungle and Drum & Bass. Or it could be traced back to the evolution of the sampler, offering the capability to cut, chop, and pitch rhythmic material that created the basis for Jungle and the more refined Drum & Bass.

One thing that we can confirm is that the compound, fast-paced rhythms of Jungle and Drum & Bass have their foundations in Breakbeat, a style that began in the 1970s.

Kool Herc, a Hip-Hop DJ, was experimenting on turntables, playing exposed drum loops (breaks) from records. He would alternate between two singles, spinning one back while the other played to produce a continual loop of just drum rhythms so breakdancers could show off their skills. DJs such as Grand Wizard Theodore and Afrika Bambaataa adopted this style, adding their twists by playing two copies of the same record but delaying one against the other to create complex asynchronous rhythms. However, it was early 1988 and the combined evolution of the sampler and the rave scene that sparked the Breakbeat revolution.

Acid House artists would sample the breaks in records, cutting and chopping the beats together to produce complex breaks that were impossible for a drummer to play. Hardcore emerged as these breaks became more and more complicated. Moving away from the standard 4/4 loops of Acid House, hardcore featured lengthy complex breaks and harsh energetic sounds that were just *too hardcore* for the other ravers.

Scorned by the media as little more than the drug-induced noise that wouldn't last more than a few months, in 1992 the commercial media machines adopted the Rave scene. Riding on this new wave, record labels no longer viewed it as a fad but as a cash cow. They diluted the market with a continuous flow of watered-down, mass-consumed *Rave* music. Indeed, the label's definition of Rave and its associated media popcorn representation became so overtly commercialized that the term Rave is now belittled by many.

In 1992, as a response to the commercialization of the music, two resident DJs – Fabio and Grooverider – pushed the hardcore sound to a new level. They increased the speed of records from the more common 120BPM to 145BPM. The influences of House and Techno were replaced with Ragga and Dancehall. This resulted in mixes with fast complex beats and deep throbbing bass lines. Although at this point Jungle didn't exist, this approach inspired artists to push boundaries. The tempo began to increase even more, reaching 160 to 180BPM.

Unlike today, where journalists fabricate names for genres in an attempt to earn a place in history, the name Jungle came from Duke Ellington. In the 1920s he would play music with fast exotic rhythms, and these were advertised on flyers as Jungle music. When Rebel MC sampled an old Dancehall track with the lyrics 'Alla the Junglists,' Jungle was born. The name became synonymous with music that exhibited a fast beat and a heavy bass. Pioneers of the genre such as Moose and Danny Jungle further bought this new genre to the spotlight.

Jungle enjoyed a good three or four years of popularity before it began to show a decline. In 1996, the genre diversified into Drum & Bass as artists such as Goldie, Reprazent, Ed Rush, and LTJ Bukem incorporated new sounds and cleaner production ethics. Goldie and Rob Playford are often credited with initiating the move from the Jungle sound to Drum & Bass with the release of the album *Timeless*. Jungle still exists today, but it has been overshadowed by the controlled and careful production ethics of Drum & Bass.

MUSICAL ANALYSIS

Like most genres, Drum & Bass has become hugely diversified. On one end of the spectrum, it can be pop oriented, featuring chord progressions and vocals with a 2-Step vibe, while on the other, it can exhibit a much harder edge. Nevertheless, there are some generalizations that all Drum & Bass records share that we can examine.

Typically, a Drum & Bass tempo will remain between 165 to 185BPM, although, in many cases, the tempo holds at 172 or 175BPM (Beatport often incorrectly labels them as 86 or 87BPM). The music is composed in any scale, but in particular A minor, G minor, F major, and G major are popular. The time signature is almost always 4/4.

For this chapter, we will examine the general production ethics in the creation of a *liquid*-style Drum & Bass track, but this approach equally applies to many subgenres of D&B. As music is an entirely individual, artistic endeavor, the purpose of this analysis is not to lay down a series of 'ground rules,' but, instead, will describe some of the general principles behind the production. In the end, it's up to the individual (i.e., you) to experiment and create music that suits your particular style.

In many examples, the drums consist of a 2-Step syncopation where the main snare sits on beat two and four of the bar, and further snares dance around this signature. This arrangement gives the listener a time and tempo frame, while the kick drum dances around the snare landing on and off the beat. While this may appear to provide a simple pattern, it's commonly backed up with a complex, fast paced hi-hat or percussive pattern sampled from records. Judicious application of EQ removes the kick pattern and the body of the snare, leaving only the top frequencies of the snare and upper percussion. Once the tempo is increased, this provides the fast, changing patterns typical of this genre.

The bass often plays at one-quarter or halftime of the drum tempo, again to keep the listener 'in time.' It should be noted, however, that the more complex the drum rhythm is, the faster the music will appear to be. Therefore, it's an idea to keep the loop relatively simple if the tempo is faster or relatively complex if the pace is slower.

To the uninitiated, Drum & Bass is considered to consist only of complex drum rhythms and bass but very few tracks are constructed from just these elements. Although these do play a significant role in the genre, it also features other instrumentation such as synthesized chords and textures, sound effects, and vocals.

PROGRAMMING

The prominent place to begin is by shaping the drum rhythm, and there are multiple approaches. Some producers will program the rhythms while others will source loops from old records. Although there is plenty of sample content on the market aimed towards Drum & Bass artists, only a few professional artists will use them because they may be recognized, so many prefer to rely on obscure vinyl records.

It's the breakbeat of the record that is sampled (i.e., the middle eight of the original record where the drummer plays his solo for a couple of bars). These samples are commonly sliced and rearranged in the DAW to form the basis of the *background* loop. This provides the foundation for the Drum & Bass rhythm, producing an intricate rhythmical 'live' pattern that would be difficult to program in a DAW. Although many Drum & Bass artists perform all this editing within the DAW, a number do still prefer to use the old samples such as the AKAI or Emu samplers. This is because these early devices introduce a different tonal feel that remains true to its original roots.

Because the sampled loop's function is to provide a background, it is common to remove the kick drum from the sample. The kicks in any sampled loops can be removed via the DAW, but a more common approach is to employ an EQ to roll off the bottom end of the loop. This eliminates the body of kick and much of the body of the snare just leaving the higher percussive elements to the mix. From here, a synthesized or sampled kick and snare are laid on top of this sample in a 2-Step syncopated fashion to complete the drum loop.

The kick drums can be programmed or taken from sample content. A standard approach is to sample a kick from a Roland TR909 or TR808 and pitch it down by any number of semitones to produce a thicker kick. It is preferable to sample rather than merely program at the pitch because pitching down in a sampler produces a different sonic character that is typical of Drum & Bass.

A kick drum can be created in any competent kick synthesizer via modulation of a sine wave's amplitude. Because the bass in this style is particularly low in frequencies, the kick sits on top of the bass rather than alongside it or below. Additionally, because the kick will often be pitched down after programming, a good starting point for a sine wave's frequency is around 150 Hz to 200 Hz.

Using an attack/decay envelope generator the pitch of the sine is modulated with a fast attack and short release. To help the kick remain prominent above the sub bass, it may be worth adding a square or triangle wave, pitched down with a swift amplifier attack and decay setting. This will produce a short, sharp click that will help the kick remain prominent in the mix. Alternatively, many kick synthesizers have samples of clicks that could be used.

The kick typically benefits from compression, so experiment by setting the compressor so that the attack misses the transient but grips the decay stage, by increasing the gain of the compressor it can make the kick appear more powerful.

The snares are sourced from sample content or vinyl records but, if from sample content, they are rarely taken from a Drum & Bass genre. Instead, an alternative style is chosen, and the snare is pitched up (or down) to create the requisite 'force pitched' characteristic timbre. Snares can, of course, also be programmed in a synthesizer.

Here, you could use a dual oscillator synth with a triangle wave for the first oscillator and noise for the second. The amplifier envelope generator uses a zero attack, sustain, and release with the decay employed to set the 'length' of the snare. Generally, in Drum & Bass, the snare is particularly 'snappy,' and often pitched up the keyboard to give the impression of a sampled sound having its frequency increased within the sampler.

If possible, employ a different amp EG for both the noise and triangle wave. By doing so, the triangle wave can be kept quite short and swift with a fast decay. The noise can be made to ring a little further by increasing its decay parameter.

This, however, should never be made too long since the snares should remain short and snappy.

If the snare has too much bottom end employ a high-pass, bandpass filter or notch filter depending on the type of sound you require. Notching out the middle frequencies will create a clean snare sound that's commonly used in this genre. Further modification is possible using a pitch envelope to modulate both oscillators positively; this will result in the sound pitching upwards towards the end, giving a brighter snappier feel to the timbre.

Once these basic timbres are synthesized, the kick and snare will often be programmed in a 2-Step fashion. This style features the snare on the first and third beat with further snares and kicks occurring off beat, dancing around the first and third beat in a syncopated fashion. Listen to any Drum & Bass record, and you'll experience this, you can count the first and third snares of each bar, with a kick usually occurring once or twice in between. A good start is to place the kick and snare on beats two and four, and then place a further kick a 1/8th *before* the snare on beat four. This creates a syncopated kick snare rhythm with a halftime rhythm (much like Dubstep) but with the hi-hats and ancillary percussion providing a frantic pace due to them running at 170BPM and above.

With the basic foundations of the beat, you can then add a secondary snare (with a different, brighter tonal sound) to dance around the snares sitting on beats two and four. Finally, you can to import sampled loops of hi-hats or *toppers* (these are all the ancillary instruments above the kick and snare) and time stretch them to 170BPM upwards. When these are mixed with the 2-Step rhythm of the kick and snare, it produces the typical Drum & Bass rhythms. This is the approach I took for the featured track in this chapter.

Although in almost all cases the hi-hats and percussion are sampled, they should also be augmented with further hats and percussion programmed by the producer. This helps to disguise the source of the sample. Programming these involves both hemiola and polymeter.

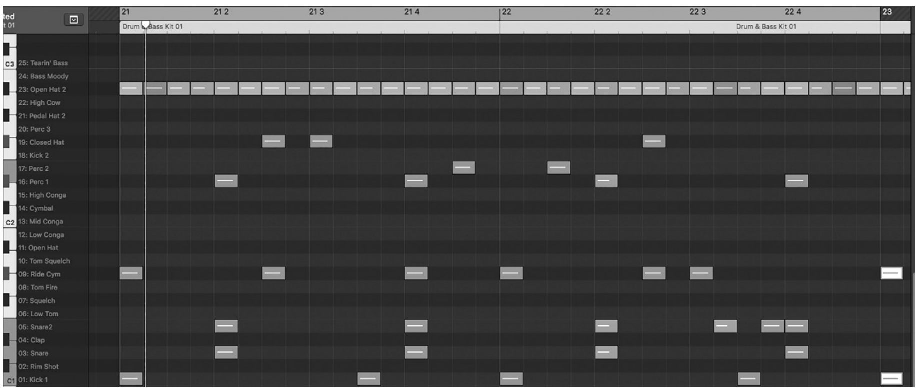

FIGURE 32.0
A basic Drum & Bass loop

Hemiola was discussed in previous chapters but consists of creating an unequal subdivision of beats within the bar and placing a hi-hat or percussive rhythmical pattern onto these unequal subdivisions. This effect is supplemented with the use of polymeter on more hi-hats, snares, or percussive elements. A typical application of polymeter here is to employ a 5/4 on a rhythmic hi-hat and 6/4 and 7/4 on additional rhythmical elements such as congas or synthesized percussive hits.

These additional hi-hats can be programmed or from sample content. In many tracks, the hats are commonly sourced from vinyl or sample content. Ancillary instrumentation is equally essential in Drum & Bass for the creation of the syncopated groove, and these sounds are again sourced from sample content or synthesized in a variety of ways. The various synthesis techniques have been discussed in details in previous chapters, and it is down to your experimentation to produce different percussive timbres that can augment the loop further.

Of particular note, Drum & Bass will also often use a call and response on the percussive elements. Here, a short conga pattern (for example) may make the call singularly or over three bars that are responded to in the second or fourth bar via a different conga pattern or a different instrument altogether.

As with most genres of EDM, each percussive instrument will be further processed with effects. The kick is commonly treated to small amounts of reverb. Here, a small room setting is often used with a long pre-delay that will skip the transient to ensure it can cut through the mix. Following reverb, distortion is usually applied. This can be applied via a distortion unit followed by EQ to sculpt the results, but some artists just use a form of distortion by pushing the mixers gain fader to its limit. In fact, this is one of the few genres of dance music where digital summing is preferred, and the artists will often run most of the faders near maximum to produce an almost crushed style of digital sound.

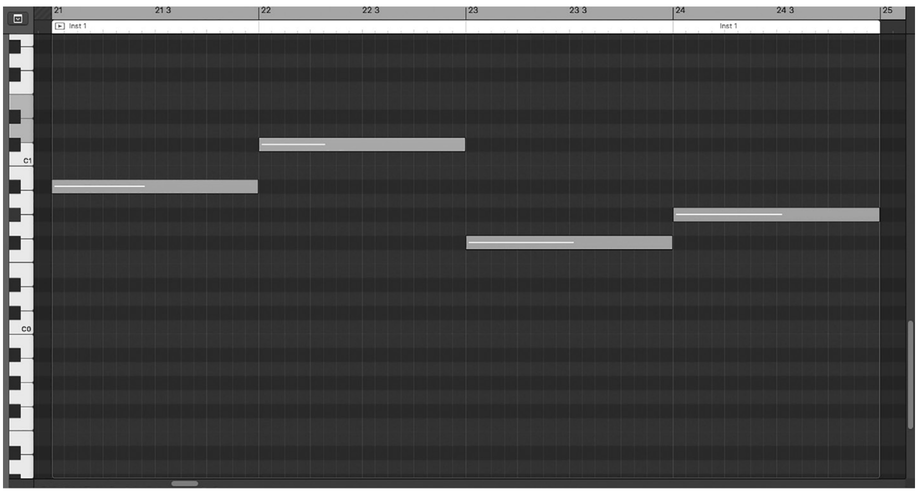

FIGURE 32.1
A secondary loop with percussive call and response answered in bar 4

Compression is applied to the kick because this can play a significant role in achieving the character of the kick. Here, the attack of the compressor should be configured, so that is skips the initial transient but captures the body and tail. The ratio is set at approximately 6:1 to 9:1 and, while listening to the kick, the threshold is reduced until the required sound is achieved.

The snares are treated with both EQ and small amounts of reverb. The reverb is typically a room style with a long pre-delay to miss the transient with a slightly longer decay to add presence to the timbre. Usually, the EQ is employed to roll off any frequencies below 800 Hz to maintain the light, snappy character of the snare.

If the loop is programmed, delay, pitch, and filter modulation are also applied to other percussive elements within the loop. The hi-hats are the first culprit to be processed, but this should be kept to short settings to prevent too much delay wash that will cloud the signal. Typically, a 1/16th or 1/8th setting with a very short delay time proves enough.

In regards to modulation of filter or pitch, the open hats can be treated to a low-pass filter that is modulated via an offset sine wave or a sample and hold waveform. This could be applied over a period of three bars while further cyclic modulation will be applied to other instrumentation over a different period of bars. Typically, odd numbered bars are chosen for the cyclic modulations because this works against the standard structural downbeat of dance music and results in a cross-syncopation of modulation.

Finally, swing quantize is applied to all of the rhythmical and percussive elements except for the kick. This is common in applications of between 61% and 71% on a 1/16th grid. After this, the loop will often be treated to some parallel compression to help the loop gel and introduce a little more energy.

Here, compression should be applied lightly, with a threshold that just captures the transients, a low ratio and a quick attack and release. If implemented more creatively, it can often breathe new life into the rhythm. With a medium threshold setting and a high ratio the returned signal can be added to the uncompressed signal, you can then experiment with the attack and release parameters to produce a rhythm that gels with the rest of the instruments.

THE BASS

The second, essential element of Drum & Bass is, obviously, the bass. This may consist of notes that play in either one quarter or half the tempo of the drum rhythm. This is accomplished by using long bass notes set at one quarter, half, or full notes and sometimes straddling over the bars of the drum loop. The notes of the bass usually remain with an octave and rarely move further than this to prevent the music from becoming too active and deterring the listener from the rhythmic interplay of the drums.

Many tracks will exhibit a slow movement in the bass to counteract the speed set by the drum rhythm. If the bass notes are kept quite lengthy, it is not uncommon to employ some change in the bass to maintain interest. You can do this via the use of filters or pitch modulation.

In the above example (the book's example), each bass note was a bar long, using the notes A#, C#, F#, and G#. The bass was a mix of a sine wave from the EXS24 sampler and a distorted triangle wave from Lennar Digital Sylenth. The triangle wave was played at the same notes as the EXS24 but was detuned in Sylenth by a semitone so that the two bass lines resulted in a form of constructive and deconstructive interference.

Because the bass in this genre is supposed to remain deep and earthshaking, a triangle wave is commonly the perfect starting point. You can use a single oscillator set to a triangle wave and positively modulate its pitch with an attack/decay envelope, then experiment with the synthesizer's amplitude envelopes. As an example of a typical Drum & Bass timbre, using a triangle wave, set the amplifier's attack to zero and increase the decay setting while listening back to the programmed motif until it produces an interesting rhythm. Next, modulate the pitch by a couple of cents using an envelope set to a slow attack and medium decay. This will create a bass timbre where the note bends slightly as it's played. Alternatively, you can use an LFO set to a sine wave with a slow rate and set it to start at the beginning of every note.

Experimentation is the key, changing the attack, decay, and release of the amp or/and filter EG from linear to convex or concave will also create new variations. For example, the decay to a curved slope setting will produce a more rounded bass timbre. Similarly, small amounts of controlled distortion or very light flanging can also add movement.

Wavetable basses are also favorites in this genre. Synthesizers such as Xfer Serum, Reveal Sound Spire, and Native Instruments' Massive can all be used to

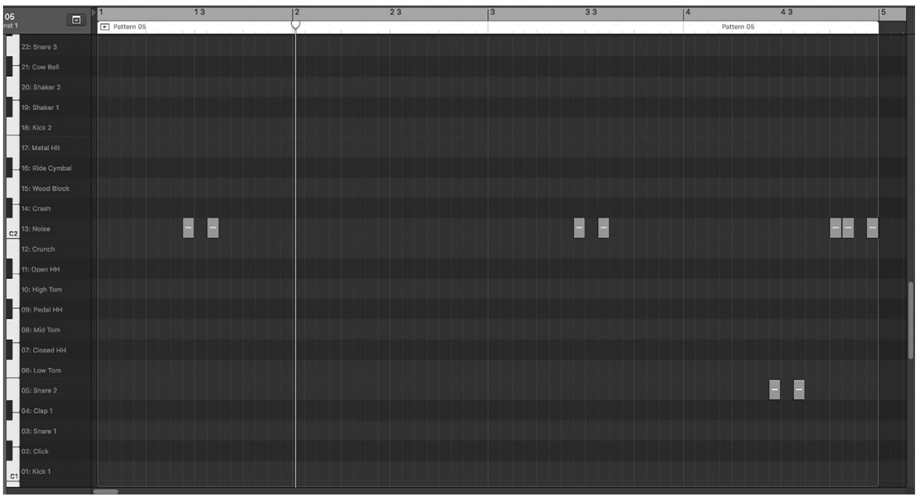

FIGURE 32.2
The bass melody in the example track

significant effect. You can pick out a couple of interesting sounding wavetables, mix them together, and then modulate the results. The amplitude envelope settings would be the same as for the analog counterpart, but popular modulation options are envelopes and LFOs to the wavetable selection (scanning), filter, and pitch.

Whether using wavetables or analog synthesis for the bass, it is important to add further low-frequency elements to underpin the record and provide sub tones. The favored synthesizer for this is Logics EXS24 sampler, because when no sample is loaded it defaults to a powerful sub-sine wave. However, you can use the sine from Sylenth or Omnisphere to create the sub. Here, the amplifier attack should be immediate, and the decay setting is then increased until it fills the lowest frequencies of the music and sits underneath the previous bass.

As the rhythmic movement and interaction with the bass and rhythms provide the basis for this genre, it's also worth experimenting with applying effects to the bass timbre. While some effects should be avoided since they tend to spread the sound across the image, in this genre the bass is one of the most critical parts of the music so small amounts of delay can create exciting fluctuations, as can flangers, phasers, and distortion. If a delay or any further stereo spreading effects are applied, however, it is advisable to employ a mid/side processor beforehand to ensure that the effects only occur above 120 Hz, leaving any frequencies below this as a mono source.

As with the drum rhythms, creative compression can also help in attaining an interesting bass timbre. As before, try accessing the compressor as a send effect with a medium threshold setting and a high ratio. The returned signal can be added to the uncompressed signal; you can then experiment with the attack and release parameters to produce an interesting bass tone. Alternatively, try pumping the bass with one of the rhythms. Set the compressor to capture the transients of a percussive loop and use it pump the bass by experimenting with the attack, ratio, and release parameters.

CHORDS

Some aficionados of the genre recommend using only the minor chords from the chosen key as a harmony. This means that while it is accepted that Drum & Bass is written in a minor key, but the chords of III, VI, and VII (in the natural minor) or III, V, and VI (in harmonic minor) are avoided. This does, of course, mean that the chords in a harmonic minor cannot cadence since the V is a major chord in the harmonic minor. However, this is entirely open to artistic interpretation.

Also, the chords can often work well when working in contrary with the bass line. Copying the bass line down to another workstation channel track and then converting this new track into a series of chords is an easy way to start. Once created, if when the bass moves up in pitch, move the chords down in pitch and vice versa.

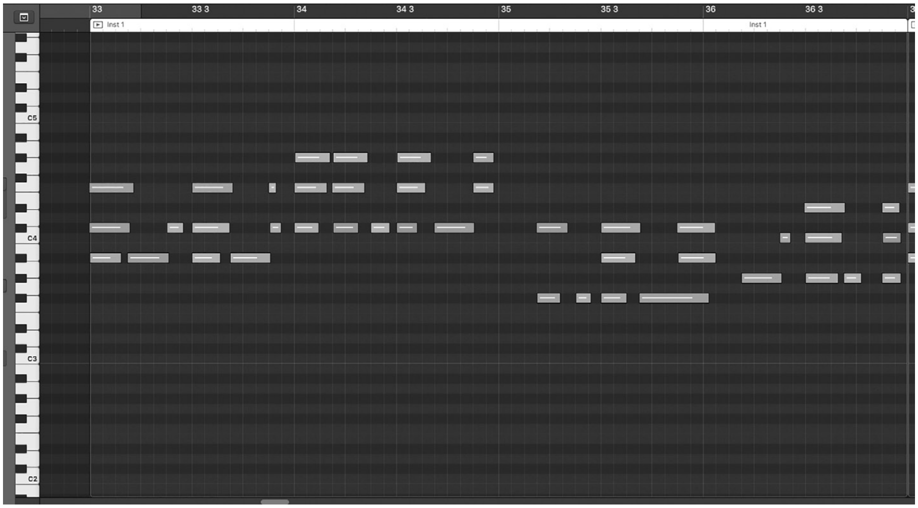

FIGURE 32.3
A piano chord progression used in the tack

With a general idea of the chord structure down, you can program (or sample) a string to fit. Strings are often more popular than pads since these are often particularly heavy in harmonic structure, and therefore take a good proportion of the mixes frequency range.

The most popular source for strings and pads is Spectrasonics' Omnisphere 2, but if you want to program your own, they can be created by mixing a triangle and square wave and detuning one of the oscillators from the other by three to five cents. The amplifiers envelope is set to a zero attack with medium sustain, and release. Using the filter envelope set it to a long attack, sustain with a medium release and short decay. Finally, adjust the filter cut-off quite low and the resonance about mid-way and modulate the pitch of either the triangle or square wave with a sine wave set to a slow rate with medium depth. If the string is going to continue for a length of time, it's worthwhile employing a sine, pulse, or triangle wave LFO to modulate the cut-off of the filter to help maintain interest. As always, this should only be considered as a starting point, and experimentation is the key to gaining good results.

Effects can also play an essential role in creating interesting strings for the genre, although these should be used conservatively so as not to detract from the bass rhythm. Often, extensive chorus effects, rotary speaker simulations, flangers, phasers, and reverb can all help to add a sense of interest. Alternatively, creatively pumping the string by running it through a compressor with a side-chain input channel programmed with a rhythmical element can breathe life into static-sounding timbres.

Alongside chords, some tracks may feature pianos, guitars, horns, or string sections to provide memorable motifs. Others will simply employ arpeggio patterns to reaffirm the fast tempo of the music. Almost every DAW features an arpeggio plug-in, so it is merely a case of feeding the chord into the arppegiator and recording the MIDI output. You can then synthesize a timbre using either wavetable or analog synthesis. These should employ high harmonic content with a fast attack and a release that is rapid enough to end the note before the next begins. The low frequencies of the sound should also be removed via a high-pass filter so that the sound doesn't conflict with the heavy bass lines.

VOCALS AND SOUND FX

One final characteristic is the addition of sound effects and vocals. The vocals in Drum & Bass can range from a complete vocal track to short vocal chants. It is a point of contention among many Drum & Bass producers as to whether a verse/chorus is part of the genre, or is, in fact, diversifying again to produce a new style of music (some refer to this as *liquid* Drum & Bass). Others, however, believe that it's merely a watered-down, commercialized version of the music. Nonetheless, whether you choose to use a few snippets of vocals, some Ragga, or MCing, or a more commercialized vocal performance is entirely up to you. It's the musicians who push boundaries who reap the most significant rewards. For the example track, I asked Talson Moon to provide the lyrics, which were recorded on a Nuemann U87 through a Great River MP500NV and compressed via a LA2A compressor.

Sound effects can naturally be generated by whatever means necessary, from sampling and contorting sounds or samples with effects and EQ. For deforming audio, the Sherman Filterbank 2, the Camelspace range of plug-ins, Glitch, Sugarbytes Effectrix, or Steinberg's GRM tools are almost a requisite for creating strange evolving timbres.

That said, the effects and processing applied are, of course, entirely open to artistic license as a result is to create anything that sounds good and fits within the mix. Transient designers can be especially useful in this genre as they permit you to remove the transients of the percussive rhythms that can evolve throughout the track with some thoughtful automation. Similarly, substantial compression can be used to squash the transient of the sounds, and, with the aid of a spectral analyzer, you can identify the frequencies that contribute to the sound while removing those surrounding it. Alternatively, pitch-shift individual notes up and by extreme amounts or apply heavy chorus or flangers/phasers to singular hi-hats or snares or try time-stretching followed by time compression to add some digital clutter and then mix this with the other loops.

Ultimately, though, as with all the chapters in this section, the purpose here is to merely reveal some of the techniques used by producers and show how the theory and technology discussed earlier in the book combine to produce a track. However, it should be viewed as offering a few basic starting ideas that *you* can evolve from.

There is no one definitive way to produce this genre and the best way to learn new techniques and production ethics is to actively listen to the current market leaders and experiment with the tools at your disposal. There are no right or wrong ways to approach any piece of music, and if it sounds right to the producer, then it usually is. New genres do not evolve from following step-by-step guides or from emulating peers; producers who experiment and push boundaries create them.

Nevertheless, with just these essential elements, it is possible to create the primary focus of the music, and from here the producer can then look towards building an arrangement. The theory behind arranging has already been discussed in an earlier chapter but merely listening to the current market leaders mixed among the method discussed in the previous chapter will very quickly reveal the current trends in both sounds and arrangement.

DMM resource: The companion files contain a short excerpt of ‘State of Mind’, a Drum & Bass track released under the name Neurokode, using the techniques described above.

Track 1 is the excerpt played through GM instruments.

Track 2 is the excerpt with full production.

‘State of Mind’ © Neurokode (R. Snoman) ft. Talson Moon.

Lyrics by Tom Larkin.

CHAPTER 33

An overview of Dub

Dub music is like a long echo delay, looping through time. Turning the rational music order into an ocean of sensation.

David Toop

It is probably wise to begin this chapter with another quote, this time from Chris Blackwell who famously said that: 'There are no facts in Jamaica.' This is because the roots of Dub, a subgenre of Reggae, is complicated to define accurately. There are many contradictory accounts of who was responsible for the genesis of the Dub movement, and how they were involved but one thing for sure, without Dub, many of the dance music genres of today simply would not exist.

For the majority, Osbourne Ruddock (aka King Tubby) alongside Lee 'Scratch' Perry, Errol Thompson, and Herman Chin Loy are considered the pioneers of the Dub genre. Back in the 1960s, King Tubby was a budding electrician who would build radios from discarded parts and also service amplifiers for sound systems. He was also an aspiring DJ, who ran a pirate radio station for a short period before eventually forming *Tubby's Hometown Hi-Fi* in 1968. More importantly for the history of Dub, however, he also maintained the studio equipment at Duke Reid's Treasure Isle studio.

There are conflicting reports that Byron Smith, a resident engineer, was asked by Ruddy Redwood to remove the vocal track for a vinyl test pressing or that Byron accidentally left the vocals out. Whatever the case, when Ruddy heard the vinyl test pressing (known as a *dubplate*), he loved the sound so much he played it at his next event with Deejay Wassy toasting over the music. It proved to be a huge success, and, if the legend is to be believed, Byron either informed King Tubby of the success while he was servicing the equipment, or King Tubby was present at the event and witnessed the success himself. Contrary to many published articles Tubby was not involved in the original track, neither was he ever an apprentice to Byron.

With the knowledge gained from servicing at the studio, Tubby began to create instrumental tracks himself. Initially using a 2-track tape machine to create new

Dub mixes, in 1971 he managed to source an MCI mixing desk. Armed with this desk, he could experiment to push his new 'Dub' sound further by employing effects such as reverb and delay. He turned his room into a remixing studio and would perform more than merely removing the vocals from tracks. Instead, he would mix records in new and inventive ways, playing the rhythms on their own, allowing DJs to toast over them, before later introducing the vocals from the same track. He would also add lengthy reverb and delays to these mixes, creating new 'versions' that would feature as B sides of many records.

As Tubby's new techniques became more widespread, producers such as Lee 'Scratch' Perry, Keith Hudson, and Augustus Pablo would visit his remixing studio for their records to receive the *Dub* treatment. Eventually, Tubby became so popular that he had to convert one of his bathrooms into a vocal booth.

It wasn't until 1973 that Dub was truly recognized as a genre in its own right. While Herman Chin Loy and Errol Thompson, a production team from Aquarius studio, realized that Dub had commercial potential, it was the release of *Black Board Jungle* by the *Upsetters* that was the first landmark recording for the genre. Produced by Lee 'Scratch' Perry and King Tubby, only 300 copies were initially pressed for distribution throughout Jamaica, but it was later rereleased as *Blackboard Jungle Dub*, with a slightly different track listing. Some purists argue that an album released three years earlier, Derrick Harriot's *The Undertaker* was the first Dub album, but this should be considered as instrumental Reggae rather than Dub.

Since its inception in the 1970s Dub remained mostly as a singular genre. Although some consider it as a subgenre of Reggae – and undoubtedly partly responsible for the sound of Jungle – it was another ten years before it began to fragment. Today, with the introduction of synthesizers there has been a new emergence of Dub with genres such as Dubtronica, Techdub, Ambient Dub, Dubstyle, Dubstep, and, more recently, Electrodub. The last is still in its infancy, mixing the bass lines of dubstep with the Ska strokes and toasting of Reggae and Dub.

MUSICAL ANALYSIS

The focus of all forms of Dub lies with 'the riddim.' This is the Jamaican patois for rhythm. There is a significant emphasis on the sub bass, surrounded by halftime drums with syncopated dancehall style hi-hat patterns and a sparse arrangement. As Dub is an offshoot of Reggae, it also typically features offbeat 'Ska' guitars and piano chords, overlaid with an MC toasting.

It is mostly composed of a 4/4 signature with a tempo of 100 to 110BPM. However, due to the halftime nature of the drums, it often results in the music appearing to be half that tempo. Notably, producers of the more recent offshoots, such as Electrodub, often increase the tempo slightly, working at 120 to 140BPM.

The music is composed of both major and minor scales, although E and B minor can be particularly useful because the fundamental tones of the root are close to 20 Hz and 30 Hz. Although these frequencies cannot be heard, Dub relies on *feeling* the subsonic frequencies as well as hearing them.

For this chapter, we will examine the general production ethics in the creation of an Electrodub track, but this approach applies equally to many genres of Dub. As music is an entirely individual, artistic endeavour, the purpose of this analysis is not to lay down a series of 'ground rules,' but, instead, will describe some of the general principles behind the production. In the end, it's up to the individual (i.e., you) to experiment and create music that suits your particular style.

The best starting point for the creation of Dub is with the drum rhythms. The genre often employs a one-drop rhythm where the kick only lands on the third beat of the bar while the snares and hi-hats move around this singular kick. A typical arrangement would be the kick to occur on the third beat of a bar, the snare on the second beat, and hats are happening on beats one and four. With Electrodub, and in the track 'See No Light' that I collaborated on with *Dubject* (one of the producers pushing this genre), for the start of the record, the kick occurs on the first beat of the bar, with a snare on the third and closed hi-hats on the second and fourth beats. This introduces a halftime rhythm to the music that belies its much higher tempo.

Notably, some Electrodub will lean more towards dubstep with regards to its rhythmical patterns. Here, a snare may occur a couple of times in the bar, one on the beat, while a second occurs offbeat. This is usually augmented by a kick that dances around these snares, supplemented by further triplet hat rhythms.

In Dub, the kick is often thick and heavy so that it can also act as a secondary bass, but it will feature a bright transient stage to cut through the sweeping sub-sonic bass instruments. These styles of kick are available on the multitude of sample content, but many producers will build their own in kick in synthesizers.

A good starting point for a Dub kick is a sine wave at around 250 Hz, sweeping down with positive pitch modulation to about 20 to 25 Hz. The pitch envelope is typically linear, but the amplitude envelope employs a longer attack so that

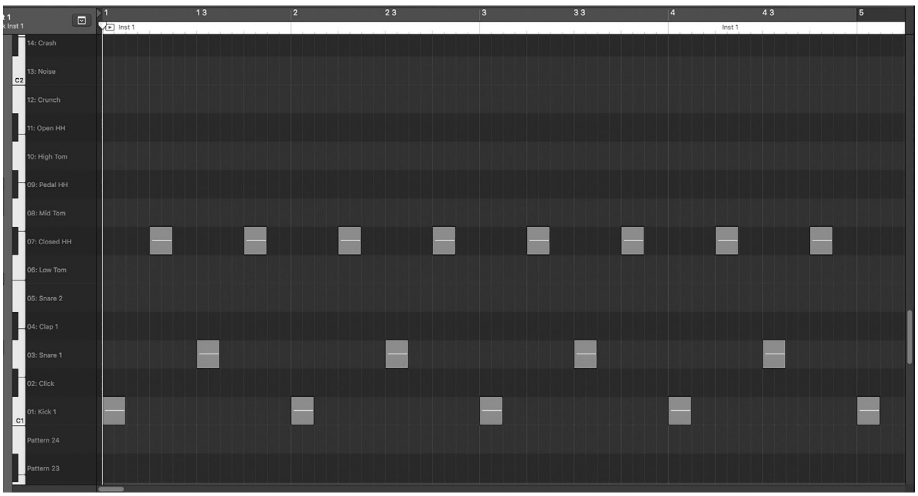

FIGURE 33.0
The primary drum pattern for Electrodub

the transient of the kick is muted, producing more of a sweeping bass than a kick timbre. The transient is replaced with a bright sample, such as a hi-hat or you could use square wave with a fast amplifier attack and decay setting. If pitched correctly, this produces a short, sharp click that when layered above the kick, helps to increase the transient edge to cut through the mix.

The kick is usually left completely dry, but it can be treated to small amounts of reverb depending on the genre. A room reverb with a long pre-delay so that it bypasses the transient combined with a short tail is a good starting point. Dynamic compression can also help achieve a solid tonal character that is typical of the genre. A good starting point is a ratio of 5:1 with a fast attack so that it bypasses the transient but captures both body and tail. Then you can reduce the threshold until the kick begins to punch.

The snares are always short and snappy and sometimes layered to create a hard transient. Snares are commonly sourced from sample content and then equalized to the producer's taste. Typically, the low frequencies are removed, and a small boost is applied to 3 or 5 kHz to bring out the snappy nature required for this genre. If a transient layer is needed, this is usually from a TR909 or TR808 and treated to processing and effects. Typically, this involves compression with a fast attack so that the compressor captures the transient and a 4:1 ratio to pull down the transient and produce a harder timbre. This is often followed by a plate reverb to add a metallic aura to the snare.

In some examples of Electrodub, if two snares occur on the same bar, then a secondary, different snare timbre is used. In 'See No Light,' the track starts with a single snare per bar, but as it progresses and the riddim become more complex, two snares occur in the same bar. Consequently, we used two different snares. Both these were sourced from sample content, equalized and then treated to delay. Dub relies on careful application of delay and snares are always treated to delay. Typically, these are dotted 1/8th or 1/4 note delays depending on the tempo and style.

The closed hi-hats in Dub often sit in a locked pattern, occurring on the beats that do not contain either a kick or a snare. In Electrodub and Dubtronica, however, they often exhibit more of a rhythmical dancehall feel by employing triplet patterns that change over the course of the four-bar pattern.

These rhythmical patterns of the hi-hats are down to creativity, but they should feel as if they've been played live, occurring off the beat with different velocities, timing variations, and a syncopated feel. Constant 1/16th patterns are avoided because these reveal the faster underlying tempo and also add a mechanical feel to the music. However, some forms of Electrodub play hats on every 1/8th to increase the speed of the piece, while further hats dance around these constant notes.

The closed hi-hat timbres often exhibit a bright, distinct tone. They will typically alternate between two different timbres or have a strong form of pitch or filter cyclic modulation applied so that it becomes evident. Typically, hi-hats will be taken from sample content and treated with numerous effects and processors such as reverb, compression, and sometimes delay. Possibly the best

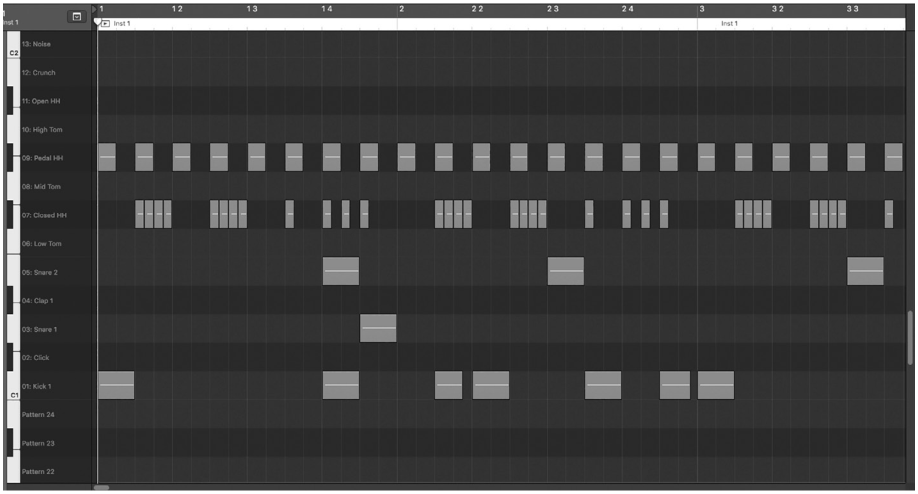

FIGURE 33.1
The more complex rhythm introduced later in the track

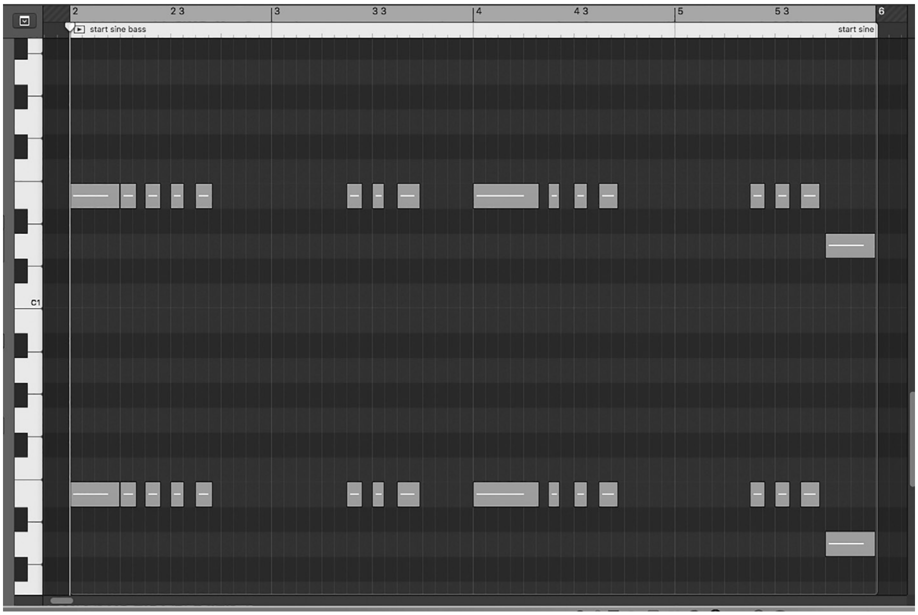

FIGURE 33.2
The starting bass in 'See No Light'

source for hats in these genres is to use shakers, placed into a sampler with the attack increased. Or the sample can be used directly in the DAW with a transient designer employed to reduce the hats' transient.

Beyond these instruments, further ancillary percussive instrumentation is rare. Many artists prefer to leave the beats sparse to maintain the slower paced feel of the music. Occasional percussive instruments may be used, but these will be spaced at every four, eight, or 16 bars and often only consists of a single percussive strike that will ring out for the duration of a bar.

Electrodub will often employ a secondary bass borrowing from the sound design techniques of Dubstep. This secondary bass will either layer over some notes of the original bass, or play in between to create a call and response action. This bass will cyclically modulate in frequency during its duration. Indeed, the bass in many tracks will remain at the same melodic pitch in the workstations piano roll, and it's the cyclic movement of the filter that creates the interest.

THE BASS

The thick, low, subsonic bass is fundamental to all forms of dub. In many cases, the bass is a recording of a real bass guitar. However, to achieve the blunt bass tone that Dub is synonymous with, the electric bass should use flatwound or upright strings. The playing style also contributes heavily to the Dub sound and is usually played consistently with the thumb on the low A and E up the neck. The bass is usually recorded via DI (recorded directly into the DAW), but a secondary signal is also recorded by mic'ing up a bass amp. In both instances, the signal is commonly patched through a dbx Subharmonic Synthesizer to boost the low energy detail.

An alternative to recording a real bass guitar is to use sample content. There are many multi-sampled bass instruments available for Kontakt, and with some judicious EQ and compression, they can produce the right style of tone. For the electronic genres of Dub, you can program a simple bass on most analog style synthesizers. Nothing more than a triangle of square wave played low on the keyboard with low-pass filter reducing most of the high-frequency harmonics will produce a deep tone.

Alternatively, sampling a 60 Hz sine wave via a 12-bit or 8-bit sampler and layering it over another 60 Hz sine wave is a commonly used technique. Some artists will also sample the kick from a TR909 or TR808, time-stretch it and then filter the results to produce a thick bass timbre. Whatever method is used, if two of these notes are played an octave apart, it provides a deep bass that can be heard on both subsonic systems as well as small speakers.

For the filter to sweep the sound, they must be rich in harmonics. Many producers will use wavetable instruments such as Xfer's Serum or Native Instruments' Massive. Here, you can choose any interesting wavetable that is rich in harmonics and can be swept via an LFO on the filter. Typically, the amplifier envelope should employ a fast attack with a short decay, high sustain, and no release. A second envelope set to a fast attack with a medium decay, no sustain, and an extended-release can be used to modulate a low-pass filter. Here, the cut-off should be set about half-way and the resonance should be set around a quarter.

This will produce the necessary bright harmonically rich timbre, but these harmonics should be swept with a filter, or the wavetable should be modulated via an envelope. If using a filter, bandpass with a low cut-off and a resonance set mid-way is a good starting point. The filter can be tempo synced and modulated via a triangle LFO at a 1/4 triplet of the tempo of the sequencer.

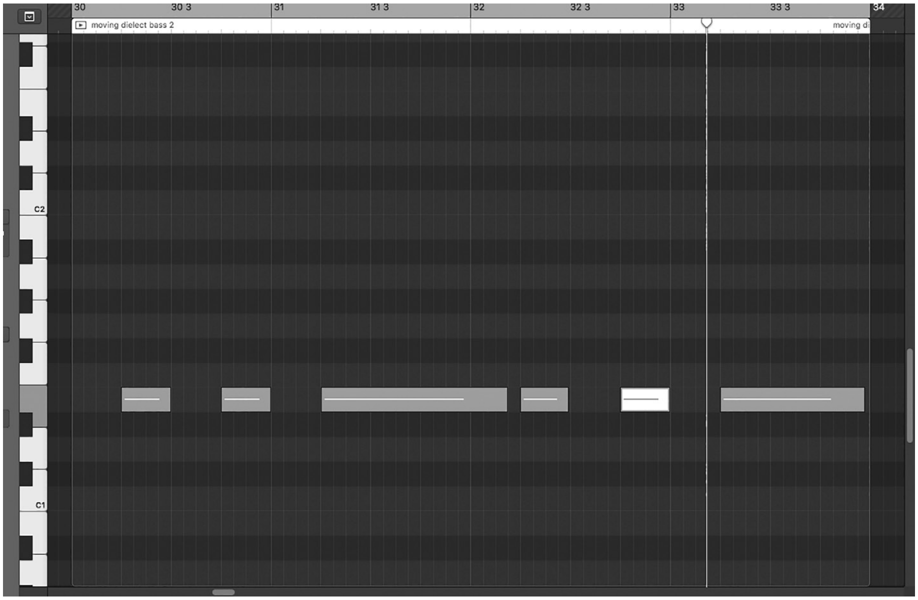

FIGURE 33.3
The layered 'dubstep' bass in 'See No Light'

The rhythmic movement and interaction with the bass and rhythms provide the basis for this genre, and therefore it's also worth experimenting by applying effects to this bass timbre. While some effects should be avoided since they will spread the sound across the image, in this genre, the bass is one of the most critical parts of the music so small amounts of automated distortion can create impressive effects.

As with the drum rhythms, creative compression can also help in attaining an interesting bass timbre. As before, try accessing the compressor as a send effect with a medium threshold setting and a high ratio. The returned signal can be added to the uncompressed signal; you can then experiment with the attack and release parameters to produce an exciting bass tone. A conventional technique here is to send the transient snare and kick to a bus (0 dB) and then use this bus as a side-chain for a compressor placed onto the bass instruments.

The settings to use for the compressor depend on the effect you wish to accomplish, but a good starting point is a fast attack with a 50ms release and a ratio of 2:1. Once applied, lower the threshold and increase the make-up gain until the effect is evident.

CHORDS, MELODIES, AND EFFECTS

There are rarely any melodies in any form of Dub, and, instead, they all borrow the *Skank* from Reggae. The *Skank*, sometimes known as *Ska stroke*, is a guitar strumming technique involving playing a short, chopped chord on the second and fourth beats of the bar.

If you want to replicate this style of sound, you should play on the down-stroke (*not* the upstroke like most recommend), and dampen the strings with your fingering hand. Typically, the chords are minor, and often they consist of nothing more than a minor third interval. If you play only the top four strings and run the results through a wah pedal with a boost at 2 kHz, it will produce the necessary Skank timbre. If you can't play (or record) a guitar, there is some sample content available of Skank guitars (although it's not as varied as many other genres).

The guitar is typically treated to a small amount of room reverb and delay. This is naturally applied to taste, but, as a starting point, use a room reverb with a two-second release time and a pre-delay long enough to bypass the transient. For delays, ping pong is best with a 1/8th on the right and 1/4 on the left and short feedback to achieve the classic Dub sound.

To complement the Skank, many tracks employ a piano. This may be used to underpin the Skank by playing the lowest note of the guitar chord, or it may play in between to produce a syncopated rhythm to offset the guitar. If the piano is underpinning, then it should become a layer rather than an instrument in its own right so it should be equalized to layer with the guitar. In 'See No Light,' the piano was used to offset the rhythm of the guitar and create a syncopated feel to the music.

The piano is typically played as a minor chord, like the guitar, and is rarely taken from a real piano. In many instances, the piano is a poor emulation from a synthesizer such as the Korg M1 or the Yamaha DX7. This is treated with EQ to remove all frequencies below 200 Hz and above 16 kHz, and is often treated with effects such as reverb, phasers, chorus, and delay. In the example track 'See No Light,' we used a sample of a piano that was EQd heavily and treated to some delay and reverb.

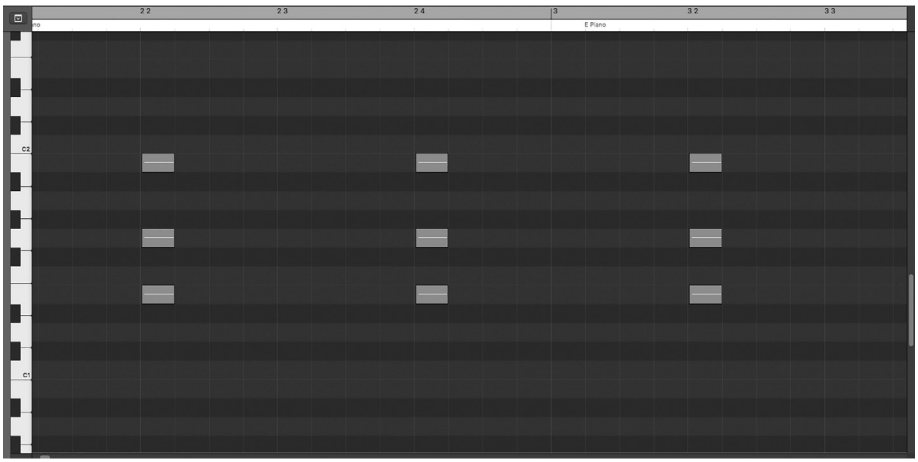

FIGURE 33.4
The Skank chord in 'See No Light'

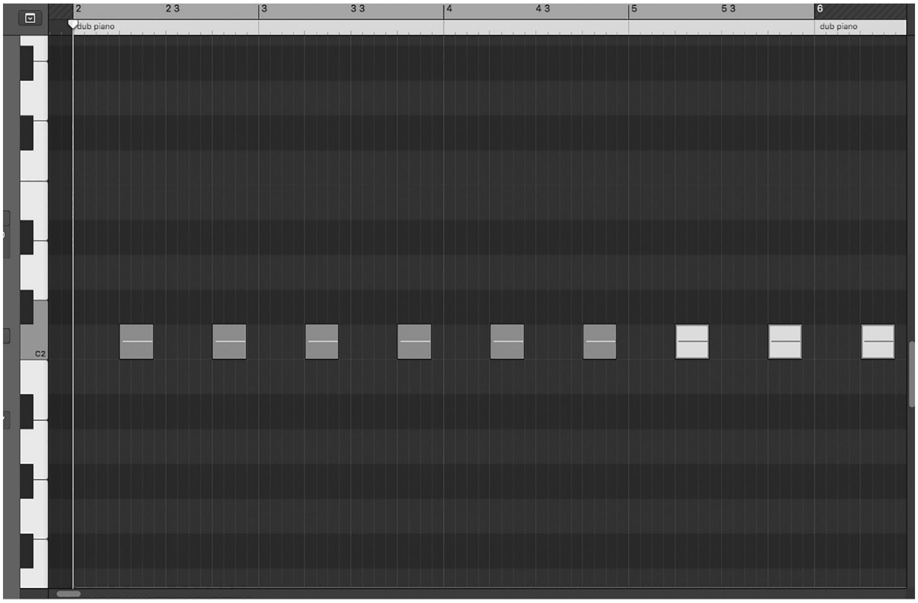

FIGURE 33.5
The offset piano in 'See No Light'

These primary instruments form the foundation of all Dub genres and provide the groundwork for the MC to toast over. From here, it's a simple case of employing risers and falls, alongside small flourishes to move from one part of the track to another. These techniques have all been discussed in detail elsewhere in this book, so I won't cover the same ground again here.

Of course, you can push these basic ideas further. Sound effects can be generated by whatever means necessary, from sampling and contorting sounds or samples with effects and EQ. For twisting audio, the Sherman Filterbank 2, the Camelspace range of plug-ins, Glitch, Sugarbytes' Effectrix, or Steinberg's GRM tools are almost a requisite for creating strange evolving timbres.

Transient designers can be especially useful in this genre as they permit you to remove the transients of the percussive rhythms that can evolve throughout the track with some thoughtful automation. Similarly, substantial compression can be used to squash the transient of the sounds, and with the aid of a spectral analyzer, you can identify the frequencies that contribute to the sound while removing those surrounding it.

Ultimately, as with all the chapters in this section, the purpose is to merely reveal some of the techniques used by producers and show how the theory and technology discussed earlier in the book combine to produce a track. However, it should be viewed as offering a few basic starting ideas that *you* can evolve from.

There is no one definitive way to produce this genre. The best way to learn new techniques and production ethics is to actively listen to the current market leaders and experiment with the tools at your disposal. There are no right or wrong ways to approach any piece of music, and if it sounds right to the producer, then it usually is. New genres do not evolve from following step-by-step guides or from emulating peers; producers who experiment and push boundaries create them.

Nevertheless, with just these essential elements, it is possible to create the primary focus of the music, and from here the producer can then look towards building an arrangement. The theory behind arranging has already been discussed in an earlier chapter but merely listening to the current market leaders mixed among the method discussed in previous chapters will reveal the current trends in both sounds and arrangement.

DMM resource: The companion files contain an excerpt of 'See No Light,' an Electrodub track I collaborated on with Dubject, using the techniques described above.

Track 1 is the excerpt played by GM instruments.

Track 2 is the excerpt with full production.

'See No Light' © Dubject.

CHAPTER 34

Mixing theory

For me, electronic music is like cooking: it's a sensual organic activity where you can mix ingredients.

Jean-Michel Jarre

Mixing electronic dance music is the last process in an extended chain of religious production ethics. Provided time has been taken throughout the project, mixing is perhaps the most straightforward and satisfying activity in the entire production. However, if the project has suffered from poor choices and decisions throughout, this will become the most time consuming and difficult.

As I hope the chapters so far have communicated, producing great dance music is not a result of relying on presets, randomly introducing processors or effects, or accepting second-rate results. When it comes to a great song, good enough isn't.

The difference between an amateur and a professional is that the professional will spend a seemingly disproportionate amount of time on the smallest and most insignificant of details. While the choice and application of a specific tube compressor for a bass, for example, may appear to impart a sonic feature so small as to be insignificant, it is these small additions that add up to a much greater whole.

Every instrument should be programmed, recorded, and processed one after the other with forethought and care, and, most importantly, with a vision of how the final record will sound. If this has been the goal, then the mix amounts to little more than positioning instruments across a virtual soundstage. If your production doesn't sound close to how you envisage the final track to sound, then you need to go back and fix the problem. There is no 'fix it in the mix.' Mixing is not a complicated process where the magic happens. The magic happens during the production and mixing is merely the cleanup.

However, even if careful choices have been made throughout, the ultimate question you must ask yourself is: *Can you feel it?*

Above the theory, musicality, sound design, programming, processing chains, effects, and the arrangement, dance music is ultimately about the vibe and *feel* of the music. As Simon Cowell's freak-mangling TV shows have demonstrated, music can be well produced, but the results are usually always flat and tepid. Dance music *must* exhibit feel above all else. It's the fundamental basis of this genre, and if *you* can't feel it or dance to it, you cannot expect anyone else too.

So before moving on, you must be able to answer the following with a resounding yes:

- Does it sound great?
- Can you feel it?
- Can you dance to it?

MIXING THEORY

It is essential to understand that mixing is a creative endeavor and there are no right or wrong ways to approach it. Unless you're completely tone deaf and have never listened to music – ever – it is difficult to create a complete shambles of a mix. Provided care has been taken over every other production aspect of the music; then at this point, the track is usually already mixed. This stage is little more than perhaps some positioning of instruments, and so a unique style of mixing will define your style as much as every other aspect of production.

The central aim of any mix is to achieve transparency and elicit an emotional response in your listeners. The latter is impossible to quantify but, for the former, each instrument should occupy a space, so it is heard clearly. To accomplish this, we first must first to understand a sound stage is and some of our hearing limitations.

Human hearing is fallible. Not only do we perceive different frequencies to be at different volumes but also the volume at which we listen to music will determine the dominant frequencies. We are more receptive to frequencies occupying the midrange when listening to music at conversation level. If the volume of the music is increased beyond this level, then the lower and higher frequencies become (albeit perceivably) louder than the midrange.

Harvey Fletcher and Wilden Munson from Bell Laboratories were the first to document this in 1933. Using many test subjects, they played a series of sine waves at differing frequencies to a listener. For each frequency, the listener was subjected to a secondary reference tone at 1000 Hz. The gain was adjusted until it was perceived to be at the same loudness as the previous sine wave. The results were averaged out and the results published as the 'Fletcher Munson contour control curve.'

As shown in Figure 34.0, the equal loudness contour is measured in Phons. This unit of measurement can express how sine waves of differing frequencies and gain can be perceived to be as equally loud as each other. Of particular note, later experiments have changed these results because Harvey and Wilden used

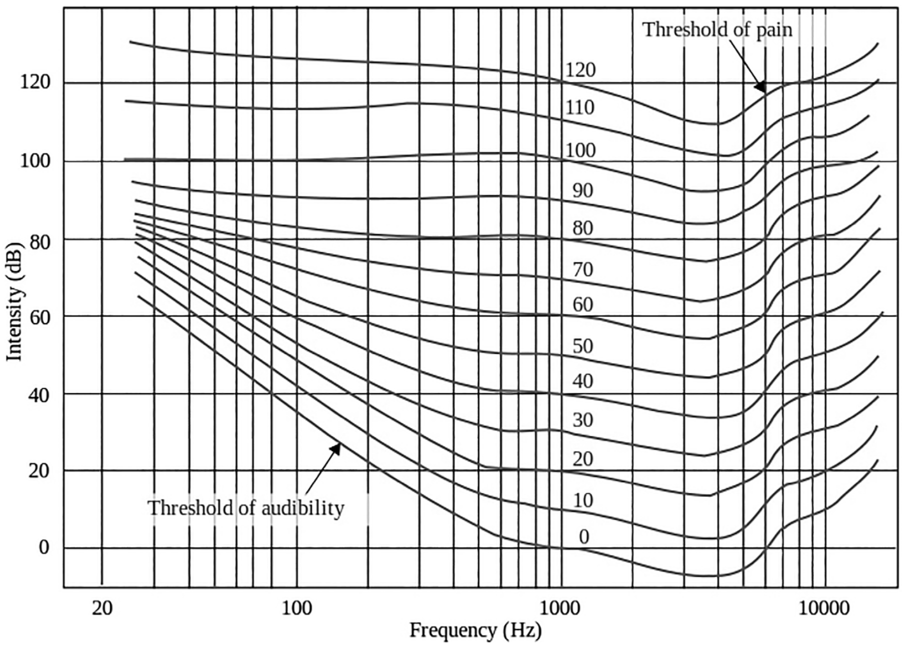

FIGURE 34.0
The equal loudness contour measured by Fletcher Munson

headphones in their tests and our hearing behaves differently with headphones on than when we listen with speakers. This has already been discussed earlier in the book, but as a refresher, when we are subjected to sounds, their position influences how we perceive frequencies. This effect is known as **H**ead-**R**elated **T**ransfer **F**unction (HRTF).

If a sound is directly in front of us, then both ears receive the signal at the same intensity until it reaches the resonant frequency of the outer ear. Here, at around 1 kHz, the sonic reflections change and we perceive this as both a frequency and volume change. Also, if the sound occurs off center, then your head gets in the way and absorbs some of the frequencies before it reaches the other ear. This results in a different frequency and gain and thus perceived loudness.

While the equal loudness contour may be incorrect, our perception *does* change with volume. So we must consider this when we approach a mix. If the bass elements are balanced at a low monitoring level, then there will be a perceived bass increase at higher volumes. Conversely, if the music is mixed at high levels, there may be too little bass at lower volumes.

Mixing appropriately for all three volumes will always be a trade-off. You'll struggle to find the perfect balance for all listening levels, and therefore the best approach is to mix just above normal conversation level. This offers two benefits; it gives you more time to mix before you suffer ear fatigue and it also provides a proportionately balanced mix.

THE SOUNDSTAGE

For a successful mix, the instruments should be placed on a virtual soundstage for them all to be heard clearly. To do this, we can envisage a three-dimensional box – the soundstage – on which you position the instruments.

Sounds placed on this virtual stage can be positioned anywhere between the left or right 'walls' using the pan parameter. They can be positioned at the front or back, or anywhere in between with gain. And the frequency content will determine whether it sits at the top of the stage (high frequencies), the middle (mid-range frequencies) or the bottom (low frequencies).

Using this as a template, the concept behind approaching a mix is to ensure that each sound occupies its unique space within this stage so that it can not only be heard but so that it also fits well with everything else. To do this, we must subdivide the soundstage into three distinct areas: the front to back perspective, the horizontal perspective, and the vertical perspective.

Front to back perspective – gain

One of the primary auditory clues we receive about the distance we are from any sound source is through the intensity of air pressure. Sound waves spread spherically outwards in all directions from an audio source, but the further these waves propagate, the less intense the sound becomes. So, the further we are from a sound source, the more the sound waves will have dissipated, and this produces a reduction in volume. The intensity of sound and its decline in amplitude is termed the inverse law. This states: 'Sound pressure decreases proportionally to the square of the distance from the source.' That is, each time the distance from the original sound source doubles, it will become 6 dB quieter.

If this law is interpreted in the context of a mix, the louder an instrument is the closer to the listener it will appear. However, even though many dance mixes seem to have a complete frontal assault on your senses, this is not the case. Indeed, the depth perception is the first thing to take into consideration when producing a great dance mix.

If all instruments were placed at the front of the soundstage, all would be at an equal gain. This would result in a cluttered mix because every instrument would be at the front of the stage fighting to be heard. Moreover, the mix would sound two-dimensional because there is no comparison. For the listener to gain a perception of depth, there must be some sounds in the background so they can determine that some are in the foreground.

This means that before even approaching a mix, you should have a good idea of what the central focus of the music should be. Naturally, both the kick and bass should be at the forefront of the soundstage because the groove reigns. But a decision must be made on what other instruments are critical to the development and genre of the track. The bright chords in Euphoric Trance, the pluck in Progressive House, or the vocals in Vocal House would all be positioned towards the front of the soundstage because these are the key elements of the music.

While the gain is the most obvious solution to position sounds at the front, it is not the only option. As frequency increases, its wavelength becomes shorter so we can assume that if a high-frequency sound has to travel a distance, there is a reduction in high-frequency detail. This is evident on cars outfitted with large component stereo systems. You can hear the low frequencies when the vehicle is at a distance, but, as it approaches, the higher frequencies become more pronounced until it has passed by, when the high frequencies begin to decay again.

We can emulate this effect of a sound at a distance with the application of EQ or compression. By applying cuts of a dB at the higher frequency ranges, it could be perceived to be distant in association with other sounds. Equally, we could employ compression with a fast attack and a slow release. The compression cycle would capture the transient, reducing its high-frequency content and provided the release was long enough, this would also cut the high frequencies.

Another characteristic of depth and distance is determined by the amount of reverberation the signal exhibits. Any natural sound will demonstrate reverberation as the sound waves propagate and reflections occur from surrounding surfaces. However, the amount of reverberation and the stereo width of these reflections depend on how far away the sound source is from the listener.

If a sound is at a distance, the stereo width of the reverberations will dissipate as they travel through the air. They will also be subjected to more reverberation. This is important to consider when applying reverb since many novice producers will wash a sound in stereo reverb to create depth in the mix and then wonder why it doesn't sound 'right' in context.

If you use reverb to place an instrument towards the rear of the soundstage, then you should employ a mono reverb signal with a long tail. This could be followed with an EQ to reduce some of the higher frequency content of both reverberation and the original signal. This emulates the natural response we would expect from the real world, even if that particular instrument or sound doesn't occur in the real world.

Of course, this means that sounds that are positioned to the front of a mix should have little or no reverb associated with them. But in many electronic dance records, reverb is used to add character and presence. Here, the reverb should be applied in stereo but should be controlled to prevent it from occupying too much of the left and right perspective.

It should also employ a pre-delay of approximately 50 to 90ms to separate the instrument from the effect. This prevents the first reflections from washing over the transient and pushing it towards the rear of the soundstage. Always keep in mind that applying effects too heavily can make sounds challenging to localize and to accomplish a good mix each instrument should be pinpointed to a specific area.

Horizontal perspective – panning

The horizontal plane is the distance and positioning of the sounds between the left and right walls of the soundstage. The significant aural clues that help us derive the impression of stereo placement are attributed to the amplitude intensity between sounds and their respective timing.

Adjusting the amplitude of a sound to produce a stereo image was discovered by Alan Blumlein in the early 1930s. An inventor at EMI's Central Research Laboratories, Blumlein researched the various ways that the ears detect the direction of a sound's source. Along with inventing a technique to permit the creation of a stereo image in gramophone records, Blumlein also theorized that to maintain realism in a film the sound should follow the moving image. The technique was employed in Walt Disney's film *Fantasia*.

Sound engineers asked Harvey Fletcher (of the equal loudness curve) if he could create the impression of sound moving from left to right for the movie. Drawing on Alan Blumlein's previous work, Fletcher concluded that if a sound source were gradually faded in amplitude from the left speaker and increased in the right, it would produce the effect of sound in motion. The engineers at Disney constructed a *potentiometer*, a volume control that varied the volume between two speakers. This was later called the 'Panoramic Potentiometer,' resulting in the pan pots that feature on every mixing desks channel.

Although the volume intensity difference between two speakers is still the most common method for panning a sound across the stereo image, there are further techniques. We can also receive directional clues from the timing between sounds. This is known as the *directional cues*, *precedence*, or *Haas effect*, a process that takes advantage of the law of the first wave front that states: 'If two coherent sound waves are separated in time by intervals of less than 30 milliseconds the first signal to reach our ears will provide the directional information.'

If a direct sound reaches the ears anywhere up to 30ms before the subsequent reflections, it is possible to determine the position of a sound. So, if you were facing the central location of the mix, any sound leaving the left speaker would be delayed in reaching the right ear and vice versa for the left ear. This effect is known as **I**nteraural **T**ime **D**elay (ITD). As sound travels at approximately 340m/sec, we can emulate this effect by delaying a mono signal by a couple of milliseconds. This produces a very similar impression to panning a sound source.

For this effect to work, we must consider that our ears are on the side of our heads, so it gets in the way of the frequencies from opposite speakers. This head-related transfer function means that, provided you are facing the center of the stereo image, some of the higher frequencies emanating from the left speaker will be reduced before they reach the right ear. Placing a cut of 1 dB at approximately 8 kHz on the delayed signal could simulate this effect.

To accomplish this, we must employ mono sounds in the mix. This is another area where many novice engineers struggle. Almost every virtual instrument, hardware instrument, and most sample content is in stereo, not mono. But employing clear stereo signals on every channel often results in a mix lacking definition. If every channel in the mix is stereo, they will merge and occupy the same area and position of the soundstage.

The soundstage should be transparent so that it is possible to aurally picture the mix in three dimensions and pinpoint the position of each instrument. Since many electronic dance mixes are busy encompassing anything from 20 to 30 different instruments coinciding they each need a specific location within the mix. If each were a stereo file, it would be difficult to locate a pan placement for each instrument. This results in unnecessary EQ and *creeping faders*.

Use of creeping faders is again typical of novice engineers. It is when they gradually increase the volume of each channel so that it can be heard above other instruments. For example, they may increase the gain of the vocals, so they stand above the cymbals, hi-hats, and snare. But then the drums appear quieter in relation – moving them towards the rear of the soundstage – so *their* volume is increased. This results in the snares, hats, and cymbals being too loud compared to the kick, so the kick is increased in gain. Then the bass is too quiet, so that's raised in gain so that it shares a volume relationship with the kick. Then the vocals are increased again … Eventually, the mixer is pushed to its maximum headroom and distortion occurs.

To avoid this, we use a careful selection of stereo and mono signals and channels. Even at this later stage, a DAW mixer will often permit you to change the signal from stereo to mono. Moving through the mix, you can convert a channel to mono and listen for any differences. While there will be slight differences in every channel switched from stereo to mono, if it is not significant then it should be set to mono.

Typically, only the main lead elements of the mix are often stereo while a majority of the percussion remains in mono. For example, it is not unusual to have stereo chords and leads while all other elements remain in mono. For Dubstep, Deep House, and any bass-driven music, it is the bass that is commonly in mono *and* stereo. The mono bass occurs below 200 Hz while the stereo bass occurs above this frequency. It's also not uncommon to use stereo for closed hi-hats, some snares, and claps and even any arrangement effects such as risers and falls.

Although the kick drum is the driving force in dance music, this should be mono. This is because we will always face the loudest, most energetic part of a mix. If the kick is stereo, it will be equally dispersed across the left and right soundstage, decreasing the central energy, and reducing impact. In mono, the kick can be placed central to the mix, and this makes it easier to perceive the position of other sounds surrounding it.

When working with mono files, we will perceive the volume of a sound by its positioning within the soundstage. This means that if a mono sound is placed centrally, it is perceived to be anywhere from 3 to 6 dBs louder than if positioned in the left or right field. Some workstation mixers will implement the panning law so that sounds that are placed centrally are subjected to 3 or 6 dB of attenuation. The amount of dB attenuation can often be configured in the DAW.

Panning provides the easiest and cleanest method to create space for two instruments that share the same frequency range. By panning one instrument to the left and the other to the right, each sound can be awarded its space in the soundstage allowing both to be heard clearly.

The vertical perspective – EQ

The final perspective is the top to bottom of the soundstage. Here, higher frequency elements sit towards the top of the soundstage while lower frequencies lean towards the bottom.

The instruments already dictate much of this vertical positioning. For example, a bass will sit towards the bottom of the soundstage while hi-hats will naturally gravitate towards the top. However, all timbres contain frequencies that do not contribute to the sound when placed into the context of a mix.

All sounds we hear are made up of any number of frequencies that are a direct result of the fundamental and its associated harmonics occurring at differing amplitudes. Also, it is this predetermined mix of harmonic frequencies that help us establish the timbre of the sound we perceive, be it a piano, a synthesized bass, or a kick drum.

When any of these sounds are in isolation, we require these harmonic frequencies to determine the signature of the sound and identify it as an instrument. However, when many instruments are mixed, the harmonic frequencies from each of the different instruments overlap one another, and these are summed together to increase the volume at each specific frequency.

This summing of harmonic frequencies results in instrument signatures that exhibit uneven harmonic structures and result in a cluttered mix. For example, the lower frequencies of a piano may be unnaturally increased through the upper harmonics of a bass summing over the piano. Here, the piano would appear to exhibit too many low frequencies and give the impression it was poorly recorded.

If you can identify the frequencies at which this layering occurs, either the bass *or* the piano could have the summed frequency area removed with the careful application of EQ. This would result in a clean sound because it produces an effect we term *open frequency masking*. Here, we perceive that both instruments have all the harmonic frequencies present; they're just hiding behind the other instrument. As simple as this premise is, the practical application is more complicated.

In the example of the summing frequencies from a bass and piano, if both instruments are melodious, the frequencies they sum at will change. This makes precise application of a static EQ difficult. The workaround is to employ a dynamic EQ instead. This could be considered a frequency dependent compressor, you can choose the problematic frequencies and when they occur, they're reduced by the dynamic EQ.

These tools are no replacement for well-trained ears, however. To identify frequency masking and other faults within a mix, you must train your ears to recognize them.

Figure 34.1 shows one of the many frequency charts available on the Internet that suggest they can help the engineer identify frequency ranges. While these

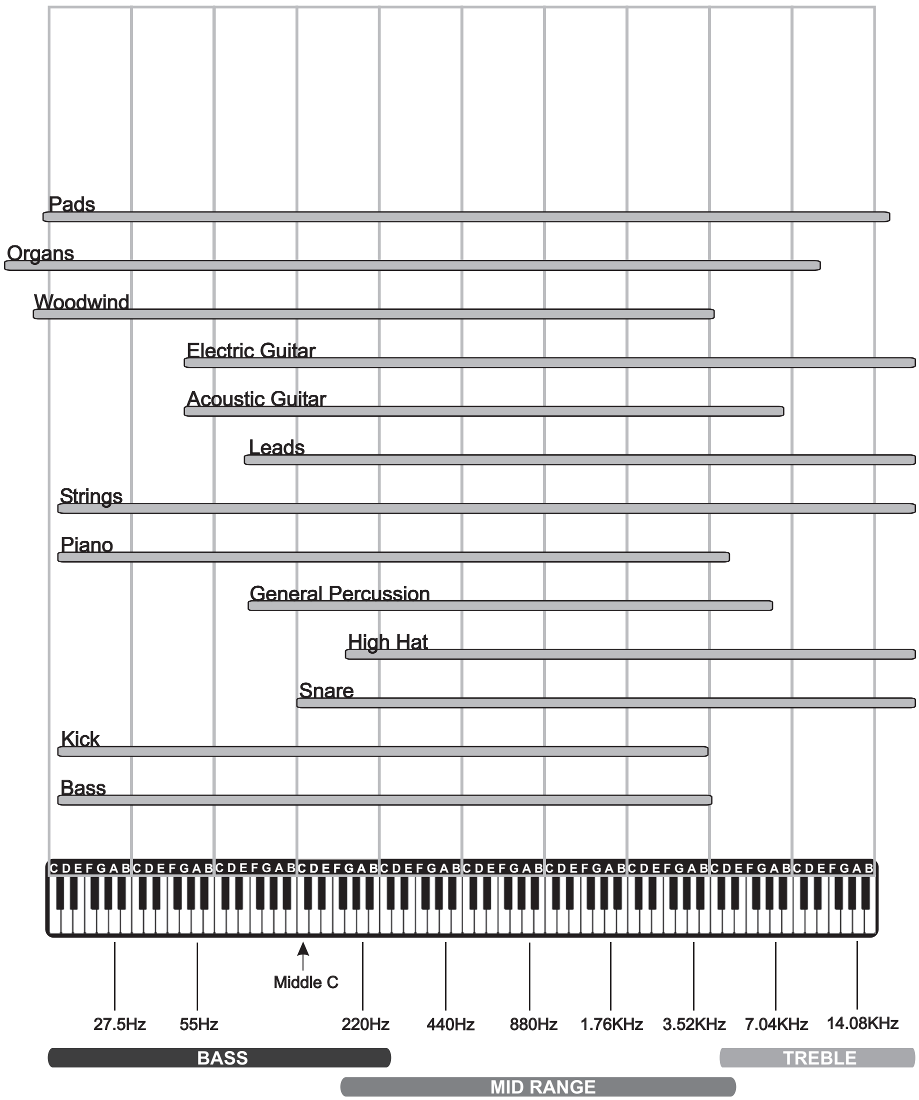

FIGURE 34.1
The frequency range of instruments

are useable to an extent, they cannot be reliably transferred to an electronic dance mix. These genres rely heavily on synthesized timbres alongside effects and processing that heavily modify the original tonality.

For example, distortion is a widespread effect to employ on bass in a dance mix, but the application of distortion will adjust the frequency range of the timbre rendering the chart useless. So, rather than rely on a frequency range of instruments, it is easier to refer to the seven EQ octaves while you train your ears.

With the EQ octave chart, you don't learn the frequency ranges that can be taken up by specific instruments, instead, its purpose is to focus on the effects that different frequency ranges have on instruments and a mix.

THE FIRST OCTAVE – 20 HZ TO 80 HZ

At frequencies lower than 40 Hz, it is challenging to determine pitch, but this area is commonly occupied by the lowest frequencies of a kick drum and some bass instruments. You should avoid applying any boosts below 40 Hz (unless you're using a Pultec EQ and then feel free to boost away at 30 Hz!). For surgical correction, you should instead concentrate on the 50 to 65 Hz area.

Small boosts here can define the bass or kick drum and produce more sub-energy into the mix. Cuts here are usually to the bass instrument to make space for the kick drum. The bass is cut instead of the kick because the kick does not change pitch and therefore frequency-masking issues are avoided if the bass is melodious.

When working in this octave range, it is important to listen on multiple systems, both large and small for inconsistencies in the energy. You should also monitor at differing volumes to compensate for the contour control curve.

THE SECOND OCTAVE – 80 HZ TO 250 HZ

This is where a proportionate amount of the low-frequency energy from most instruments and vocals reside. This range is affected when applying the 'bass boost' on most home stereos. Areas to concentrate on with boosts and cuts are typically at 120 Hz to 180 Hz. Lifts here will increase the fatness of most dance instruments while reductions thin them out. As before, you should also listen to this area at differing volumes to compensate for the contour control curve.

THE THIRD OCTAVE – 250 HZ TO 600 HZ

This frequency range is the main culprit for mixes that sound muddy or ill defined. It is also often the cause of mixes that are fatiguing to listen to. If too many harmonic frequencies clash in this area instruments and vocals will appear indistinct or boxy. Here, you should focus your EQ at 300 and 400Hz. Boosts at these frequencies will help define an instruments presence and clarity, alongside the body of vocals. Cuts will reduce boxy or muddy sounds.

THE FOURTH OCTAVE – 600 HZ TO 2 KHZ

The fourth octave can equally be attributed to instruments presence. Small boosts at 1.5 kHz will increase the attack and body of some instruments. However, it is far more likely that this area is cut because this range can be attributed to a honky or nasal sound. Typically, the central focus for EQ cuts is at 800 Hz, 1 kHz, and 2 kHz.

THE FIFTH OCTAVE – 2 KHZ TO 4 KHZ

The fifth octave is the *dance producer's transient octave*. The attack of most of the rhythmical elements resides in this area. The focus for EQ cuts and boosts are between 2.5 kHz and 3 kHz. Boost applied here can increase the projection and transients of most percussive elements, including the kick drum's 'bite.' Cuts will often push instruments towards the rear of the soundstage.

THE SIXTH OCTAVE – 4 KHZ TO 7 KHZ

The sixth octave is the distinction range for most instruments. EQ is typically centered on 5 kHz with boosts to increase air and any sonic features of the timbre while cuts will reduce sonic harshness.

THE SEVENTH OCTAVE – 7 KHZ TO 20 KHZ

The seventh octave contains the higher frequency elements of cymbals and hi-hats. Some engineers apply a small shelving boost at 12,000 Hz to make the music appear more hi-fidelity because this will increase detail and sheen without introducing any aural fatigue. The commonly adjusted frequencies with EQ here reside at 8 kHz, 10 kHz, and 15 kHz. Boosts applied at these frequencies will increase a timbre's clarity or air. Cuts here can remove harsh resonances from instruments and vocal sibilance.

CHAPTER 35

Practical mixing

I can't wait to hear it with a real mix instead of the douche mix.

John Kalodner

Armed with the theory of mixing, we can approach the mix on a more practical level. The first stage consists of cleaning up the audio tracks with a noise gate (if necessary) and removing unwanted frequencies from each instrument track with some aggressive EQ.

While all the harmonics are required to reproduce a sound in isolation, in the context of a mix many of the upper and lower frequencies can be removed with a shelving filter. For example, a hi-hat will often exhibit frequencies as low as 200 Hz and as high as 20 kHz. But, the majority of its energy – or the energy that matters – occurs between 6 kHz and 16 kHz. Therefore, any frequencies above or below this do not necessarily contribute to the sound in the context of a mix. If you insert a spectral analyzer onto a track, it can be surprising how much energy is taken up by instruments.

As shown in Figure 35.0, the energy of the bass resides between 30 Hz and 900 Hz. While there are frequencies above this, they do not *contribute* to the energy of the timbre and can be removed with a shelving filter. By doing so, it creates more space within the mix for other instruments to sit more comfortably.

Once the unwanted/unrequired frequencies have been removed from all of the instrument tracks, we can begin to set the relative volume levels of each track. Here, you have to take the musical style into account and identify the defining elements of the genre. Typically for electronic dance music, this means that the drums and bass should sit at the front alongside the tracks main lead elements or vocal track.

A standard approach is to mute all channels bar the entire drum section and commence by mixing these first. As the kick drum should be the most prominent part of the music this should be set to –18 dBFS (decibel full scale) on the DAW mixer and then the snare, claps, hi-hats, and percussion instruments can

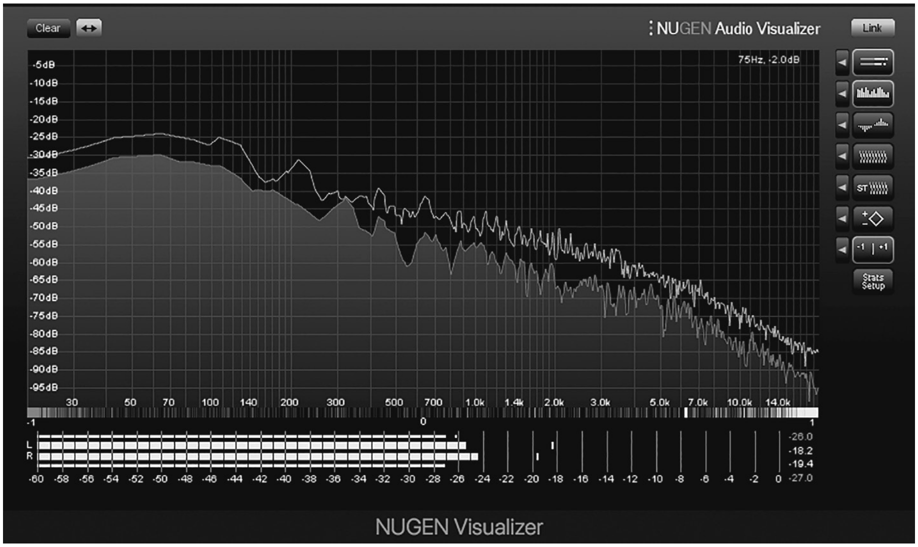

FIGURE 35.0
Spectral analysis of a bass timbre

be introduced at their relative levels to the kick drum. If the music appears quiet, you can always increase the output volume on the monitors, amplifier, or audio interface but you should avoid increasing the gain of the kick drum channel above −18 dBFS to leave headroom in the mix.

A conventional and useful technique for mixing is to monitor the output of the mix in mono. Some DAWs feature plug-ins that you can insert on the mix bus to mono the output or many monitor controllers will allow you to switch monitoring to mono. By listening in mono, the stereo information is summed, and this makes it far easier to level the instruments.

While many engineers will recommend introducing an instrument into a mix and applying any corrective EQ before introducing the next, it is not recommended for dance music. This additive EQ approach works well for recorded music because we know how a real instrument should sound and therefore it's possible to EQ before introducing another.

With electronic dance music, we should balance the drum levels first, and then apply EQ if necessary. The trick to mixing electronic dance music is to first balance and then EQ the drums so they sound full and complete, and then mix in the bass. The underlying groove generated by the drums and bass is the fundamental element to this style of music, so these should always be mixed first. After these, you can introduce further instruments in order of importance.

Some engineers will choose to mix through a bus compressor such as the SSL G series compressor, the API 2500, or the Focusrite Red 3. This is placed on the stereo output bus (or on the outputs of the audio interface), the levels are adjusted, and the track is mixed with the compressor engaged all the while. This is often termed as *mixing into a compressor*. There are as many proponents as there are opponents to this approach.

Proponents argue that mixing through a compressor helps you achieve the sound of dance music because it glues the mix together, and is a preferable alternative to placing the compressor on afterward as it can upset the mix. However, opponents suggest that this approach can severely restrict the dynamic range leaving the mastering engineer with little to no dynamics to work with.

From taking both approaches over the years, it depends entirely on the mix. Some mixes benefit from mixing through a compressor, but others do not, especially if they have been subjected to heavy compression techniques during the production stage. The problem with a single compressor is that the compressor is always driven by the loudest element of the mix (usually the kick) and this can result in *unwanted* pumping that is difficult to remove during mastering. If you want to mix through a compressor, then I would strongly recommend using a compressor with a side-chain. By doing so, the compressor can ignore the low frequency pulse of the kick and is therefore less likely to pump the mix too heavily.

An alternative approach is to employ *Braurizing*. This is a technique developed by mixing engineer Michael Brauer. For electronic dance music applications (Michael Brauer does not mix this genre) it involves dividing the mix into four groups as follows:

- Group 1: Drums and bass.
- Group 2: Vocals.
- Group 3: Leads.
- Group 4: Strings, keys, and sound FX.

Each of these groups is sent to different tonal coloring compressors and EQ's post-fader. These four processed signals then recombine at the stereo output. The purpose of this technique is to permit you to push the different groups harder (or more softly) into the subsequent compressors without the compressor overdriving or pumping other elements of the mix. The purpose here is *not* compression, but, instead, to use the compressor to impart tonal coloring on each group. Consequently, you need to use tonal compressors such as the Empirical Labs' Distressor (for the kick and bass), a Vari-Mu for the vocals, an opto for the leads, and a VCA for the strings, keys, and FX. This approach does avoid pumping but also produces a polished tone that, generally speaking, is not appropriate for most forms of dance music.

SUMMING MIXERS

Some engineers may also choose to mix through an analog summing mixer. These are commonly rack-mounted hardware mixers featuring 16 or 32 mono inputs but lack many of the features of a professional desk. Their sole purpose is to receive multiple channels from a DAW and sum them together to a stereo output. The theory is that digital summing is perfect and so lacks the character of analog. By using an analog device to sum channels together in the DAW, it produces a more robust soundstage with air and presence.

This is an expensive route and something that I would only recommend to professionals. *Good* summing mixers can cost in excess of £3000, and you will also require a quality multi-output audio interface (to feed individual channels from the DAW into the summing mixer), and these can add another £2000 to the price.

Because these summing mixers have limited inputs and many DAW mixers may consist of 40 to 60 channels, you must first group channels to send out to the summing mixer. How these are arranged depends on the artist, but typically, the mix is divided into two mono channels and seven stereo channels. An example of group summing is as follows:

- Channel 1: Mono kick.
- Channel 2: Mono bass.
- Channel 3/4: Percussion (both mono and stereo).
- Channel 5/6: Stereo lead instruments.
- Channel 7/8: Keys.
- Channel 9/10: Vocals (including stereo harmonies and backing vocals).
- Channel 11/12: Strings.
- Channel 13/14: Sound effects.
- Channel 15/16: Auxiliary buses.

Once the groups are fed into the summing mixer, the stereo output of the summing mixer is patched into the monitors, and you then proceed to set levels and mix *through* the summing mixer. You can hear an example of a mix created in a

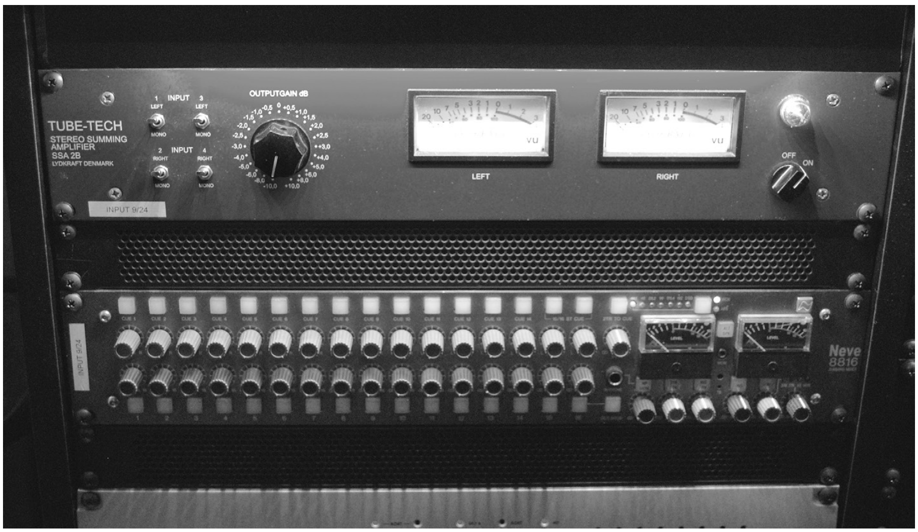

FIGURE 35.1
The author's summing mixers – Tube-Tech SSA2B and Neve 8816

DAW and then through a summing mixer on the DMM resources pack, available from www.dancemusicproduction.com.

Most, if not all of the effects are usually applied during the production process, so there is little requirement to introduce further effects beyond the application of reverb. However, during a mix, you may choose to be more creative and want to apply additional effects. In this instance, it is important to keep in mind that *empty* spaces in a mix do not have to be filled with reverb or echo delays.

When it comes to effects and mixing, less is invariably always more. Any effects, and in particular reverb, can clutter up a mix resulting in a loss of clarity. Since there will probably already be plenty going on, adding further effects will often make the mix busier and its essential to maintain some space in between the individual instruments.

A common mistake is to employ effects on every instrument when only one or two may be needed throughout. Therefore, before applying any effects, you should ask yourself *why* you're applying them: is it to enhance the mix or make a poorly programmed sound appear better? If it's the latter, then you should look towards using an alternative sound rather than try to disguise it with effects.

PANNING

After volume level and effects, panning is used to position each instrument. This is where your creativity and careful listening of mixes from the same genre come into play as there are no 'strict' rules for creating a good soundstage. They should be positioned where you feel they are most suited. That's not particularly helpful for a manual, however, so what follows are *recommended* starting positions:

- Kick. Positioned central so that both speakers share the energy.
- Snare or clap. Positioned from 3 o'clock or central.
- Hi-hats. Positioned from 3 o'clock or at the far left of the soundstage, perhaps with a delayed version on the right.
- Cymbals. Positioned central or from 1 to 3 o'clock.
- Further percussion. Positioned so that the different timbres are in the left or right of the soundstage.

As mentioned previously, we shouldn't be too concerned with the application of EQ. The focus is to change the relative amplitude levels, clean up effects and apply panning. If EQ is used at this early a stage, it is possible that you may force instruments into the frequency ranges required by further instrumentation. The purpose should be to accomplish the best possible results without resorting to equalization.

Once the drums have been mixed, they can be assigned to a **V**oltage **C**ontrolled **A**mplifier (VCA). This is a single gain control on the mixer that will control the relative volumes of all the drums together. More importantly, if you're using a hardware mixer interface, a VCA can spill.

This means that you could VCA the leads, drums, basses, effects, and vocals and mix with only five faders. But, if you need to edit the individual volumes of any submix, by selecting the governing VCA fader, all of the different channels contained within that submix will spill across the hardware faders allowing you to adjust the relative amplitudes. This can make mixing large track counts a lot easier.

If you're planning on employing further effects or processing to a drum bus, then rather than use a VCA, you will need to use a bus subgroup. This is because a VCA is nothing more than a gain control whereas a subgroup offers a gain control and the ability to insert processing and effects. The downside is that a subgroup will not spill.

After the drums have been mixed, you can then move onto the bass and further instruments. Introduce each instrument with gain and then move onto panning to their respective locations concerning mix priority. This usually means introducing the bass first so that it can be mixed with the drums to provide the central groove of the music, and then follow this with the leads, pads, strings, and sound effects.

In regards to panning these further instruments, the kick, bass and lead instruments (or vocal) should be placed center stage. This is particularly important for the bass and kick because it permits both speakers to share the low-frequency energy. As for the rest, it is open to interpretation and creative input. The best solution to panning instruments is to have a good reason as to why you're placing them there in the first place. Always have a plan and purpose for positioning and don't just throw sounds around the image regardless.

Suggested panning starting points for instruments are as follows:

- Bass. Low energy should be positioned centrally so that both speakers share the energy.
- Vocals. Often positioned centrally because you always expect the vocalist to be center stage but sometimes they are double-tracked and applied to both left and right of the stage too.
- Backing vocals. Occasionally stereo so they're spread from 2 o'clock to 4 o'clock but, if mono, there are often placed to the left or right of the soundstage.
- Synth leads. These are often in stereo and therefore positioned wholly left and right with perhaps an additional mono version placed centrally.
- Synthesized strings. Positioned at 4 o'clock or stereo spread at 9 o'clock and 3 o'clock.
- Guitars. Commonly stereo positioned at 3 o'clock and 9'oclock, on each side.
- Pianos. Commonly stereo with the high notes in the left speaker and low notes in the right.
- Wind instruments. Positioned in the left or right speaker (or both) at 3 o'clock and 9'oclock.
- Sound effects. Wherever there is space left in the mix!

In direct contradiction to the above, in dance music, the positioning of instruments is rarely natural, so feel free to experiment. In fact, it may be essential to exaggerate the positions of each instrument to make a mix appear clear and defined. That said, avoid positioning a sound in a different area of the soundstage to prevent any small frequency clashes with other instruments. In these circumstances, you should apply EQ.

This form of 'additive gain' mixing – introducing instruments step by step – is the most common form of mixing but it is not the only approach. I have worked with some artists who prefer a subtractive path by listening to all instruments and then merely balancing levels and panning until they are happy with the positioning and sound. Both are perfectly valid provided the final mix projects clarity.

EQ

After positioning the sounds in the soundstage with volume and panning, EQ may be required to correct any problematic frequencies. If two timbres of similar frequencies are mixed, some of the harmonic content is 'masked' resulting in the instruments disappearing behind one another and creating a boost in frequencies at that point. This 'frequency masking' results in a mix that sounds cluttered and ill defined.

To prevent this, we must focus on the masked area with EQ to cut the problematic frequencies. This means you need to able to identify and locate these problematic areas before they can be repaired. If you suspect that two instruments are masking, a useful technique is to mono both and set them to unity gain. Then, pan one instrument to the left and begin to move it slowly all the way to the right of the soundstage. If you can hear the sound running clearly from the left speaker through to the center of the mix and then off to right it's unlikely that there are any conflicts. However, if the signal disappears when drawing close to the position of the second instrument there is masking occurring.

Once masking has been identified, you will need to tune into the masked frequencies and remove them with some EQ. One of best techniques for identifying problematic frequencies is to set the EQ unit to a low Q and apply a boost of 6 to 8 dB. With this boost applied to the instrument, begin to sweep through the frequencies and make a note of each frequency you come across that makes your hair stand on end or appears to share similarities with the secondary instrument. These frequencies can then be removed with either low or high cuts.

This approach must be applied with caution. If one (or both) of the instruments are melodious, then the offending frequency may change as the track plays, and in this instance, you will have to employ a dynamic EQ.

With the application of EQ, we should always look to cut first to repair any problematic frequencies rather than boost. This is because cuts sound more natural to our ears because we are used to hearing a reduction in frequencies due to acoustic absorption of walls, objects, and materials. Plus, boosts will increase the gain of those particular frequencies, and that may result in clipping the signal of the audio channel. What follows is a rough guide to EQ and the frequencies that you may need to adjust to avoid masking problems.

Drum kick

A typical dance music kick consists of two major components: the attack and the low-frequency impact. The attack usually resides around 3 to 6 kHz, and the low-end impact commonly occupies between 40 and 120 Hz.

If the kick seems very low without a prominent attack stage, then set a high Q value and apply considerable gain reduction to create a notch filter. With this, use the frequency control to sweep around 3 to 6 kHz and place a cut just below the attack. This has the effect of increasing the frequencies located directly above.

If this approach doesn't produce the results, set a thin Q but this time apply 5 dB of gain and sweep the frequencies again to see if this helps it pull out of the mix. Taking this latter approach can push the track into distortion, so you may need to reduce the gain of the channel if necessary.

If the attack of the kick is prominent, but it doesn't seem to exhibit any 'punch,' the low-frequency energy may be missing. You can try small gain boosts around 40 to 120 Hz, but this may not fix the fault. A fast attack on a compressor may help to introduce more punch as it'll clamp down on the transient, reducing the high-frequency content but this could change the perception of the mix.

Snare and claps

The lower frequencies of both snares and claps often cloud a mix and are not necessary. The first step here is to employ a shelving (or high-pass) filter to remove all the frequency content below 150 Hz.

The 'snap' of most snares and claps usually resides around 2 to 10 kHz, while the main body can reside anywhere between 400 to 1 kHz. Applying cuts or boosts and sweeping between these ranges should help you find the elements that you need to bring out or remove. Here, cuts at 400 and 800 Hz should help it sit better while a small dB boost (or a notch cut before) at 8 or 10 kHz can help to brighten its 'snap.'

Hi-hats and cymbals

Naturally, these instruments contain very little low-end information that's of any use. Consequently, they benefit from a high-pass filter to remove all the frequencies below 300 Hz.

The presence of these instruments lies between 1 to 6 kHz while the brightness can reside as high 8 to 12 kHz. A shelving filter set to boost all frequencies above 8 kHz can bring out the, but it's advisable to roll off all frequencies above 15 kHz at the same time to prevent any hiss. If there is a lack of presence, then small dB boosts with a Q of about an octave at 600 Hz may add some presence.

Toms, congas, and general percussion

All these instruments have frequencies as low as 100 Hz but are not required so shelve off all frequencies below 200 Hz. They should not need any boosts in the mix as they rarely play such a large part but a Q of approximately half an octave applied between 300 and 800 Hz can often increase the higher end noticeably.

Bass

Bass is the most challenging instrument to fit into any dance mix because its interaction with the kick drum produces the essential groove, but it's also fraught with problems.

The main problem with mixing bass often comes from the choice of timbre and the arrangement of the mix. While dance music exhibits a high presence of groove, this isn't from using big, exuberant, harmonically rich sounds throughout. The human ear works under the concept of contrast, so for one sound to appear significant, the rest should be smaller. Of course, this presents a problem if you're working with a hefty kick and large bass; the two occupy similar frequencies that can result in a muddied bottom end.

This can be particularly evident if the bass notes are long as there will be little 'silence' between the bass and kicks making it difficult for the listener to perceive a difference between the two. Consequently, if the genre requires a huge, deep bass timbre, the kick should be made tighter by rolling off some of the conflicting lower frequencies and the higher frequency elements should be boosted with EQ to make it appear 'snappy.' Alternatively, if the kick should be felt in the chest, the bass can be made lighter by rolling off the conflicting lower frequencies and boosting the higher elements.

If you need both huge kick and bass elements in the mix, the arrangement should be created so that the bass and kick do not occur at the same point in time. In fact, most genres of dance will employ this technique by creating the offbeat bass and sidechaining any bass frequencies that occur under the kick.

If this isn't possible and both bass and kick must sit on the same beat, then you will have to resort to either side-chain or aggressive EQ on the bass. We have no expectations of how a bass should sound so if it's overlapping with the kick, you shouldn't be afraid to make some dynamic tonal adjustments.

For synthetic instruments, small dB boosts with a thin Q at 60 to 80 Hz will fatten up a wimpy bass that's hiding behind the kick. If the bass still appears weak, you should look towards replacing the timbre. It is risky to boost frequencies lower than these because it'll be challenging to judge frequencies on near-field speaker monitors accurately.

If the bass is lacking punch, then a Q of ½ an octave with a small cut or boost and sweeping the frequency range between 120 and 180 Hz may increase the punch to help it to pull through the kick. Alternatively, small lifts of ½ an octave at 200 to 300 Hz may pronounce the bass rasp. In some mixes, the highest frequencies of this rasp may begin to conflict with the midrange instruments so you may need to employ a shelving filter to remove the conflicting higher frequencies.

Another common problem with bass is the volume in the mix. While the bass timbre may sound okay, there may not be enough gain for it to pull to the front of the mix. The best way to overcome this is to introduce small amounts of controlled distortion. The best for this is the amp or speaker simulators.

Amp simulators emulate the response of a typical cabinet, so they roll off the higher frequency elements that are otherwise introduced through distortion units. As a result, not only are more harmonics introduced into the bass timbre without its sounding distorted, but you can use small EQ cuts to mold the sound into the mix.

If the bass is sequenced from a virtual instrument rather than a bounced audio file, then, before applying any effects, you should attempt to correct the sound in the synthesizer. We perceive volume from the shape of its amplitude envelope and harmonic content so simple actions such as opening the filter cut-off or reducing the attack and release stage on both the amplitude and filter envelopes can make it appear more prominent.

If both these envelopes are employing a fast attack and release, an alternative is to layer a kick over the bass. A light kick from a typical rock kit followed by some EQ can help to increase the attack stage.

If the bass is a real bass guitar, a different approach is required. Bass cabs will roll off most high frequencies, reducing most high-range conflicts, but they can also lack in bottom-end presence. Consequently, it's common to layer a sine wave underneath the guitar to add some additional bottom-end weight.

Bass guitars will also fluctuate wildly in dynamics, and these must be controlled with compression. The bass should remain even in dynamics throughout the mix and fluctuations will result in its disappearing behind other instrumentation.

Vocals

The groove relationship and syncopation between the vocals and the rhythm is a standard feature of dance music. So, you should exercise care in getting the vocals to sit correctly in the mix. Vocals should be compressed so that they maintain a constant level throughout the mix without disappearing behind instruments. A good starting point is to set the threshold so that most of the vocal range is compressed with a ratio of 3:1 and an attack to allow the initial transient to pull through unmolested.

Ideally, you should opt for a compressor that will add 'character' to the vocals. The preferred compressors for vocals are the LA-2A and the UREI 1176, both of which are available in software form. But the Soft Tube Tube-Tech CL1B should also be considered a strong contender for vocal compression.

Vocals rarely require any massive EQ adjustments but small dB boosts with a ½ octave Q at 10 kHz can make the vocals appear defined as it increases the consonants. Alternatively, if the vocals appear particularly muddy, an octave Q placing a 2dB cut at a center frequency of approximately 400 Hz should remove these problems.

If, however, they seem to lack any real energy, a favorite technique is to speed (and pitch) the vocals up by a couple of cents. While the resulting sound may appear 'off the mark' to pitch-perfect musicians, it produces higher energy levels that are suited towards dance vocals.

Lead synthesizers/pianos/strings

The rest of the instruments in a mix will commonly exhibit fundamentals in the midrange. These should be processed and have EQ applied in order of importance working, progressively towards the least essential instruments.

If vocals feature in the music, there will often be frequency masking where the vocals and midrange instruments meet so consider leaving the vocals alone and apply EQ cuts to the instruments. Alternatively, the midrange can benefit from being inserted into a compressor or noise gate with the vocals entering the side-chain so that the midrange dips whenever the vocals are present. This must be applied cautiously, and a 'duck' of 1dB is usually sufficient, any more, and the vocals may become detached from the music.

Most midrange instruments will contain frequencies lower than necessary, and while you may not be able to hear them, they will still affect the lower midrange and bass frequencies. To prevent this, employ a shelving filter to remove any frequencies that are not contributing to the sound of the mix. Set the high shelf filter with maximum cut and starting from the lower frequencies sweep up the range until the effect is noticeable on the instrument. From this point sweep back down the spectrum until the 'missing' frequencies return and then stop. This same process can be applied to the higher frequencies if some are present and do not contribute to the sound.

Keyboard leads will need to be towards the front of the mix but the exact frequencies to adjust are dependent on the instrument and mix. As a starting point for many midrange instruments, it's worth setting the Q at an octave and applying a cut of 2 to 3 dB while sweeping across 400 to 800 Hz and 1 to 5 kHz. This often removes the muddy frequencies and can increase the presence of most midrange instruments.

Above all, instruments within a mix should always be kept in perspective and you can do this by continually asking the following questions while mixing:

- Is the instrument brighter than hi-hats?
- Is the instrument brighter than a vocal?

- Is the instrument brighter than a piano?
- Is the instrument brighter than a guitar?
- Is the instrument brighter than a bass?

MID/SIDE PROCESSING

While many of the problems with a mix can be corrected with the careful application of EQ, some instruments and effects may benefit from mid/side processing. In fact, mid/side processing is often applied to create mixes that sound huge on club systems but sound more refined on small systems. Sometimes termed sum and difference, mid/side is available through some plug-ins such as iZotope's Ozone and Brainworx BX Digital, DynEQ, and Mono Maker.

These are an evolution of the mid/side microphone recording technique. This involved two microphones, one with a cardioid pattern and a second with a figure of eight pattern. The cardioid pattern would point towards the sound source, and the figure of eight would record the left and right ambiance field of the cardioid microphone. This approach permitted engineers to alter the stereo width of a microphone recording by increasing the ambiance volume and made it possible to convert the signal to mono by reducing it to just the cardioid recording.

A similar scenario to this exists within any stereo recording or stereo file. Here, the left and right information are called the sum, but the information contained in the center of this field is slightly different termed the 'difference.' Through employing a mid/side matrix, it's possible to treat the sum and the difference

FIGURE 35.2
The BX Digital V3

independently with EQ and dynamics. While this is useful for mastering engineers if they need to EQ or change the dynamics in a complete mix, it also has many uses for the mix engineer.

By employing a mid/side processor such as the BX Digital, it is possible to mono any frequencies that are below a set threshold. So, if you have an ample stereo bass, it would be possible to set the threshold to, say, 75 Hz so that any frequencies below this become mono while any frequencies above remain in stereo.

A widespread use is to increase ambiance in a recording and create mixes suitable for both clubs and stereo systems simultaneously. This is accomplished by placing the M/S processor after a stereo reverb unit. Since the reverberations are in stereo, the M/S processor can be used to reduce almost all of the reverb from the center channel (typically with EQ or compression) but leave it present in the left and right field. Since many club systems are mono, when played through a club system there is only a small amount of reverb present, but when performed via a hi-fi or iPod system, the left and right field exhibits a more significant reverb.

Similarly, reverb could be applied to an instrument and the frequency threshold adjusted so that the reverb becomes mono below a specific frequency and remains in stereo above. What's more, on a complete mix it is possible to boost the sum information (the left and right field) while leaving the center unaffected. This has the effect of increasing the stereo ambiance of the music.

Another option is to employ a mid/side dynamics processor, such as the BX DynEQ to home in on specific frequencies and compress the mono and stereo signals differently. Here, if employed on a full mix, both the kick and bass could be compressed and increased in volume compared to the side information resulting in a much more significant presence of the groove.

FIGURE 35.3
The BX DynEQ

COMMON MIXING PROBLEMS AND SOLUTIONS

Frequency masking

Frequency masking is one of the most common problems experienced when mixing, where the frequencies of two instruments compete for space in the soundstage. Here, you should identify the most important the two and pan or aggressively EQ the secondary sound to fit into the mix.

If this doesn't work, then ask if the conflicting instrument contributes enough to remain in the mix. Reducing the gain of the offending channel will not bring the problem under control because it will still contribute frequencies. Instead, it may just be easier to mute the channel altogether. If you have to keep the two instruments in the mix, consider leaving one of them out and reintroduce it later when it may not conflict with the secondary instrument.

Clarity and energy

All good mixes operate on the principle of contrast, i.e., the ability to hear each instrument clearly and locate its position in the stereo spectrum. It can be easy to introduce multiple instruments and channels to disguise weak sounds or arrangements, but this will not produce a great mix. A cluttered, dense mix lacks energy and cohesion whereas a great mix will feature fewer instruments. If the music appears cluttered, you should aim to remove the non-essential instruments until some of the energy returns.

If no instruments can be removed, then look towards removing the unwanted frequencies by notching frequencies from offending tracks. This may result in the instrument sounding 'odd' in solo but if the instrument must play in an exposed part then the EQ can be automated, or two different versions could be used.

Additionally, when working with the groove of the record, the silence between the groove elements produces an effect that makes it appear both louder and more energetic. So refrain from adding too many percussive elements. More importantly, good dance mixes do not draw attention to every part of the mix. Keep a good contrast by only making the essential sounds big and up front, and place the rest in the background.

Prioritize the mix

During the mixing stage always prioritize the main elements of the mix and approach these first. A common pitfall is to spend a full day tweaking EQ on a hi-hat or cymbal without first maintaining an even good balance of the essential elements. Always mix the most important aspects first, and you'll find that many of the 'secondary' timbres will tend to look after themselves.

Relative volume

Analytical listening will tire your ears, so avoid monitoring at a loud volume because this will only hasten the process. The ideal standard monitoring volume is around conversation level (85 dB). But, keep the Fletcher Munson contour control in mind and after every volume or EQ adjustment, reference the mix again at various gain levels. It can also be beneficial to monitor the mix in mono when setting and adjusting the volume levels, as this will reveal the overall balance of the instruments more clearly.

EQ

EQ can be used to shape all instruments, but you can only apply so much before the instrument loses its character. Always bypass EQ every few minutes to make a note of the tonal adjustments you're making. Remember that, while an EQ instrument may not sound correct in isolation, it is unimportant provided that it sounds right with the rest of the mix.

Cut EQ rather than boost

Our ears are used to hearing a reduction in frequencies rather than boosts because this mimics real life. While some boosts may be required for creative reasons, look towards mostly cutting to prevent the mix from sounding too artificial. You can effectively boost some frequencies by cutting others, as the volume relationship between them will change. This will produce a mix with clarity and detail.

Avoid using EQ as a volume control

If you have to boost frequencies for timbre volume or design you should not have to boost by more than 5 dB. If you have to go higher than this, the sound was poorly recorded or the wrong choice for the mix.

The magic Q

A 'Q' setting of 1 1/3 octave is generally suitable for most instruments and produces the best results. That said, if the instrument is heavily melodic or you are working with vocals, wider Q settings are preferred and a typical starting point is about two octaves. Drums and most percussion instruments will benefit from a Q of half an octave.

Shelf EQ

Shelf equalizers are generally used to cut rather than boost because they work at the extremes of the audio range. Using a shelving filter to boost the low frequencies will only accentuate low end rumble because there's little sound this low. Similarly, using a shelf to boost the high range will increase all the frequencies above the cut-off point and there's very little high frequency energy above 16 kHz.

Good enough …

… Isn't.

It is either right or wrong.

Table 35.0 Chart indicating frequencies for mixing

Frequencies	Musical effect	General uses
30–60 Hz	These frequencies produce some of the bottom end power but if boosted too much can cloud the harmonic content, introduce noise and make the mix appear muddy	• Boosts of a dB or so may increase the weight of bass instruments for Drum & Bass • Cuts of a few dB may reduce any booming and will increase the perception of harmonic overtones, helping the bass become more defined
60–125 Hz	These frequencies also contribute to the bottom end of the track but if boosted too heavily can result in the mix losing its bottom end cohesion resulting in a mushy, 'boomy' sound	• Boosts of a dB or so may increase the weight of kick drums and bass instruments and add weight to some snares, guitars, horns, and pianos • Cuts of a few dB may reduce the boom of bass instruments and guitars
125–250 Hz	The fundamental of bass usually resides here. These frequencies contribute to the body of the mix but boosting too heavily will remove energy from the mix	• Small boosts here may produce tighter bass timbres and kicks, and add weight to snares and some vocals • Cuts of a few dB can often tighten up the bottom end weight and produce clarity
250–450 Hz	The fundamentals of most string and percussion instruments reside here along with the lower end of some male vocalists	• Small boosts may add body to vocals and kicks and produce snappier snare timbres. It may also tighten up guitars and pianos • Cuts of a few dB may decrease any muddiness from mid-range instruments and vocals
450–800 Hz	The fundamentals and harmonics of most string and keyboard instruments reside here, along with some frequencies of the human voice. Cuts are preferred here as boosting can introduce fatigue	• Small boosts may add some weight to the bass elements of instruments at low volumes • Cuts of a few dB will reduce a boxy sound, and may help to add clarity to the mix
800–1.5 kHz	This area commonly consists of the harmonic content of most instruments so small boosts can often add extra warmth. The 'pluck' of most bass instruments and click of the drum kicks attack also reside here	• Boosts of a dB or so can add warmth to instruments, increase the clarity of bass, kick drums and some vocals, and help instruments pull out of the mix • Small dB cuts can help electric and acoustic guitars sit better in a mix by reducing the dull tones

Frequencies	Musical effect	General uses
1.5–4 kHz	This area also contains the harmonic structure of most instruments so small boosts here may also add warmth. The body of most hi-hats and cymbals also reside here along with the vocals, BVs, and pianos	• Boosts of a dB or so can add warmth to instruments and increase the attack of pianos and electric/acoustic guitars • Small dB cuts can hide any out-of-tune vocals (although they should be in tune!) and increase the breath aspects of most vocals
4–10 kHz	Finger plucks/attacks from guitars, the attack of pianos and some kick drums, snares along with the harmonics and fundamentals of synthesizers and vocals reside here	• Boosts of a few dB can increase the attack on kick drums, hi-hats, cymbals, finger plucks, synthesizer timbres, and pianos. It can also make a snare appear more 'snappy' and increase vocal presence • Small dB cuts may reduce sibilance on vocals, thin out guitars, synthesizers, cymbals, and hi-hats, and make some sounds appear more transparent or distant
10–15 kHz	This area consists of the higher range of vocals, acoustic guitars, hi-hats, and cymbals and can also contribute to the depth and air in a mix	• Boosts of a few dB may increase the brightness of acoustic guitars, pianos, synthesizers, hi-hats, cymbals, string instruments and vocals
15–20 kHz	These frequencies often define the overall 'air' of the mix but may also contain the highest elements of some synthesizers, hi-hats, and cymbals	• Boosts here will only increase background noise such as hiss or make a mix appear harsh and penetrating. Nonetheless, some engineers may apply a shelving boost around this area (or at 10 kHz) to produce the Bandaxall curve

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.

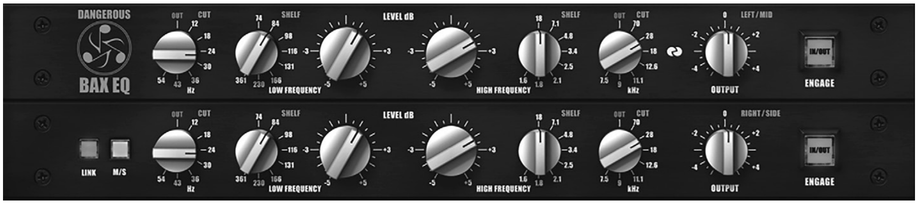

FIGURE 36.3
UAD BAX EQ

- 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.

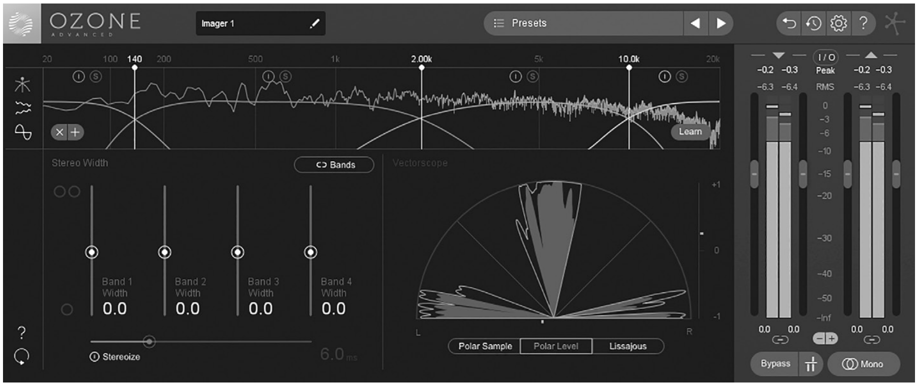

FIGURE 36.4
Ozone 8 Imager's Vectorscope

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 37

Publishing and promotion

Competitions are for horses, not artists.

Bela Bartok

Once you've composed, arranged, produced, mixed and mastered your music, it's time to reveal your artistry to the world. For more information about how you should approach publishing and promotion of your music, Matt Caldwell has contributed this chapter to the book.

Matt is a promotions specialist and director of multiple businesses operating in the dance music industry. His agency Matt Caldwell PR is home to a library of the industry's leading artists and record labels and is regarded as one of the leading service providers of PR for electronic music projects. With over a decade of experience in the music business, he's experienced nearly every role in the industry from BBC Radio 1 featured producer to managing promotions and DJing with the world's biggest nightclub Privilege Ibiza. Matt has delved into most sectors of the business at high levels culminating in an understanding of the inner workings of the industry and has contributed to a string of articles and features talking about these subjects for a broad spread of international dance music press.

PUBLISHING AND PROMOTION

I'd like to thank the author Rick Snoman for reaching out in interest of my contribution to this wonderful series of books. A decade ago I was myself a student of the industry sitting in classes at university and this very book was a regular source of reference for lecturers and students alike. To be asked ten years to later share my acquired knowledge on the subjects covered is a privilege and also a testament to the importance of this book and what it includes. I'd also like to thank those who contributed creatively to the chapter: Steven Bartolo, Jamie Stewart, and Marcus Knight.

The world of promotion, PR, and how to turn great dance music records into popular ones is both an art and a science. There are endless libraries of incredible tracks that never see the light of day and, conversely, a whole lot that become globally famous that perhaps lack a level of quality that we might expect. Why is this? How do songs that we see at the top of the charts get there? How are artists getting their music on the best record labels in the business? Where are the industry's leading DJs picking up new music to fill their shows with? Once I've reached the end of the road creatively when producing music and have exported my mastered work into a final version for the world to hear … then what happens? What is PR? What does a PR agent do? Do I need one? These are questions that every ambitious budding producer will ask himself and rightly so. We invest years of time and often vast sums of money into production equipment, formal education, and training to get our sound just right, but once we have that polished piece of music at the end of the production journey, it might *feel* like the end of the process, but we're just getting started.

This chapter will introduce the concepts and methods used by people on both sides of the fence, both artists and music industry professionals alike. We will look at what it takes to establish a project into the public domain successfully and how to navigate those waters. Second, we will develop an understanding of the roles of those who work in the industry outside of the production process and develop an understanding of how to implement their unique skillsets. Our goal here is outlay a road map of what to do, who to speak to, where to push your music towards and how to give your project the very best shot at success.

This is where the game changes from the confines of your studio set-up's walls into the world. When we're convinced that our music is ready to be unleashed into the public domain, it can seem like a daunting prospect to start this journey and never more so than in modern times. It takes a lot of hard work, highs and lows, stress, and just about every emotion that can be imagined, but if the music is great and the passion is resilient, the possibilities are endless in this wild industry that we all know and love.

STAGE 1: PREPARATION, BRAND, AND PROFILE

It's tempting to try to flash through the foundations of promotion when you've got a killer demo sitting there ready to take the world by storm, but it's important to do the groundwork first. It's just like that first time you uploaded something to SoundCloud in a rush to show your best friends what you've been working on, only to realize a couple of hours later that the song was far from finished. Inevitably, you delete the upload and end up back in your DAW that same day. You can get away with this with your buddies, but jump the start and do this to a serious record label A&R or industry manager and you'll quickly come to regret your premature moves. When you're one of 100 demos for the day to listen through, first impressions are vital. Presentation and image are critical, which means that groundwork, too, is critical.

Artist names

Chances are you already have your project's name decided, but even if you do, have a long and hard think about it and make a decision for the foreseeable future. It may seem obvious but going back and changing your name becomes more and more problematic as you invest time and energy into your project. Your first 1000 fans are harder to come by than your subsequent additions, so being committed to your name and brand is essential. Be sure to check online that a popular artist is not already using your name as that is going to be a significant headache waiting to happen. Check SoundCloud, Facebook, Twitter, Beatport, Wikipedia, and the likes of to uncover any projects already using the name.

Profiles and biographies

Having a presence online is fundamental in the modern music industry. It would be a bad idea to plot your future based on the paths of the rare outliers that blew up overnight and went from their bedroom to a festival headliner in a few weeks with one colossal release that got picked up by a major label. For everyone else, the truth is that it takes months and years of grinding through the ranks to develop a project that has international traction. Your profile is your story and fans, and music industry professionals are drawn to these texts to get to know you. Some artists think that they don't need a biography until they've got something big to brag about, but that couldn't be further from the truth. Your artist bio can detail your upbringing, backstory, musical influences, lifestyle, and more. If you picture your favorite artists, you know things about their story beyond just the music that they release and these details make them interesting to us. Communicating that narrative is a hugely effective way to build up a personal relationship with your fans and followers beyond just the music. The industry standard is 300 to 500 words, and a functioning artist would usually review this document annually, hopefully adding a year of successes to each new edition.

Image

When looking at new talent for the first time, industry professionals or even fans themselves see before they hear. You can lose interest and trust in a product before you even try it if the presentation and image are wrong and music is no different. Luckily, technology is now at a point where our smartphones are capable of photographs that go some way to rivaling professional DSLR cameras, and graphic design knowledge is mainstream enough that you likely know or are someone capable of putting together a digital logo. The availability of these skills and technologies means that there is no excuse for not having both press photos and logos for your project and this should be implemented right away, from the day you go live online.

Press photos are another chance to broadcast your brand, whatever that may be. Whether you're the conventional techno artist with the serious face, black t-shirt, and sunglasses look leaning into a brick wall background, or a crazy and creative personality who comes up with a unique scene that we've never seen before, having a professional looking press shot is one of the first good decisions you can make when launching a new artist project. Anonymity or focus away from the person behind the music does, of course, exist, and if you're going down that path, then you need something else to keep your fans' eyes busy. Whether that's Deadmau5's mouse head helmets, Major Lazer's animations, or Aphex Twin's eerie characters and manipulation of human form, people expect something to look at, whatever the genre may be.

Logos

Whether it's the famous Ministry of Sound stamp, Metalheadz' iconic skull logo, Hot Creations' palm tree, Spinnin' Records' famous S, Carl Cox's iconic branding that covers the island of Ibiza – one thing is for sure: fans love a memorable logo and attractive branding. Accessibility to graphic design software is greater than ever before. If you do not know where to start, you'll likely know someone who does. The likes of Adobe Illustrator and Photoshop are available at reasonable prices that enable access on a monthly basis meaning that you can sign up to use any software from their Creative Cloud for around £20 per month. This convenient setup means that users can subscribe for just a few weeks to get their design requirements completed, without having to invest the full price of these applications, which were historically expensive when ordered in full before this model was standardized.

As with most things in the creative domain, if this does not seem like a task you wish to invest your energy in, there are plentiful options online for hiring freelancers to handle design work for you. Freelance marketplaces such as those previously mentioned in the chapter are stacked with designers of every skill level meaning that whatever your budget, there will be someone out there willing to fulfill your design needs.

STAGE 2: GETTING ONLINE

Once we've established our project name, profile, image, and branding, it's time to implement this package into the online domain. There is an endless stream of websites, social media networks, and music platforms available and which ones and how many you decide to use is up to you. We're going to focus on the current largest scale options but don't limit yourself to those mentioned. Some genres are more active on different platforms, so it's advisable to research your style thoroughly and understand where your target market is spending time online. The Internet moves quickly, and websites come and go, so it's a smart move to be the first, rather than the last person in your network to show up on new online platforms. Just be sure that you don't leave yourself with too much to do, as it's better to be active on two or three platforms than stagnating on 20.

A lot of new artists shy away from responsibilities when it comes to the business side of music projects, and you can understand why. It's a whole lot more fun to sit in the studio all day working on music rather than be putting in long and tiring days of creating and managing websites, biographies, and social media channels. The problem is that this side of the business is critical and unless you've got the budget to have someone else taking care of these tasks for you, it's a necessary part of the life of a developing musician in any genre of music. It's a tiny percentage of the world's active musicians who enjoy the luxury of having no responsibilities beyond producing new music and playing shows, so you'd better get used to the idea of being your management team and taking on the less attractive jobs required, at least for now. Everyone wants to be an international producer or DJ, but a much smaller percentage of those have the heart, drive, and passion that's required to make it happen. Finding and harnessing that energy is what separates the hobbyist from the professional and that battle is usually won and lost on the Internet where the world finds and follows new music.

Websites

Having your official website has never been easier and setting one up is one of the best ways to show the dance music industry that you're serious about what you do. We know this already, but often forget to apply it to what we do in music. If you want to know more about any product or service, we look it up on Google. If nothing shows up on search engines, we're instantly skeptical and lose trust.

First up, you need to register a domain name on a site such as GoDaddy, 123-Reg, or 1and1. These services are mostly similar, and you can expect to pay anywhere from £10 to £20 per year for a dot.com domain. Extras such as domain privacy, server space, and mailing facilities are optional depending on your requirements, but at least one inbox is recommended as it adds a professional layer to your future communications, an aspect we'll get to later in the chapter.

In 2018 the most popular website building tool for small businesses and independent projects is WordPress. The system is an open-source Content Management System (CMS) and is widely regarded as the best approach to building professional websites without having to spend years learning how to do it. What WordPress does well is remove the necessity of understanding and learning any coding languages, meaning that with just a few tutorials and a good base template, you can have a slick looking website that's styled to your taste without breaking the bank.

Where WordPress comes into a league above the competition is with user-generated content, which makes life much easier for those that know little about website development. Think of WordPress as a DAW for your website and just like when making music, you can source an array of plug-ins that do a variety of tasks to help your page look great. Websites such as the Envato Market offer over 3 million digital products developed by the community. They offer a vast range of options from entire website design templates, through to smart and innovative plug-ins that enable you to add functionality to your website to do just about anything that you desire. For a modest budget, it is entirely possible to put together a professional website with just a few days of effort and a bit of patience. Alternatively, you could outsource the job on one of the many online gig marketplaces available. You can find very reasonably priced web designers, developers, app creators, and pretty much anyone you'll ever need to assist your website's construction. Sites like Freelancer, Peopleperhour, and Fiverr are thriving communities of global technology service providers who are often sole-traders, which results in a competitive market and prices that are much cheaper than you might expect. If you've got the time and patience to learn the basics of the WordPress system yourself, it would be a smart move to do this, as the modern musician is much more than just an artist and producer of music.

Once settled on a set-up and template, you'll want to build the necessary arrangement of pages. Generally speaking, a musician's website will have the following:

- homepage
- about/bio
- shows/tour dates
- music
- gallery
- news/blog
- contact
- links to social media.

You should do a little research before starting the build to get an idea of what your site should look like. It seems that most producers and DJs have a similar set-up and by sourcing a suitable template and adding the pages above, you'll have the majority of what's required and what you do from there is entirely down to your creative vision.

Social media

Facebook, YouTube, Twitter, Instagram, Google+, Pinterest, Snapchat, LinkedIn, Tumblr, Reddit, Vimeo, Vine, Periscope – the list is endless. The impact that social media have had on the music industry is colossal as fans no longer expect two or three releases a year and the occasional interview. If you take a look at the majority of the world's most prominent artists and their social media outputs, there's a dizzying stream of updates, blogs, news, photos, competitions, videos, polls, and everything in between. There are no limits when it comes to social media and those who are willing to invest the time and energy into running a 24/7 social media machine are reaping the rewards. We are all locked in a new online world of constant content and accessibility, which has made us hungry and expectant for more. I'm sure you already follow numerous examples of this new world, and it's clear to see that those who are committed to social media are developing massive reaches of potential fans with unlimited possibilities. It's no longer the case that hyperactive social media accounts are restricted to Hollywood or reality TV stars. Whether you're merging into the world of House, Techno, Trance, Dubstep, Drum & Bass, Electronica, Hip-Hop, Trap, Grime, EDM, Garage, or anywhere in between, you'll find a large proportion of the key players are posting multiple updates every day to masses of fans who just can't get enough.

The important thing to understand about social media is that innovation is never more than a handful of years away and the networks that are huge now could well be a distant memory in five years' time. Because of this inevitable nature of technology, it is essential that music industry professionals keep a close eye on developments in the world of social media and make every effort to ensure that we don't get left behind.

In 2018 the most prevalent social media networks in the music industry are Facebook, Twitter, and Instagram and each of those comes with its unique population and format. Facebook enables users to create dedicated artist fan pages, which are an entity of their own aside from your profile. It is viewed as a bad idea to not have an artist page, and any respectable record label, management agency, or industry peer is likely to expect to see a social media presence for any artist being considered for a deal of any description. Twitter, in comparison, is often more informal and conversational. Tweets are limited in character length, which, in turn, results in a higher frequency of posts. If Facebook is for announcements and news, then Twitter is much more like an extension of the brain where the rawest and least prepared thoughts seem to make it onto feeds and engage followers. Twitter users are perhaps less interested in content and more interested in volume and the personal touch. Facebook is also generally viewed as a more formal setting for music industry accounts and given that Twitter is the less censored of the two, content such as public spats (often staged or exaggerated to entice fans) and the likes of usually gravitate away from Facebook for figures who use both.

Music platforms

Streaming is king for modern music consumption on the back of a declining sales market for the industry as a whole. Previous editions of this and other books in the genre would have predicted today's structure, but we've nearly evolved in totality from the format of paying for each song that we wish to have access to whenever we want. In dance music, especially, which is the focus of this book, sales charts on digital download stores are a better indicator of what's popular with DJs rather than fans as they make up a high percentage of total digital sales. Access to high-quality streaming platforms has transformed the consumer industry, and, in today's world, an individual's music library is much more likely to be a series of playlists and browser bookmarks than a stack of vinyl or CDs on a shelf. Because of this incontestable fact and the lack of evidence to suggest any return to the old model, it is vital that budding producers and DJs have a stable and evolving understanding of the role that music streaming plays in our industry. This should not be interpreted as 'vinyl is dead' or that there is no future for physical sales, because there are still genres with firm roots in the distribution and sale of vinyl and other physical products, but the online streaming domain is here to stay for the foreseeable future.

Much like social media, music platforms will evolve, and outdated systems will get left behind, but it's our job as providers of music to remain ahead of the curve and ensure that we're giving fans and followers our content in the formats that they demand. We can delve into the details of what's popular now, but the smartest move is to keep a regular check on the next big thing and ensure that you're not late to the party.

SoundCloud

At the time of writing, SoundCloud is the industry standard platform for hosting and sharing unreleased streamed music. The platform borrowed concepts from the likes of Myspace before it and centralized their efforts around the needs of the independent musician by blending the favorite social media profile set-up with an ability to upload, comment, like, and repost songs. The capacity to upload audio in private opened up the possibility of using the platform as a tool for hosting demos that need to be kept under wraps. Up until SoundCloud became the standard, a record label's demo inbox was a data-heavy place to be with prospects overloading mail servers with large attachments making life hard for A&Rs to manage. When you're dealing in large WAV files and have to work through a never ending parade of demos, you can imagine the relief when SoundCloud took the weight away by simplifying the process down to a URL that brought you to an online portal where the audio is hosted, streamed, and downloaded if required.

It would be wise to focus on building a following on SoundCloud presence, especially if you plan on using the platform to submit demos or DJ promos. Record label A&Rs tend to respond with better regularity and detail when they are given easy access to private streams, which makes the maintenance and groundwork of your profile that more important. Industry peers in all situations are susceptible to developing judgments based on more than just the quality of the music being pitched to them. This fact is fuel to encourage artists to look as professional and reliable as possible on every platform that you're using to interact with people who make decisions that can have an impact on your music career. Importing and correctly formatting the documents we've already mentioned such as quality press photos, artist biographies, and social media profiles will go a long way towards gaining the trust of anyone browsing your page. It is not uncommon to see high-profile DJs, producers, A&Rs, and others skip even previewing a demo or promo if the sender does not appear serious. Given that we spend weeks and months perfecting our music in the studio, it would be wise to treat our public presentation and image with the same significance. How we appear is just as important as how we sound which makes adopting this attitude from an early stage a smart move for the next generation of industry professionals. It may appear a superficial viewpoint, but the evidence supporting it is clear – it's not all about the music anymore.

Tips for succeeding with SoundCloud include:

- Follow artists who align with the music that you make. It's beneficial to embed yourself into the culture and networks that make up your genres.
- Source artists and record labels that are at a similar level to your work and make friends to build your network. Synergy is a powerful thing for up and coming projects in the music industry.
- Support other artists whom you appreciate.
- Don't be silent. Leave comments and stay social. Every comment that you leave is a new link to your profile, and this incremental promotion of your work adds up over time.
- Repost the work of your friends; they'll probably return the favor.
- Use SoundCloud as a tool to source collaborators.
- Find and collect popular groups and channels that post music that's relevant to your style.

Implement these and your efforts to the SoundCloud platform, and it will become a powerful and enjoyable tool for your projects.

YouTube

In 2018, the top ten highest traffic English-speaking websites in the world include Google, Facebook, Wikipedia, Reddit, Yahoo!, and Amazon. These are comprised of search engines, social networks, and e-commerce. The outlier is YouTube, the only media-streaming platform to make this list of web powerhouses. This stat makes the popular video-sharing portal an indispensable tool for music projects. SoundCloud might be a fantastic tool to use within the industry, but the sheer volume of music consumers with active YouTube accounts must not be overlooked. From social gatherings to users working through the day with headphones plugged in, YouTube is one of the most popular music players in the world.

Tips for succeeding with YouTube include the following:

- The visual presentation is important, so ensure that the background image or video is appealing. It's common for artists and labels to post the cover artwork as the video background.
- Prioritise YouTube over SoundCloud as your media manager for promotion that is targeted at fans rather than your music industry network. Many more people hold YouTube accounts compared to SoundCloud, which increases the likelihood of picking up new listeners, subscribers, and followers.
- Standard practice for most professional dance music record labels and artists is to premiere a one to two-minute preview of an upcoming release

on YouTube in the weeks running up to public sale date. These uploads are used to draw attention to the project and can be used when promoting on social media and in the online press where embeddable media content is in highly desirable.

- Always ensure that every upload is accompanied with a well-written subtext block describing the content.
- Finish your upload with your web and social media links to drive traffic to your profiles.
- Much like SoundCloud, keep records and contact info of popular and relevant channels that post the genres of music that you produce. We'll get to details on how to promote to these networks later in the chapter, but your contacts and network will pay dividends in the long run if well managed.

Spotify

The rise of Spotify and other streaming platforms has led to the dawn of a new era for the music industry. As technology shifted away from physical sales into digital forms, something else happened as a side effect. The average modern music consumer not only wants access to Mp3 or WAV files, but they also want to be able to hear our releases in full, on demand, and without parting with any money. This triggered a huge problem for the industry to contend with as the days were numbered for the sale of music being a primary income source.

What Spotify did enabled this ultra-low-cost reality of today's industry to remain profitable to artists and record labels. Setting up a subscription business model that acted much like a Netflix for music, Spotify created a new economy that's driven by subscribers and advertising revenue. This enabled the platform a generate money without directly forcing users to pay for every song they consume. Artists and suppliers of music are then paid a nominal amount per stream, which partly solved the issue of public demand for free music.

There are still serious problems in this new reality, and we're still working it all out. From an industry standpoint, the amount that artists and labels generate in profits is deemed as far too low, but when the choice is this or succumb to piracy, this concept remains the most attractive that we have.

If we look at the average profit per stream against that of a sale, we see a rapidly collapsing value in the price of a song.

Spotify stream (at the time of writing) – £0.0027

Beatport Mp3 Sale – £1–£2

CD single – £3–£5

Vinyl single – £7.50–£10

At every level from independent artist up to the major label, the profits possible from the sale of music have all but collapsed. At £10 per unit, a vinyl single needs to sell 1000 copies to generate £10,000 in sales. To earn the same amount on Spotify, a song would need to rack up a huge 3,703,703 plays. This culture shift has transformed the industry and tasked artists, labels, and managers to source other ways of generating profits.

Tips for succeeding with Spotify include:

- Sign up for Spotify for Artists. The platform offers a profile and tools for artists, which enables management of your presentation on the software.
- If Spotify is your platform of choice for offering streams, be sure to link to your account and uploads on social media posts, press releases, and anywhere else that you spread news of your releases.
- Post playlists. One of the most popular features on Spotify is the playlist function. Users can subscribe to your posts and follow what you're listening to. An artist with a well-stocked playlist and a large following can become a powerful tastemaker.
- Submit your music to popular playlists. A considerable percentage of Spotify users leave the discovery of new music to viral playlisters. Build a network of relevant contacts that feature the genres that you make and pitch for support on these powerful playlists.
- Advertise. Advertising revenue powers Spotify and you too can place ads which will preview a snippet of your music in between public plays on the platform.

STAGE 3: PROMOTION

Introduction

Now that we have made significant progress towards ensuring that our projects are presented professionally and that our brand and story is strong, it's nearly time to jump in and start promoting our work and building the future. The music industry promotion world is as open ended as the music production side of things. When you open up your DAW, an infinite number of possibilities are in front of you and the roads you take, and the decisions you make will dictate the final product. Many producers starting their journey will grumble that they cannot compete with major artists as they do not have the same access to high-end synthesizers, monitoring speakers, hardware FX modules, a vintage Neve console, or a staff of engineers. Others, however, will emerge through the noise and disprove these sentiments with releases that rival the best in the business from the humble borders of a rudimentary bedroom set-up. Promotion works the very same way. Sure, it's easier to have a substantial budget for promotion, access to specialist publicists, friends in high places, media connections, or anything else that makes the process easier, but to dismiss the possibility of success independently or with a modest budget is just plain wrong. Creativity, focus, dedication, planning, attention to detail, and self-belief can be enough to make serious waves in the dance music industry, so we're going to untangle the details and understand what we need to do to stand out from the crowd.

Roles in the dance music industry

Before you proceed with launching press releases and PR campaigns for a new project, it would be smart first to understand the kinds of people that you're likely to come across throughout your promotional endeavors. This list is void of music production roles such as producers and mastering engineers and is primarily focused on people you'll encounter in the promotional phase of your projects. New roles are appearing all the time as the online media world evolves, but this sample A to Z should serve as a fitting starting point.

A&R

Employed by record labels and tasked with finding the next big thing, an A&R is usually the person your demo submission ends up with and will often be the one who decides if you're getting a record deal or not.

ARTIST MANAGER

A good artist manager is like having a second paid or arms that takes care of business. It is the manager's job to source opportunities for their talent roster and ensures that deals and contracts are as desirable as possible.

BLOGGER

Bloggers are the modern day journalists of the online world. With the rise of online publications, some dance music bloggers have followings to rival the most prominent artists and labels in the business, so don't underestimate their influence.

BOOKING AGENT/MANAGER

A booking agent is tasked with sourcing and booking shows and bartering the best possible fees for artists who perform live. The manager will take care of terms and contracts for shows and liaise with promoters on matters such as travel and accommodation, logistics, and an artist's rider.

CONTRIBUTOR

A contributor is a term that's grown in popularity in recent years. These are usually entry-level writers at a publication, interns, or third-party writers. They are good contacts to have for startup projects because they are often as keen for new connections as you should be.

DISTRIBUTOR

A good distribution service will ensure that all of your releases make it onto the world's most popular download stores. Manually distributing your releases to hundreds or thousands of stores would be a painstaking task and is best left to the pros.

EDITOR

In traditional print press, the editor is the decision maker and the last stop on the production line of a feature or article. It is the editor's job to direct the overall course of a publication by conveying ideas for features that are usually designated to staff writers or freelance contributors.

IN-HOUSE WRITER

Making up the bulk of the content on most publications, the in-house writers are your typical journalist types who are usually employed directly by the press outlet that they contribute to.

LABEL MANAGER

Usually the owner, the label manager has the final say on new signings and is the driving force behind the day-to-day operations of a modern record label. You'll encounter label managers when signing with a new imprint, planning promotional campaigns, and dealing with the collection of royalties and sales data.

LEGAL REPRESENTATION

Lawyers who specialize in the music industry can assist in contractual negotiations, registration of trademarks and copyrights, business affairs, legal counsel, sample clearance, work permits, and much more.

MUSIC DIRECTOR/PLAYLIST MANAGER (RADIO)

Ever wondered why the same songs are played all day on the radio? Music directors and playlist managers decide what makes it onto revered playlists that most daytime DJs and hosts are contracted to play.

PLAYLIST CURATOR (ONLINE)

The rise in Spotify and other online streaming media players has spawned a new sector of roles in the music industry for playlist curators. Much like their radio counterparts, they are the keyholders to playlists that are rapidly becoming the most powerful method of exposure for new music.

PROMOTER (VENUE)

Promoters are the minds behind events and nightclub calendars. A promoter's job is to book artists for shows and handle the promotion of events to ensure that venues are busy. They will regularly deal with booking agents to confirm headline talents for live shows.

PR/PUBLICIST

A reliable PR manager will take charge of the publicity and promotion of your project. They will be tasked with overseeing pretty much everything discussed in this chapter. Their quality media connections mean that they can secure an artist features, interviews, reviews, and more in the press. When you need to build a following, a publicist can be of great assistance.

PUBLISHER

A publisher is where you will assign the copyright to your music. In return for a percentage, they will ensure that artists and labels are paid due royalties when your work is broadcasted. They monitor the world for radio plays, TV coverage, and all other commercial uses of music and recoup due to royalty payments for these airings.

TOUR MANAGER

Reserved for larger scale projects, a tour manager oversees the direction for a calendar of dates for an artist. Touring in blocks is one of the significant sources of income for established artists on the road. They will deal with managers (artist, booking & PR) to compile successful tours that can be local, domestic or international.

As you can see, there are a lot of roles in the industry beyond the production of music. Traditionally, it was a popular notion that the music industry was a risky career choice with a shortage of jobs but in the modern world, this is no longer a truthful statement.

PR and promotion basics

PLANNING A PR CAMPAIGN

When preparing a campaign, one of the smartest things we can do is take a look at successful examples in the genre we're working on. More often than not, the releases currently topping the sales charts will have undergone a structured campaign based on the previous experiences of the artists, label, management and PR team on the job. Their collective expertise and efforts will culminate into what we know as a campaign, and by taking a minute to research these releases, we can learn a thing or two about their methodology.

HOW TO WRITE A PRESS RELEASE

Your press release is your story, and you should write this with your journalist hat on. Before you write a single word, load up your favorite online dance music news site or blog and jump into the reviews section and get reading. Pay attention to the paragraph and sentence structure and learn the vocabulary used to creatively describe the sounds and moods that can translate to your project. Take inspiration, but don't plagiarize and you'll quickly pick up the correct form of how to write about music on an imaginative level that engages the reader. Put yourself in the shoes of a writer at a magazine and dedicate the same patience and effort on writing about your music as you do producing it. You should strive to be as expressive, flowing, and persuasive with your words as you can manage as this is one thing that when done well can compete with any publicist or writer's work. No one has a deeper connection with music than the creator of it, so if you can develop the skills to translate your musical ideas and inspiration into impassioned text, your press releases will shine.

Tips for writing engaging press releases:

- Try to remain between 250 and 500 words per press release. Any less looks lazy, and beyond that limit, you'll likely lose the interest of the reader.
- Use a headline to draw interest.
- Go easy on the stacked superlatives. People get instantly apprehensive when we tell them that something is going to be the most unbelievable, incredible, astonishing, fantastic, world-conquering, monumental, and overpowering thing they're ever going to hear.
- Send press releases as PDF files, not Word documents. This creates a professional image and looks more official.
- Sign off with a call to action, i.e., 'listen to the preview of the upcoming release below,' or link to a store to buy or download what we're selling.
- Don't forget to link readers to your website and social media profiles that we've set up.
- Accompany your press release with a personal email and pitch. Be as friendly and interesting as you can.
- Never, ever blast a press release to 100 different sites or writers on CC in the same mail. You'll get blacklisted quicker than you pasted their email addresses into your mail client.

PRESS RELEASE DISTRIBUTION

Now that we have a professional press release ready it's time to get it sent out to our targets. Much like the preparation for writing, our process for sending starts with research. Find a new release in the genre that you're working with that you know is doing well and begin some searches on the likes of Google and social media networks. You'll find most popular releases are being well documented in the online dance music press. Follow the links through to the articles, and in most cases, you'll be able to find the name and contact details of who made the post. Note down the contact and publication name and begin to build a list of press targets that we think could be interested in what we're promoting. The key here is to try our best to source the writer with priority, but if that search is unfruitful, head to the contact page of the publication, and you'll usually find an inbox address for submissions. The benefit of going directly to the writer is that we have clear evidence that they are writing about music that is similar to what we're promoting, this raises the likelihood of sparking their interest. If a publication has multiple contributors, they likely have a specialist interest in one or more subgenres. By going straight to the people who appear closest to related to the music that we're working with, we're improving the probability that we'll be sending them something they like.

When it comes to sending the emails, start off with a personal message. Editors and contributors are inundated with proposals every day and are inclined to skip through emails that look robotic and spam like. Tread the line between professional and personal, and you'll find that response rates will improve as you develop your skill set. Keep records of the kinds of message that you send and develop an understanding of what works and what does not. Borrow some logic from the marketing world and A/B test different press texts and writing styles and tweak the formula over time to find your perfect style. If you get five out of ten responses from one format and one out of ten for another, you can see what people are responding better too and use these data to crack the code.

BUILDING YOUR INDUSTRY NETWORK

Starting out as a producer especially can seem like a solitary and anti-social lifestyle, and in some ways it is. When we first catch the bug for creating music it often leads to extended periods of time locked in dark rooms and rarely much else. Live that life for a couple of years, and you'll find that all of this social interaction, pitching, contacts, and networking business covered in this chapter feels a bit out of our comfort zone. Unnatural as it might feel at first when the time comes, and you want to take your music to the masses, the music industry, like most, is a social business.

Building a network is a lifelong project rather than something you do in the evening, but understanding this early on will pay dividends in the long run. How you build your industry network is entirely up to you, but rest assured that no one ever made it in this business without working with other people in one way or another. A large-scale dance music project is not a one-man job.

Use contact books on your computers, store as much detail on your contacts as you can get your hands on and use this info to build rapport. Backup these contact lists on your cloud and even consider keeping a physical copy. In ten years' time you'll have a black book of pretty much everyone you'll ever need to know, and these connections will make your life in the industry a much smoother ride. Your contacts and connections are one of your most important assets so look after them as this side of the business is not an area you want to be starting from square one at more than once.

ONLINE PRESS

Below is a starter pack of leading online publications, blogs and press spots that feature dance music news and content. Don't stop here, however, as there are hundreds and thousands out there that should be targeted by up and comers making their first efforts towards securing feature coverage. This list included a selection that covers a wide range of subgenres so be sure to do some research into which are relevant to your music before you go ahead and reach out with any proposals. Using the procedures and concepts covered in this chapter and the production skills developed with Rick's help throughout this book, you should be well on the way to being ready to take your music to the press:

- *Attack Magazine*
- *Caveman Sound*
- *Daily Beat*
- *Dance and Rave*
- *Dance Rebels*
- *Dancing Astronaut*
- *Data Transmission*
- *Discobelle*
- *DJ Mag*
- *DMC World Magazine*
- *Earmilk*
- *Eat Sleep EDM*
- *EDM Chicago*
- *EDM Joy*
- *EDM Maniac*
- *EDM Nations*
- *EDM Sauce*
- *EDM.com*
- *EDMBoutique*
- *EDMDroid*
- *EDMTunes*
- *Electro Wow*
- *FACT Magazine*
- *Feel My Bicep*
- *Future Music*
- *HousePlanet*
- *IDM Mag*
- *iHouseU*
- *Magnetic Magazine*
- *Mixing.DJ*
- *Mixmag*
- *Music Crowns*
- *THE Music Essentials*
- *The Music Ninja*
- *Noiseporn*
- *Noisey/Vice*
- *One EDM*
- *Pigeons & Planes*
- *RaverRafting*

- *Resident Advisor*
- *Run The Trap*
- *Salacious Sound*
- *Smash The Club*
- *SOH Blog*
- *Soundplate*
- *Stoney Roads*
- *thatDROP*
- *This Song Is Sick*
- *ThisSongSlaps*
- *UKF*
- *We Rave You*
- *XLR8R*
- *Your EDM*

THE IMPORTANCE OF COMMUNICATION

Contrary to some public scrutiny, most bloggers, writers, and editors aren't mean and are usually good people. They do what they do because they love to listen to and write about music, so your submissions will be welcome if you adhere to their expectations, which are fair. Don't forget that we're essentially asking for a favor when sending pitches, so treat people with respect and take the time to leave a personal touch. Unless you're paying for sponsored content, these writers owe you nothing and talking to them like they do is a surefire way to fail in your campaigns.

How you communicate and present yourself in the dance music industry will dictate a lot of your future achievements and disappointments. Treating your industry contacts with respect from day one is one of the smartest things that we can do. From interns up to editors and managers of every shape and size, having a reputation for being a decent human being will make others gravitate towards you. The intern at a magazine today could end up being the editor in years to come, so don't miss out on chances to build your network. There's no better approach to that ongoing managing task than being a nice person to deal with. It's no coincidence that dance music legends like Carl Cox who walk around with a beaming smile and vibe that people want to be around remain at the top of the game from decade to decade.

MANAGING YOUR PR BUDGET

Having a budget can make a big difference to your options for promotion, but it is not the only factor. Here we will give some insight into what you can reasonably expect for a variety of costs. These figures are estimates based on averages for one release:

£100. Reach out to a selection of your favorite blogs and inquire about sponsored posts. Be upfront about your limited budget and ask if they can offer any form of feature coverage for what you have available. You may be under budget in some situations, but you might get lucky and find an opening. If this proves unsuccessful, try up and coming blogs that may be able to cater for your budget.

£250. For around this figure, you could hire an introductory level PR campaign to handle promotion for a release. Alternatively, you could likely book a couple of sponsored features on medium-sized online publications.

£500. At this point, your budget would be sufficient enough to book a large press feature with a key media site or a series of mid or lower level bookings. Again, you could also find a professional PR service at this rate.

£1000+. At this point and above, you would be best managed by a PR agency. A well-connected publicist will have working relationships with press contacts, which means they will be able to book sponsored content for you at lower rates than you would be charged. Aside from the budget their expertise and influence will ensure that you're well covered in the media to raise your profile.

DJ PROMOTION AND TASTEMAKERS

Aside from the press side promotion, the online world has made the distribution of our music to DJs across the globe a widely available option. Gone are the days of having to press vinyl promos or shipping CDs around the world in the hope of catching the ear of our favorite DJs. Today's online world has simplified the process and given us both platforms to host or send our promos and services that do the promoting for you.

Services such as Inflyte, Label Worx, and FATdrop have fantastic functionality that offers a range of service that makes sending high-quality promos to target DJs a breeze. They improve every year with new advances in piracy protection, advanced database management, and in-depth reporting, which makes the process as stress free as possible. The model is mostly the same with these platforms, and you'll need to build your contacts for DJs, but this ensures that your promos are tightly managed and circulated.

There are other hirable services out there from radio pluggers and promotion services such as Power, which specializes indirectly pushing releases to top radio figures and leading DJs or others such as Reaktion, which is popular with up and coming artists and labels.

REPORTS AND DATA

Keeping solid records is a great way to understand the reach and success of your campaigns. Using spreadsheets in Excel or similar software to record movement on your promotional work is a great way to learn what works best for you. A press report is also a valuable document to send to record labels, managers, and other industry peers when you need to convince people that you're serious. An A&R who sees your recent promotional activity is more likely to view you as a desirable asset when you're sending a demo. As we've touched on previously, there's more to progression in the music game than just good-quality music.

THE FUTURE OF DANCE MUSIC AND PROMOTION

It is important to remain present with the concepts and trends of promotion in the music industry. New platforms come and go as technology, and public demand evolves. This means that five years from now what we use to promote and consume music will not be same as what it is today. What we will know in ten years will make today's platforms look prehistoric. As an artist or music industry professional it is vital that we remain at the forefront of these changes.

In the near future, it's highly likely that DJs will no longer pay for music in the way that we do now. We can use the way that music consumption has evolved to assume that a subscription service is just around the corner, which will enable DJs to have complete and instant access to the entire music catalog via a cloud. These tools will have the functionality of Traktor or Serato, which will offer something that the likes of YouTube and Spotify currently do not. When this technology arrives, that will be another huge dent in the sales of music. DJs are one of the last groups that still pay for music on a per release basis, but this will become obsolete before long.

Streaming will continue to evolve and open up new ways to consume music. Brands such as Boiler Room are taking the nightclub experience and delivering it to the comfort of your home, and rapidly expanding startups such as Goat Shed are bringing the radio format into the digital realm by bettering the audio-only nature of radios by integrating social-media-based live video streams. This merger of old technology into the modern world will only accelerate, and the future music industry will only support media that remain consumable on digital domains.

Sending a press release by email will likely become defunct and will be replaced by new technology and platforms that will simplify the process for the dance music press and artists or labels alike. Every year another longstanding magazine switches off the printers for good and goes digital, and this trend will likely continue. Before long the thought of going to a shop to pay for a magazine will seem preposterous as the possibilities of the digital realm will become so superior that the format will likely become redundant.

The changes are rapid and will continue to accelerate further still as we integrate our lives into cyberspace. Nobody knows what the future will look like, but what we do know for sure is that change is inevitable and those that want to conquer the future will need to be adaptable and ready for whatever to industry and technology brings.

Useful links include:

www.label-worx.com (distribution, demo management, artist and label services)

www.mattcaldwell.co.uk (PR and artist development services)

www.1001tracklists.com (DJ tracklist database)

www.fatdrop.co.uk (promo platform)

www.power.co.uk (DJ promo service)

www.inflyteapp.com (promo management tool)

www.reaktion.net (DJ promo pool)

www.knightpublishing.com.au (publishing).

CHAPTER 38
A DJ's perspective

Most EDM today is just weaponised elevator music.

Rick Snoman

Electronic music producers and DJs share a symbiotic relationship with one another. Without DJs, our records wouldn't reach a majority of listeners and, without records, DJs would have nothing to play. With this in mind, it seems that there is no better way to close the book than to interview a DJ to gain their perspective and advice on DJing. For this edition, I asked Tom Rogers to submit a chapter for the book. Tom is a certified audio engineer from the Institute of Audio Research, has produced records played by Armin Van Buuren and Blasterjaxx, and played dozens of shows including a year-long residency in New York.

Today's DJ has a wide variety of options available to her. The birth of DJ culture and subsequent shaping of the DJ set-up occurred around the time of the first, very limited, personal computers. As we have gone from the Atari to the MacBook, the options and power afforded by digital components have reshaped our ability to DJ several times. First with CDJs, and now with hybrid and controller set-ups. Our ability to modify recorded music has never been more customizable and expansive. It's a great time to be a DJ.

This first section acts as a primer for your various options in hardware and software. It gives an overview of their features, drawbacks, and potential reasons you might choose them. There are also some notes on various mixing and production techniques, should something in particular appeal to you. There is no 'best' set-up, only different ways to access and manipulate records. Most of them use the same key components, mixers, EQ, 'decks' or turntables, and faders.

VINYL

Vinyl is the 'original' set-up (after tape decks). It is considered the most difficult to use, and the industry noted Technics brand would run you a pretty penny.

Many clubs no longer have a pair available for you to play with, so if you do choose this path, you'd have to bring them yourself. Although this is less true of Hip-Hop clubs, which put more emphasis on scratching. Many DJs, such as A Trak, Laidback Luke, etc., who enjoy this style of performance will sometimes use vinyl. Even then, the most common vinyl set-up is a hybrid, between the turntables and control disks for a program like Serato.

What makes vinyl hard is that it lacks any visual representation, as the BPM of the track has to be detected by ear. There is no way to save cue points, without marking the track. Vinyl records tend to 'warp' over time, or if stored in a temperature-varied environment, which can affect the sound. Replacing records is a common but costly procedure.

This medium also limits the bass levels of old records, or the needle would skip and stop the record, freezing a dancefloor. You will find that while some enthusiasts claim the sound is warmer, there is a distinct lack of power when compared to modern commercial music. Needles for the turntables also have to be replaced. Many DJs will not allow you to use theirs, and expect you to show up with your own needles – so be aware of this if the club does happen to have vinyl tables present.

All in all, vinyl is respected for being the turntable we saw most when the art of DJing was in utero. It is considered nostalgic and respectable. Ultimately, digital technology has made many of its defining features obsolete or provided a cost-effective alternative.

Still, the vinyl set-up is the set-up of the purist. If you are looking to learn the hardest or oldest standing form, this one is for you. I do not think it's necessary as a starting point at this time unless you are interested in scratching or another advanced form of *turntablism*.

CDJS

CDJs are the common, 'traditional' set-up. Many clubs will have them ready to go when you walk in, so there is the benefit of essentially having your home set-up wherever you go. Many people opt for a customized, 'all-in-one' version of the Pioneer mixer. Namely, the XDJ controllers, which have very similar functionality at a fraction of the cost of a commercial set up.

The CDJs introduced many of the features all digital equipment now uses. They enable the DJ to view a record's BPM, enabling beat matching with expert precision. Digital cue points allow you to program and jump to 'bookmarks' within a track. A visual digital waveform allows you to keep track of the overall arc of the song. Not to mention feature that means the CDJs will start blinking when you run towards the end of a record, in case you were distracted.

However, one significant drawback to the CDJs is their increasing cost and inherent inability to keep up with modern digital features. It takes much longer for

Pioneer to incorporate similar technology into its products, and then they will often be available in only the newest versions. As it adds touch pads, sync, and other features that have existed in the controller realm for several years, there is some ability to customize sets.

The main advantage of DJing in the modern world is comfortably mixing two of your songs in a club you've never been to before. The traditional set-up is so commonplace that once you get a handle on the CDJs, you no longer have to worry about the gear set-up at any respectable club. This level of comfort is a major selling point for new DJs and a big reason why many will recommend knowing how to use them.

TRAKTOR

For artists who understand the value of CDJs but wish to have a slightly more interactive performance, Traktor is the next step. Most of its interface is set up onscreen like a traditional CD and mixer set-up, along with several live decks and various effects panels that can be customized for extra mobility.

Many Techno and Progressive House artists, who are interested in creating a tailored experience to the room each night, have moved to Traktor. The ability to add additional percussion loops, vocal clips, and have everything automatically aligned in the time grid allows for more expansive, on-the-fly performances.

Traktor has 'main decks that are set up like a CDJ: you can see the waveform of the song, various cue points, and your BPM meters. What takes this to the next level is it's 'live decks,' smaller versions that are ideal for throwing on a loop of drums or a vocal sample. It also has a sync feature, allowing you to bypass beat-matching altogether and instantly trigger these effects in time.

Many artists from the era of CDJs enjoy this because it can be daunting to match several extra components while managing the song transitions. The extra time you buy yourself from not having to beat match manually allows for combinations and overlays that would be impossible to set up in time by the human hand.

ABLETON

Mostly used as a production DAW by many artists, Ableton is an extremely flexible live performance engine. You can create both simple configurations similar to the CDJS and Traktor, or you can get more advanced.

Because Ableton is a full audio workstation, you have the ability to add synthesizers and audio editing techniques to your live performance. Provided the necessary warping treatment is done ahead of time, virtually any clip or instrument can begin in time, on the next bar, when triggered. This takes things a step further than Traktor, as it allows you to potentially recreate or remix your songs on the fly with your full production arsenal.

A feature none of the other methods has is that its BPM can be changed on the fly by using the keyboard. Ableton enables you to control the pitching and altering of sounds within seconds. This means that you have much more control than previously experienced when using the turntables or CDJs, therefore able to change BPM with ease. If you're interested in jumping quickly into a rock record, then speeding it up to 128 and mixing a new record in time, this feature could be vital.

A consideration you need to make when using Ableton in this environment, however, is that it is possible for the software to crash or overload. For that reason, you may want to use a high-quality external soundcard, and look into other improvements, such as linked computers or additional RAM, if you would like to run a considerably large set-up with external synths.

COMMON CONTROLLER SET-UPS

Whether using traditional turntables or a controller, you will usually need a mixer. These come in different sizes and allow for different numbers of connected devices. The most common are two channel mixers (for two turntables), and four channel mixers (for heavy mixing, or a quicker transition between DJs). Mixing can be done via the DJ controller or the club mixer. 'All-in-one' set-ups, which contain a mixer and two controller surfaces mimicking the 'decks,' may or may not need to connect to the mixer as they have their own soundcard. For some of these set-ups, their internal mixer can plug directly into the sound system.

There are other controllers we have not covered in this section, but they all follow the basic premise and features listed above. Serato is a favored component for vinyl set-ups as mentioned earlier. Ableton has a hardware controller (Ableton Live), and Traktor has its own version of controller as well. Many third-party controllers are created for further customization, but it's essential to read the reviews on these, and make sure they will navigate through your chosen platform in the way you like. It's not a good feeling to purchase a new controller that works brilliantly, only to find out it's lacking a feature you've come accustomed to having in your performance.

CHOOSING YOUR SET-UP

There was, at a time, a 'right' and 'wrong' answer for this. Perhaps five or six years ago, my best advice would be to ignore all options and choose the CDJs. As beatmatching and traditional mixing have faded away from being the be-all-and-end-all in club circles, and the rising popularity of producer-DJs has embraced controllers in many forms, this is no longer the case.

Now, while it's useful to consider how the equipment can be positioned within the booth, your range of performance techniques and ability to fit them in the performance area has significantly widened. It's important to look at what kind of performance you want to deliver; not everyone is interested in continually triggering samples and being watched as a guitarist might be.

At the same time, some artists feel uncomfortable mixing merely two records on CDJs. This is particularly an issue that producers reading this manual might encounter, as the production process is decidedly more technical. Finding ways to pass the time without over-doing the effects is an issue I have had when considering traditional set-ups. What it comes down to is knowing who you are as an artist. With world-touring acts using all of these set-ups at this point, there is no 'one way' you must learn.

Consider your music: some songs are not meant to be blended with others, or structurally have too much going on. The result can feel forced and unpleasant. Other times, livening up more extended sections or introducing additional drums can help adjust to the current size of a dancefloor.

BASIC FUNCTIONS OF DJING

Volume faders and crossfaders

These are the most used and simplest controllers on your DJ mixer. A volume fader is a vertical slider, which controls the level of the song. 'Fading out' of one song requires you to move the fader downward in a controlled motion.

A crossfader is a special kind of fader, which moves in a horizontal motion instead. They are usually at the very bottom of your mixing surface. When you have the crossfader all the way to one side, *only* sound from that side will come through the speakers. When you move it to the center, you will hear both audio sources at equal volume. As you slide the fader, you will hear one track disappear and the other gradually come in. This is the key component of a DJ transition. Some mixers have multiple algorithms you can choose from for how the volumes change as you move the fader. Others are mostly the same as lowering one volume fader and raising the other in time.

It's worth noting that I've seen many professional artists prefer to use the volume faders instead of the crossfader. The main reason for this is since new dance music is so loud and polished, you may leave the full volume of the first song up for quite a while. A conventional technique is to inch up the second song while using the EQs, and quickly fade out the first track when a pause occurs (drum fill, breakdown, etc.).

EQ

The three-band EQ featured on most mixers is a simplistic version of your usual EQ. It will control fixed bands — the 'high' range, 'midrange,' and 'low' range.

Typically mixing two dance records, you will mostly use the low EQ. Many DJs will wait for a part where the bass and kick are not playing, and 'kill' this knob by turning it close to or fully to the left. They will, at the same time or shortly after, bring in the lows of their new record. Playing the low end of both songs is usually overpowering, and crowds out the rest of the sounds. Occasionally, if the kicks are in perfect time, you may experience a phenomenon called *phase cancellation*. This would cause the kick to 'disappear' and become inaudible, which is not desirable for the dancefloor.

Use of the midrange allows you to balance the melodic portions. If they clash or appear 'busy' together, you can lower one significantly. Sometimes the new track can come in with little or no mids if the crowd is enjoying the last song's exiting sequence. Other times, you'll want to kill the old track's midrange immediately.

The high frequencies are similar but rarely needed. Both these styles are more to taste than the adjustment of the lows, which is mainly dictated by the kicks.

Intros and outros of tracks tend to be structured for your transition purposes, and expertly crafted ones require very little work. As a producer, your job is to make these sections as distinctive as possible, while keeping it straightforward, so the DJ can mix with seconds to spare.

Beatmatching/sync

This matters less when mixing pop or Hip-Hop music, but in dance, the tracks will usually need to be beatmatched. That means controlling the time of the songs so that the kicks play over one another at the same pace. This will require adjusting the pitch fader on traditional turntables, or the tempo/BPM slider on digital set-ups.

The act of beatmatching first involves getting the songs to move at the same speed. Then, you will want the first beat to be lined up — this is usually done out of the audience's view through the headphone cue. By changing your headphone cue to the 'mix' side, you will hear the incoming song while the crowd will not. This lets you get things perfect before introducing them to the speakers.

Most DJ headphones are bulky because of their need to isolate sound in crowded rooms and do a good job of blocking out the noise. Sometimes monitors (extra speakers) are in the DJ booth facing the DJ to help manage the transitions. This is because distance and speakers facing away from you affect our perception of hearing. You can also balance the monitor balance if you prefer.

Many DJs use the cue points and added functionality of digital mediums to keep triggering their second song, until it matches in time. CDJs allow you to do this as well but require some light nudging of the turntable to make things precise. The 'sync' feature on digital set-ups and newer CDJs, provided you programmed the beat grid ahead of time, will do this automatically. The speed and triggering of the second song will conform to the first.

Before digital, getting this right was key to having a decent DJ transition in dance music. The kicks falling out of time creates a galloping sound, or train tracks, which is sometimes referred to as a 'trainwreck,' and it really sounds unpleasant.

Cue points and looping

Cue points are audio 'bookmarks' within your song, which you can program and trigger with the push of a button. This allows you to jump to key parts of a song quickly, and make real-time decisions with the audience. If the vocal is doing very well and everyone is singing along, you may want to have mapped the start of it to a cue point. Then when it's almost done, triggering this will cause a replay, much to your audience's enjoyment.

It's also a good idea to have the intro, and the outro of a track cued up, in case you are tight on time and needed to even out a transition. Alongside beat matching, *phrase matching* is the alignment of musical sections (intro, verse 1, chorus, outro). At the basic stage, you want the intro of one track to play over the outro of the other — creating a continuous overlay into the next song. Many times, the cue points are used to jump to these sections if the song is not a hit with the dancefloor, or if you are interested in mixing between big hits quickly for energy.

Loops are a close cousin of cue points; they do exactly as they say. A looped section will repeat indefinitely until you cancel the loop. Many artists will use looping to 'save' a transition when they missed their entry part. If you were four bars late, for example, using a four bar-loop once and then ending it will correct the mistake. If an audience is dancing to a drop that is about to end, you can keep that section going with a loop. They are usually controlled by the number of bars (four, eight, 16, 32). With the digital controllers, loops can be set up to trigger perfectly in time. With traditional set-ups, you will have to attempt to do the triggering as quickly as possible, or the track will be slightly out of time. A poor manual loop may sound 'choppy' if it cuts out too early before or after a kick drum.

Mixer effects

Most mixers and software come with effects, and the ability to either change them out or add additional effects units. This is where you have access to some of your production elements but on a very zoomed out scale. Reverb, delay, flanger, and other effects are all here for you to quickly add to the master output (or one channel).

These can diversify your transitions or allow you to amplify a particular moment. Many DJs enjoy using the colored filter during build-ups. It has a high resonance, which creates an entertaining noise if turned quickly. Using that will filter out the lows and build more of an impact on the drop, as well as engage the audience with the sound of the resonance. DJs also experiment with reverb, delay, and flanger at this time.

Mixer effects often affect the master output — meaning you can use them to stop or exit a record and still have the effect. Some DJs will use reverb in large portions, then finish the first record. The reverb will hang after the song playing is no longer in use. That can allow them to talk to the audience, build anticipation, or quickly change between BPM (as this avoids needing to match the songs in time). Other versions of this involve smart or impressive ways to end the set, such as slowing down the final song until people understand the party is over.

An example process

Here's a quick walkthrough on a basic 'A to B' (Deck 1 to Deck 2) transition, to expand your understanding of these concepts.

Deck 1 is reaching its outro, and the song is almost over. We use our cue point in Deck 2, which is set to the first beat of the track. We beatmatch or sync the songs, and launch Deck 2's first beat as the outro of Deck 1 begins. At that time, we start using our faders. The low EQ on Deck 2 is entirely cut, and we're slowly inching up the mids as the song develops. When Deck 1 has a moment that the kick is not playing, we use this 'gap' in beats to kill its bass EQ.

We then filter in our Deck 2's bass with controlled motion, dropping in seamlessly. At this point, we can start lowering Deck 1's fader and bringing our EQs to full level on Deck 2. When Deck 2 has a breakdown coming, or a key sound taking up the listener's attention, we quickly complete the crossfade. Deck 1 is now out of audience listening and can be loaded up with the next song.

HARMONIC MIXING

How it works

Harmonic mixing has two main elements: the DJ's knowing the key of the track he is playing and also having the knowledge of key compatibility. The Camelot wheel was created by Mark Davis. It uses the information off the circle of 5ths and is a visual representation demonstrating which keys are compatible. It is considered possible to use the Camelot wheel without any musical knowledge.

In the purest form, harmonic mixing is following this wheel either clockwise (increased energy) or counterclockwise (tension, anticipation). You can use key-analyzing software to label your songs by key and learn the wheel diagram with no knowledge of music. Just think of it as a roadmap that allows the musical parts of the songs not to clash. This creates more pleasant transitions, and, on the best ones, the songs will complement one another, creating a new moment in the transition.

Breaking the rules

There are more ways to cross the wheel than to follow it in either direction simply. For example, a straight jump is possible between any key (A minor, for example), and its sharp equivalent (A♯ minor). A minor uses no sharps or flats, while A♯ minor uses all of them. This contrast, while less pleasing than a 5ths, generally tends to 'work' with mixing music.

Some DJs prefer to move in 3rd or 5th intervals (A minor to C minor, or E minor). There are expansive ways to mix keys, but these versions are the most common for DJ transitions. You're free to ignore this and mix what 'feels right,' however, harmonic mixing has become standard at this point. The songs have developed musically, and it is easy to hear when the bass of one track clashes with the new song. These can feel forced and jarring. So it is a good idea to at least understand how to travel the wheel easily.

Key-analyzing software

Many software companies offer key analysis (and it's becoming an added part of some digital mixing media). The most popular is *MixedInKey*, which will merely scan a folder and rename your songs in its custom format (8A, 6B, etc.). This is perfect for the non-musical DJ, as it replaces many of those letters with numbers. A is minor, B is major in the MixedinKey labels. All of their numbers correspond to one another, meaning that I can move from 8A to 7A, and that will be a 5th. I can also move from 8A to 8B for a major to minor key change. This can be helpful in quickly following the patterns for beginners, as all you have to do is move in increasing or decreasing numbers to have harmonic mixes.

'DIGGING' FOR MUSIC

This is a term coined from the time where vinyl was the most popular medium, and you had to physically move through or 'dig' through crates to find various records. Other terms like pulled, selected, or 'finding gems' all describe this task. In addition to playing your music, DJs in the modern era have the luxury of selecting, sampling, or editing together any given sounds in existence. Interesting spoken word sections, radio interviews, or other sound bytes may be incorporated in addition to the songs themselves.

Here is a short list of ways to '*fan out*' your search for new music. Some genres and artists feel it is more important to deliver fresh, undiscovered tracks than the hits of the moment, while others do not do this at all. If you produce albums and are looking to perform mostly your music, endlessly hunting for the latest techno records may seem unnecessary. If you do not produce yet at all, diversifying your songs from the other local acts can help you stand out and secure more gigs:

- 1001Tracklists.com. Regularly updated tracklists of famous DJs on tour, for their radio shows, or for festival performances.
- DJ Promo Pools. Once establishing yourself as a producer, you will start to receive promo mailers from labels you've signed with, DJs, and other sources with music to download. Usually, they require you to give your thoughts on a song in a short, few sentences and rate it before downloading.
- Manual searching. You have an artist that you like so you go through the full catalogs, of each label they have released on. It's possible multiple artists on the same label have similar sounds, which can be added to develop music in this style.

Beatport. Mainly a music retailer, many will 'dig' using Beatport by going to a genre, and sorting by 'all new releases.' This will show you most music on the website in that style, and keep you up to date.

BUILDING A SET AND CROWD READING

There are various styles of set building and mixing, mashups, slow build, non-stop energy. These can be considered when listening to artists and applied to your mixes. Sometimes, choosing a different method of mixing technique can be what sets you apart. You will notice certain genres or sound structures work better with specific mixing techniques than others. However, there are no hard and fast rules. The commonly done version is not always the best, and the best version does not always work in the room you are currently DJing in. Knowing how to make these different kinds of decision and deciding what your preferences are can help you achieve the emotional reaction you aim to create. Here's a typical example:

Slow build: Same energy level. The new track will have slightly more elements, for example, added percussion loops or the subtle introduction of a vocal chop. The speed is mainly characterized by how slowly these elements are being added, across how many songs, across how long the mix is. Progressive House and Deep Tech artists live in this space and can create long, hypnotic sequences by selectively bringing in new elements.

Opening (the art of opening)

Newer DJs will often take a moment to understand the power and value in creating a low-energy, slowly building opening set that adequately prepares a room for the headliner. Most events will expect a certain level of this knowledge, some more than others. It is your job to understand how to create an entertaining set with a limited energy level. Opening sets require a good deal of preparation as far as finding similar textured songs (if the leading artist plays very dark, we do not want to start with ambient, beach-type records). They also require some flexibility based on how many people are actually on the dancefloor. Ramping up the energy when the room is not leaving the bar yet is a surefire way to lock them in the corners.

One of my favorite opening tricks is to grab a long ambient track — I tend to choose Jean Michael Jarre's 'Waiting for Cousteau,' or an IDM record with no drums. Because I use Ableton for my performances, I can trigger samples on command. There's no use 'wasting' great Deep House songs early in the night if nobody is currently present to hear them. So I will keep the ambient track running until the first group steps into the room and onto the dancefloor. At that point, I will deliver on their anticipation of something happening, and trigger a kick drum sample. Depending on their reaction, you can play with this as long as you like, and then bring in the first drum beat. The result is a more active dance floor from the start, because of this form of musical interaction.

Being able to tell when elements are needed or taken away is *crucial*. For example, if a crowd later in this same night reacts well to a more aggressive bassline, I may want to ignore my initial plans and play more tracks with that energy. If it looks like that has worked several times but is starting to get old, you want to be able to anticipate that and have the new element songs coming in *just* before they would get bored. This is something many DJs will refer to as 'playing to the room,' or 'reading the crowd.'

A significant error new DJs make is playing songs from the headliner's sets, or songs they have produced. Even if they are not planning on playing them tonight, this can be seen as bad form and should be avoided. This is part of a specific code or etiquette affiliated with the opening DJ set, and it is wise to steer clear of it. There are very few scenarios in which the headlining DJ will enjoy hearing his or her song when they are not the ones playing it.

Peak time/headlining sets

The peak time set faces different challenges. Keeping the audience engaged is still essential, but there are different dynamics in play. At this time of the night, the room is here to see you do your thing. The most significant DJs in the world are also producers — so they face the challenge of trying to include their big hits, and spreading them over the night to maintain interest. If you find yourself in this position early on, you may not have too many of your own tracks. The challenge here would be trying to find similar songs, which raise the energy level and showcase your original material.

If you have approached this from the other angle and are looking to be known for your mixing ability, you may want to dig deep and find obscure records to begin. Creating new energy and cultivating an open-minded surprise from your crowd can be an excellent place to start the night.

For these reasons, there's more wiggle room with building sets. They do not necessarily have to be the slow build of an opening set. Finding smooth ways to jump between different songs you've released over the years, for example, might be hard to do within your set time. Some artists may leverage the fact that the crowd is there to see them and take more significant risks. The music may stop, the artist may use a microphone, or there might be other sudden cuts between songs. Festival headliners may go as far as coordinating with lighting techs to create 'events' within the show, which emphasize a change in style.

Closing sets

What do you give someone after they've gotten everything they came for? The closing DJ's job is to figure out how to keep people on the dancefloor, people who are getting more and more tired and have already heard what they came to hear. There's no one way to handle this. Some DJs will purposely choose a collection of harder or faster music that is naturally higher energy than the headliner. However, it's possible to overshoot and select a style the room does not appreciate. When going into further hours of the morning, other closing DJs like to use tracks with lots of drums, emphasizing their danceability. When going this route, they may not use tracks with long breakdowns, and try to maintain a continuous flow of music for the room to enter and leave at will.

Unlike the opening DJ, the closing DJ may want to play some of the headliner's songs. Although there is a time and a place, you may find yourself at an event where they did not play a favorite song. This decision might entertain and please an audience who wanted to hear that track the most and encourage them to listen to what else you have to offer.

Last year I performed a closing set for a Billboard #1 artist in my genre. He had since changed his sound over the years and played a few of his older hits to finish the night. However, he did not play all of them — and the songs he did play from this era were during the last ten or so minutes. Having done my homework on the artist and knowing how the night would go, I was prepared to play mainly songs from that time. I knew people would feel nostalgic towards the end of the set and might enjoy hearing an expansion on that. The older style of music was also a bit faster, which helped maintain the energy of the room.

Many times, the headliner will not stay through the full night. If the artist is touring, it might be time to sleep before boarding the next flight. To keep to proper etiquette as we discussed in our opening section, I waited until he left the building to play the big hit he decided not to. This elicited a great reaction, didn't upset the headliner, and kept the night going strong.

Some artists may not appreciate this either way. It's essential to gauge reactions and make sure you're not stepping on any toes as a support DJ. Knowing when it's safe to bend or break a rule comes with time. Hopefully, I've given you some guidelines for when the moment comes to test the waters.

ORGANIZING YOUR MUSIC

It's important to consider how long it will take you to find a given song during the performance. There are times in a DJ set where you only have between 30 seconds and two minutes to decide what the next track will be. In this scenario, you do not want to be hunting through hundreds of records. Even on a USB, scrolling through in time to land that perfect transition can become an issue.

Naming systems

Sorting tracks by different energy, style, etc. can be helpful in quickly cutting into songs you need. Some artists will create additional folders for various parts of the night, for example, deep opening tracks, peak hour closing. Others may organize literal track lists, to be pre-practiced and mixed live, so the next song is always the next selection on their roster. There are varying opinions on how much of the performance should be undecided and on the fly, versus how much preparation is required to deliver a quality performance. Again, who you are as an artist can come into play and help make a lot of these decisions.

Energy levels

In Ableton, for example, I like to organize specific styles or subgenres of music by coloring the session clips. Each track has a clip with its title, some of my psy-trance records will be green, and I will clot a lot of green records together, then sort them by key for quick mixing decisions. Other times I will want to quickly see and jump to a blue section for more euphoric, uplifting sounds. Others have had success using features like song rating or adding icons (ex. asterisks*, xs) at the beginning of song titles. If you are to use one of these methods, it's essential to keep the sorting system at the front of the songs. Otherwise, as the name scrolls on a USB or gets cut off by a limited controller display, you will be unable to determine the record's sorting with enough speed.

Go-to playlists

This is your greatest hits, tried and tested on the dance floor, songs you've mixed a ton of times in practice sessions, and so on. Many times, the go-to playlist is labeled for the gig you're playing and is the best results from your practice session. You might want a master version of this playlist, containing the best of the last few shows, in addition to your tailored ideas for the night at hand.

SUBMITTING DEMOS

In this day and age, you will not likely be asked to go through this process. However, there will be times where a particular promoter or club would like to hear you play a specific style, and request a mix to approve. In this circumstance, your demo should be roughly an hour long. We are assuming the promoter will listen to enough to evaluate the quality, but may ultimately skim or reference the first 20 or 30 minutes. Sitting down to listen to prospective artists' sounds requires a lot of time, especially if they are considering multiple artists. For that reason, you may want to make decisions highlighting the earlier section of your mix. We also want to have enough included that if the listener *does* want to hear a full journey, they can get a clear picture of how you will deliver that.

SPEAKING TO PROMOTERS

What they're looking for

First and foremost, you have to understand what the promoter needs to validate a booking. Yes, your mixing has to be decent, but 'how good' it needs to be can vary on the type of night. Promoters are trying to break even or profit from their expenses on an event. Many promoters will want to know you can either outright sell tickets (some DJs frown on this, as it's seen as 'doing their job for them'), or at least bring a crowd of friends to the table.

Being present at local shows is a great way to get introduced to these people, and demonstrate by being friendly with people at the show, that you might be able to secure some sales for the promoter. They will want to know you can provide the kind of DJ set they like, which is where the demo may come into play. Other times, being able to describe the style with detail and key artists is enough. Recommendations from a DJ they already book can go a long way but can be tough to secure if you do not have a friendship. A lot of what to say to promoters and when is a practiced balance between 'not out there enough' and 'too pushy.'

Your first gig

Once you've got the gig, you're going to want to show up right. Bring anything you need, backup cables, etc. to make sure you can get into the booth and get music coming out of the speakers with minimal help. You may want to practice connecting your items to a club mixer so you can spot-correct any technicalities. This gets easier with time.

Show up as early as the promoter will be able to let you inside, usually before the public. Make sure all of your sound is working, and there's time to correct any last-minute issues.

Don't drink too much, nobody likes a sloppy DJ, but they don't appreciate you treating your job like 'just another party.' Some beers, or my personal choice, Red Bull on ice, so it appears to be an alcoholic beverage, are fine for socializing, thanking who came out, and being part of the room. But you want to look professional, fully functioning, and ready to work. Even after your set, as this can help with future bookings.

Be sure to thank the promoter for bringing you out, introduce him politely to any friends who may have come out for you (to show that value was brought), and try to wait as long as possible to be paid. Some promoters wait until the end of the night, others when the bar has made enough money. Be sure to bring it up, but do not annoy anyone. Promoters often have many problems or circumstances occurring during a night, and you want to make their job as easy as possible so they will pick you again!

DJ etiquette in the club

In addition to the previous notes, there are some light guidelines for how to behave when you're not performing that night. A lot of DJs enjoy being taken seriously and may overstep by judging the current DJ's transitions; you don't want to be 'that guy.' If it's positive, by all means. The main idea is you don't want to push onto any potential friend or business connection your DJing prowess. Subtle is critical, but consistent help. You will again, find this balance with time.

Try not to overdrink, or always be the one plugging yourself. Sometimes just dancing, being with friends, and being visible is enough to be approached by a promoter. Come early if you want to introduce yourself, that's when it's least busy. Most issues at this stage were solved before opening the club, and there is some 'down time' before playing host for the full room. Closing up can be frantic unless you're offering to help out, which might be a good foot in the door. Some set-ups can use some extra hands on a breakdown, larger venues will have a crew for this, but a local lounge night may get to know you from being a friendly face and helping hand.

EARLY MISTAKES TO AVOID

While mixing, it can take time to learn your way out of some bad habits. Here are some common ones that hopefully, accelerate your timeline.

Using too many effects

Effects are most exciting when saved for critical moments, and spaced out well between each other. Using the flanger too often is an early DJ move that tends to come up often, and dilutes the impact such a thing can have on the night. It might sound good early on, but you will have bored the audience of this when you're further into your set time. This is usually caused by being looked at and needing to 'do something,' as waiting between transitions can feel awkward. Crowd interaction helps here and is why so many famous DJs have a physical performance as well.

Only popular songs

Other DJs will consider it 'easy' for you to pick the hottest tracks and mix them, being able to build a set in various styles, and select songs the room may not know (called 'educating the crowd'), is the mark of a good DJ. You want to be the source of the music as much as possible, and putting in the time to find less conventional or known songs will help diversify you.

'Redline' the mixer

Mixers have a color-coded system that lets you know when the sound is distorting. 'Red' is characterized as distortion; you may sometimes have to push a sound system past the conventional limit, depending on the room, but in general, there's a saying: 'Red is dead.' It sounds bad and can clear a dance floor. Using too much volume gain or adding extra EQ to the songs can overdo this. Remember producers are talented people, and light adjustments are usually best.

Incorrect perception of the night

Sometimes lines can get crossed, or you may not have understood as thoroughly as you thought you had. There are nights where 'mellow' music means one thing, and this promoter wanted a very different version. Having backup music, or a few ideas of what is required can help here. If you walk in thinking Techno night was all about the hard-hitting sounds of drumcode records, you may need to adjust when you walk in and hear much slower nuanced sounds. You don't want to play the wrong set, but this goes beyond that. You may see individual styles of crowd interaction will not work here or may be preferred. The environment is not as clear until you are physically in it, or have been to many of this promoter's events. For this reason, research in both music and events is your saving grace.

IN CLOSING

Like producing, DJing is an ongoing, evolving skill set. The way you prefer to mix right now may not be the way you prefer to mix five years from now. Accepting this fluid structure and improving on it over time lends itself to the best results. Being able to understand where you are as an artist, or in a technical sense, can help you decide what will give the best performance. None of these methods or versions of performing is off limits to you, even if you start somewhere else first. It's more important just to start.

As I said earlier, you're welcome to incorporate not just songs, but any *sounds*, into your performance. There may be alternate ways of performing your songs and the songs of others in the future. Always be on the lookout for new ways to expand this living, breathing world of energy you create with your audience.

Appendix A

Binary and hex

All hardware circuits, even computers, are based on a series of interconnected switches that can be in one of two states: on or off. By continually switching these on and off in different configurations, it's possible for the hardware to count and perform complex calculations. This switching is accomplished by sending a series of bytes down an electrical cable, which means that they can only exist in one of two states too. If it is equal to 1 then the switch is turned on while if it's a 0 then the switch is turned off. Because of this, all computers must count using base 2, or 'binary' rather than our usual method of counting which is base 10.

COUNTING IN BINARY

Our normal method of counting was developed because we have ten fingers and using this system any one digit in a number has a value that is based on the position of the digit in the number. Thus, when we are working to the power of 10, every time a digit moves one to the left its result is to the power of 10, then by 100, then by 1000 and so forth. Consequently, the number 17,593 could be seen as follows:

10000000	1000000	100000	10000	1000	100	10	1
0	0	0	1	7	5	9	3

Or (1 x 10000) + (7 x 1000) + (5 x 100) + (9 x 10) + (3 x 1) = 17,360

From this example, we can see that each position in a number essentially adds a 0 to the meaning of a digit. Or, the value of each position is 10 times the value of the previous position (moving from right to left). This method of counting is also implemented in binary, but rather than use a base 10 system, we use a base 2 system. So, rather than saying that 10 to the power of 0 = 1, 10 to the power of 1 = 10, 10 to the power of 2 = 100, 10 to the power of 3 = 1000, it's 2 to the power of 0, 2 to the power of 1, 2 to the power of 2, 2 to the power of 4 and so forth:

128	64	32	16	8	4	2	1
0	0	0	0	0	0	0	0

Looking at the above table we are assuming that there are eight hardware switches grouped together producing what is essentially an 8-bit byte. Also, as they are all switched off the sum produced will be 0. However, if we were to introduce some positive bits we could sum them together to produce a decimal number:

128	64	32	16	8	4	2	1
0	1	1	0	1	1	0	1

(0 x 128) + (1 x 64) + (1 x 32) + (0 x 16) + (1 x 8) + (1 x 4) + (0 x 2) + (1 x 1) = 109

Thus, the decimal equivalent of the binary 01101101 would be 109. Additionally, we could also determine that the maximum number that could be calculated via binary would be 255 (11111111).

As MIDI uses this same 8-bit system when communicating with any devices it would seem sensible to assume that it should be able to offer this same maximum parameter (of 255) but this isn't the case. Similar to CC messages, two forms of information need to be transmitted: a status bit and a data byte (which is composed of seven bits). The status bit informs the synth of an incoming message that is arriving while the following data byte informs it by how much the parameter should be adjusted. Because of this initial status bit, only seven other bits are left to provide the information resulting in a maximum decimal value of 127, hence the maximum number of any CC message can only be 127. To transmit numbers larger than this, an 8-bit byte has to be split into two halves and then converted into another numeration format: hexadecimal.

When we split a byte into two halves, both halves are commonly referred to as 'nibbles.' If the previous example were to be broken down into two nibbles, it would become 0110 and 1101. These could then be individually summed together as 0110 = 96 and 1101 = 13 (96 + 13 = 109) to produce the result again. However, by splitting a byte into two and then converting it into a hexadecimal value, it's possible to produce much higher values. The reason behind this is that hexadecimal works to base 16, meaning that it is possible to access up to 16,383 parameters in a synth, much more than the standard 127 offered through control change messages.

COUNTING IN HEXADECIMAL

Hex uses a base-16 numbering system but as there are not enough symbols to represent 16 different digits, as soon as the number 10 is reached it has to be converted into letters. Thus, to represent number 10–15, the letters A to F are used.

Recall how we count in decimal. Counting upwards, we count from 1 through to 9 and then we place a 1 to the left of this and go back to 0, making the number

10 (which actually means 1 to the power of 10 plus 0 to the power of 1). With hexadecimal, we count beyond 9 but because our entire numeric system is only based on 9 digits, with hexadecimal, letters are used instead as below so that we can count up to 15.

F	E	D	C	B	A	9	8	7	6	5	4	3	2	1	0
15	14	13	12	11	10	9	8	7	6	5	4	3	2	1	0

When we count the zero, just as we do with the decimal counting system, we have used a total of 16 numbers. You can see from the decimal to hexadecimal conversion table that whenever we count up to F, increase the number to the left by 1. This is also the same as the more common decimal counting system, for example, if you count from 11 through to 19, you there are no more digits to use to you change the 9 to 0 and increase the number to the left by 1, giving us 20.

You can see that hexadecimal counting is exactly the same as decimal except in hexadecimal we count up to F before increasing the number to the left by a factor of 1.

It follows that in decimal, the number 24 actually means '2 x 10 to the power of 1 + 4 x 10 to the power of 0 = 24,' which is what you or I already understand it to be, because we use this every day like counting on our fingers.

In hexadecimal, the number 24 actually means '2 x 16 to the power of 1 + 4 x 16 to the power of 0 = 36.' Logically it follows then that the number CE in hexadecimal means '12 x 16 to the power of 1 + 14 x 16 to the power of 0 = 206' in decimal.

Using this principle, it's easy to convert hexadecimal into decimal.

Using this method, it's possible to count until we reach FF, which essentially means 15 x 16 + 15 units or 255 in decimal. Creating a new position gives 100 or 16 to the power of 2 which is 256 in decimal. Continuing to count using this base system, it would be possible to produce two nibbles, both with a value of 7F resulting in a total of 16,383. (That is, 127 x 128 + 127 = 16,383!)

For instance, suppose we wished to convert the decimal number 12,720 into hexadecimal:

- The largest power of 16 that fits into 12,720 is 16 ^ 3 = 4,096
  It fits three times and gives a remainder of 432. This number is derived from the fact that:

  (3 x 4,096) = 12,288
  (12,720–12,288) = 432

  The first hex number is *3*

- The largest power of 16 that fits into 432 is 16 ^ 2 = 256
  It fits once and gives a remainder of 176. This number is derives from the fact that:

  (1 x 256) = 256
  (432–256) = 176

  The second hex number is *1*
- The largest power of 16 that fits into 176 is 16 ^ 1 = 16
  It fits 11 times with no remainder (this remainder equates to 1 to the power of0, which is 0 and will make 0 the last digit in the hex number). The third hex number is therefore *B*
- The hex equivalent of the decimal number 12,720 is, therefore, 31B0

> To clarify: to convert 31B0 hexadecimal into decimal the math is as follows:
> 0 x (16 to the power of 0) = 0
> B (which is equivalent to 11 in decimal) x (16 to the power of 1) = 176
> 1 x (16 to the power of 2) = 256
> 3 x (16 to the power of 3) = 12288
> 0 + 176 + 256 + 12,288 = 12,720 in decimal

As another example, suppose we wanted to convert the decimal number 14,683:

- The largest power of 16 that fits into 14,683 is 16 ^ 3 = 4,096
  It fits three times and gives a remainder of 2,395
- This remainder is derived from the fact that:

  (3 x 4,096) = 12,288
  (14,683–12,288) = 2,395

  The first hex number is *3*
- The largest power of 16 that fits into 2,395 is 16 ^ 2 = 256
  It fits nine times and gives a remainder of 91
- This remainder is derived from the fact that:

  (9 x 256) = 2,304
  (2,395–2,304) = 91

  The second hex number is *9*
- The largest power of 16 that fits into 91 is 16 ^ 1 = 1
  It fits five times and gives a remainder of 11
- This remainder is derived from the fact that:

  (5 x 16) = 80
  (91–80) = 11

  The third hex number is *5*

- The remainder of 11 has no power of 16 that will divide into it as a whole number so the last digit of the hex will be the hex equivalent of 11 which is B. This gives the final hex number is therefore 395B.

Again we can check this by converting 295B hexadecimal into decimal. The math is as follows:
B x (16 to the power of 0) = 11
5 x (16 to the power of 1) = 80
9 x (16 to the power of 2) = 2,304
3 x (16 to the power of 3) = 12,288
11 + 80 + 2,304 + 12,288 = 14,683 in decimal

Appendix B

Decimal to hexadecimal conversion table

Dec	Hex	Dec	Hex	Dec	Hex	Dec	Hex	Dec	Hex	Dec	Hex
0	0	44	2C	88	58	132	84	176	B0	220	DC
1	1	45	2D	89	59	133	85	177	B1	221	DD
2	2	46	2E	90	5A	134	86	178	B2	222	DE
3	3	47	2F	91	5B	135	87	179	B3	223	DF
4	4	48	30	92	5C	136	88	180	B4	224	E0
5	5	49	31	93	5D	137	89	181	B5	225	E1
6	6	50	32	94	5E	138	8A	182	B6	226	E2
7	7	51	33	95	5F	139	8B	183	B7	227	E3
8	8	52	34	96	60	140	8C	184	B8	228	E4
9	9	53	35	97	61	141	8D	185	B9	229	E5
10	A	54	36	98	62	142	8E	186	BA	230	E6
11	B	55	37	99	63	143	8F	187	BB	231	E7
12	C	56	38	100	64	144	90	188	BC	232	E8
13	D	57	39	101	65	145	91	189	BD	233	E9
14	E	58	3A	102	66	146	92	190	BE	234	EA
15	F	59	3B	103	67	147	93	191	BF	235	EB
16	10	60	3C	104	68	148	94	192	C0	236	EC
17	11	61	3D	105	69	149	95	193	C1	237	ED
18	12	62	3E	106	6A	150	96	194	C2	238	EE

Dec	Hex	Dec	Hex	Dec	Hex	Dec	Hex	Dec	Hex	Dec	Hex
19	13	63	3F	107	6B	151	97	195	C3	239	EF
20	14	64	40	108	6C	152	98	196	C4	240	F0
21	15	65	41	109	6D	153	99	197	C5	241	F1
22	16	66	42	110	6E	154	9A	198	C6	242	F2
23	17	67	43	111	6F	155	9B	199	C7	243	F3
24	18	68	44	112	70	156	9C	200	C8	244	F4
25	19	69	45	113	71	157	9D	201	C9	245	F5
26	1A	70	46	114	72	158	9E	202	CA	246	F6
27	1B	71	47	115	73	159	9F	203	CB	247	F7
28	1C	72	48	116	74	160	A0	204	CC	248	F8
29	1D	73	49	117	75	161	A1	205	CD	249	F9
30	1E	74	4A	118	76	162	A2	206	CE	250	FA
31	1F	75	4B	119	77	163	A3	207	CF	251	FB
32	20	76	4C	120	78	164	A4	208	D0	252	FC
33	21	77	4D	121	79	165	A5	209	D1	253	FD
34	22	78	4E	122	7A	166	A6	210	D2	254	FE
35	23	79	4F	123	7B	167	A7	211	D3	255	FF
36	24	80	50	124	7C	168	A8	212	D4		
37	25	81	51	125	7D	169	A9	213	D5		
38	26	82	52	126	7E	170	AA	214	D6		
39	27	83	53	127	7F	171	AB	215	D7		
40	28	84	54	128	80	172	AC	216	D8		
41	29	85	55	129	81	173	AD	217	D9		
42	2A	86	56	130	82	174	AE	218	DA		
43	2B	87	57	131	83	175	AF	219	DB		

Appendix C

General MIDI instrument patch maps

Program no.	Instrument set (piano)	Program no.	Instrument (chromatic perc)
1	Acoustic grand	9	Celesta
2	Bright acoustic	10	Glockenspiel
3	Electric grand	11	Music box
4	Honky tonk	12	Vibraphone
5	Electric piano 1	13	Marimba
6	Electric piano 2	14	Xylophone
7	Harpsichord	15	Tubular bells
8	Clav	16	Dulcimer

Program no.	Instrument (organ)	Program no.	Instrument (guitar)
17	Drawbar organ	25	Nylon acoustic guitar
18	Percussive organ	26	Steel acoustic guitar
19	Rock organ	27	Jazz electric guitar
20	Church organ	28	Clean electric guitar
21	Reed organ	29	Muted electric guitar
22	Accordion	30	Overdrive guitar
23	Harmonica	31	Distortion guitar
24	Tango accordion	32	Guitar harmonics

Program no.	Instrument (bass)	Program no.	Instrument (strings)
33	Acoustic bass	41	Violin
34	Finger bass	42	Viola
35	Pick bass	43	Cello
36	Fretless bass	44	Contrabass
37	Slap bass 1	45	Tremolo strings
38	Slap bass 2	46	Pizzicato
39	Synth bass 1	47	Orchestral
40	Synth bass 2	48	Timpani

Program no.	Instrument (ensemble)	Program no.	Instrument (brass)
49	String ensemble 1	57	Trumpet
50	String ensemble 2	58	Trombone
51	Synth strings 1	59	Tuba
52	Synth strings 2	60	Muted trumpet
53	Choir aahs	61	French horn
54	Choir oohs	62	Brass section
55	Synth voice	63	Synth brass 2
56	Orchestral hit	64	Synth brass 2

Program no.	Instrument (reed)	Program no.	Instrument (pipe)
65	Soprano sax	73	Piccolo
66	Alto sax	74	Flute
67	Tenor sax	75	Recorder
68	Baritone sax	76	Pan flute
69	Oboe	77	Blown bottle
70	English horn	78	Skakuhachi
71	Bassoon	79	Whistle
72	Clarinet	80	Ocarina

Program no.	Instrument (synth lead)	Program no.	Instrument (synth pad)
81	Square lead	89	New age pad
82	Sawtooth lead	90	Warm pad
83	Calliope lead	91	Polysynth pad
84	Chiff lead	92	Choir pad
85	Charang lead	93	Bowed pad
86	Voice lead	94	Metallic pad
87	Fifths lead	95	Halo pad
88	Bass and lead	96	Sweep pad

Program no.	Instrument (synth effects)	Program no.	Instrument (ethnic)
97	Rain FX	105	Sitar
98	Soundtrack FX	106	Banjo
99	Crystal FX	107	Shamisen
100	Atmosphere FX	108	Koto
101	Brightness FX	109	Kalimba
102	Goblins FX	110	Bagpipe
103	Echoes FX	111	Fiddle
104	Sci-fi FX	112	Shanai

Program no.	Instrument (percussive)	Program no.	Instrument (sound FX)
113	Tinkle bell	121	Guitar fret noise
114	Agogo	122	Breath noise
115	Steel drums	123	Seashore
116	Woodblock	124	Bird tweet
117	Taiko drums	125	Telephone ring
118	Melodic toms	126	Helicopter
119	Synth drum	127	Applause
120	Reverse cymbal	128	Gunshot

GENERAL MIDI PERCUSSION SET

MIDI key	Drum sound	MIDI key	Drum sound
35	Acoustic bass drum	59	Ride cymbal
36	Bass drum	60	Hi bongo
37	Side stick	61	Low bongo
38	Acoustic snare	62	Mute hi bongo
39	Hand clap	63	Open hi bongo
40	Electric snare	64	Low conga
41	Low floor tom	65	High timbale
42	Closed hi-hats	66	Low timbale
43	High floor tom	67	High agogo
44	Pedal hi-hat	68	Low agogo
45	Low tom	69	Cabasa
46	Open hi-hat	70	Maracas
47	Low mid tom	71	Short whistle
48	High mid tom	72	Long whistle
49	Crash cymbal	73	Short guiro
50	High tom	74	Long guiro
51	Ride cymbal	75	Claves
52	Chinese cymbal	76	Hi wood block
53	Ride bell	77	Low wood block
54	Tambourine	78	Mute cuica
55	Splash cymbal	79	Open cuica
56	Cowbell	80	Mute triangle
57	Crash cymbal	81	Open triangle
58	Vibraslap		

Appendix D

General MIDI CC list

CC	Function	CC	Function	CC	Function
0	Bank select	19	General control 4	49	General purpose controller 2
1	Mod wheel	20–31	Undefined	50	General purpose controller 3
2	Breath controller	32	Bank select	51	General purpose controller 4
3	Undefined	33	Mod wheel	52–63	Undefined
4	Foot controller	34	Breath control	64	Damper pedal (on/off)
5	Portamento time	35	Undefined	65	Portamento (on/off)
6	Data entry	36	Foot control	66	Sustenuto (on/off)
7	Channel volume	37	Portamento time	67	Soft pedal (on/off)
8	Balance	38	Data entry	68	Legato footswitch
9	Undefined	39	Channel volume	69	Hold 2
10	Pan	40	Balance	70	Sound controller 1 (sound variation)
11	Expression	41	Undefined	71	Sound controller 2 (timbre)
12	Effect control 1	42	Pan	72	Sound controller 3 (release time)
13	Effect control 2	43	Expression controller	73	Sound controller 4 (attack time)
14–15	Undefined	44	Effect control 1	74	Sound controller 5 (brightness)
16	General control 1	45	Effect control 2	75	Sound controller 6
17	General control 2	46–47	Undefined	76	Sound controller 7
18	General control 3	48	General purpose controller 1	77	Sound controller 8

CC	Function	CC	Function	CC	Function
78	Sound controller 9	93	Effects 3 (chorus) depth	120	All sound off
79	Sound controller 10	94	Effects 4 (detune) depth	121	Reset all controllers
80	General purpose controller 5	95	Effects 5 (phaser) depth	122	Local control on/off
81	General purpose controller 5	96	Data entry +1	123	All notes off
82	General purpose controller 5	97	Data entry –1	124	Omni mode off (+ all notes off)
83	General purpose controller 5	98	Non-registered parameter number LSB[1]	125	Omni mode on (+ all notes off)
84	Portamento control	99	Non-registered parameter number MSB	126	Poly mode on/off (+ all notes off)
85–90	Undefined	100	Registered parameter number LSB	127	Poly mode on
91	Effects 1 (reverb) depth	101	Registered parameter number MSB[2]		
92	Effects 2 (tremolo) depth	102–119	Undefined		

[1] *Least significant bit*
[2] *Most significant bit*

Appendix E

Sequencer note divisions

The following charts display the number of clock pulses for each note value for the four most popular PPQN resolutions – 96, 192, 240, and 384.

96 PPQN

Note value	PPQN	Note value	PPQN	Note value	PPQN
Whole	384	Dotted whole	576	Triplet whole	256
Half	192	Dotted half	288	Triplet half	128
Quarter	96	Dotted quarter	144	Triplet quarter	64
Eighth	48	Dotted eighth	72	Triplet eighth	32
1/16th	24	Dotted 1/16th	36	Triplet 1/16th	16
32nd	12	Dotted 32nd	18	Triplet 32nd	8
64th	6	Dotted 64th	9	Triplet 64th	4
128th	3	Dotted 128th	N/A	Triplet 128th	2

192 PPQN

Note value	PPQN	Note value	PPQN	Note value	PPQN
Whole	768	Dotted whole	1152	Triplet whole	512
Half	384	Dotted half	576	Triplet half	256
Quarter	192	Dotted quarter	288	Triplet quarter	128
Eighth	96	Dotted eighth	144	Triplet eighth	64
1/16th	48	Dotted 1/16th	73	Triplet 1/16th	32
32nd	24	Dotted 32nd	36	Triplet 32nd	16
64th	12	Dotted 64th	18	Triplet 64th	8
128th	6	Dotted 128th	9	Triplet 128th	4

240 PPQN

Note value	PPQN	Note value	PPQN	Note value	PPQN
Whole	960	Dotted whole	1440	Triplet whole	640
Half	480	Dotted half	720	Triplet half	320
Quarter	240	Dotted quarter	360	Triplet quarter	160
Eighth	120	Dotted eighth	180	Triplet eighth	80
1/16th	60	Dotted 1/16th	90	Triplet 1/16th	40
32nd	30	Dotted 32nd	45	Triplet 32nd	20
64th	15	Dotted 64th	N/A	Triplet 64th	10
128th	N/A	Dotted 128th	N/A	Triplet 128th	5

384 PPQN

Note value	PPQN	Note value	PPQN	Note value	PPQN
Whole	1536	Dotted whole	2304	Triplet whole	1024
Half	768	Dotted half	1152	Triplet half	512
Quarter	384	Dotted quarter	576	Triplet quarter	256
Eighth	192	Dotted eighth	288	Triplet eighth	128
1/16th	96	Dotted 1/16th	144	Triplet 1/16th	64
32nd	48	Dotted 32nd	72	Triplet 32nd	32
64th	24	Dotted 64th	36	Triplet 64th	16
128th	12	Dotted 128th	18	Triplet 128th	8

Appendix F

Tempo delay time chart

If song tempo is 128 BPM then set delay time to 469 Ms for quarter-note delay, 234 Ms for eighth-note delay, 156 Ms for eighth-triplet delay, or 117 Ms for 1/16th note delay.

Tempo	1/4	1/8	1/8T	1/16	Tempo	1/4	1/8	1/8T	1/16	Tempo	1/4	1/8	1/8T	1/16
80	750	375	250	188	118	508	254	169	127	156	385	192	128	96
81	741	370	247	185	119	504	252	168	126	157	382	191	127	96
82	732	366	244	183	120	500	250	167	125	158	380	190	127	95
83	723	361	241	181	121	496	248	165	124	159	377	189	126	94
84	714	357	238	179	122	492	246	164	123	160	375	188	125	94
85	706	353	235	176	123	488	244	163	122	161	373	186	124	93
86	698	349	233	174	124	484	242	161	121	162	370	185	123	92
87	690	345	230	172	125	480	240	160	120	163	368	184	123	92
88	682	341	227	170	126	476	238	159	119	164	366	183	122	91
89	674	337	225	169	127	472	236	157	118	165	364	182	121	91
90	667	333	222	167	128	469	234	156	117	166	361	181	120	90
91	659	330	220	165	129	465	233	155	116	167	359	180	120	90
92	652	326	217	163	130	462	231	154	115	168	357	179	119	89
93	645	323	215	161	131	458	229	153	115	169	355	178	118	88
94	638	319	213	160	132	455	227	152	114	170	353	176	118	88
95	632	316	211	158	133	451	226	150	113	171	351	175	117	88
96	625	313	208	156	134	448	224	149	112	172	349	174	116	87
97	619	309	206	155	135	444	222	148	111	173	347	173	116	87

Tempo	1/4	1/8	1/8T	1/16	Tempo	1/4	1/8	1/8T	1/16	Tempo	1/4	1/8	1/8T	1/16
98	612	306	204	153	136	441	221	147	110	174	345	172	115	86
99	606	303	204	153	137	438	219	146	109	175	343	171	114	86
100	600	300	200	150	138	435	217	145	109	176	341	170	114	85
101	594	297	198	149	139	432	216	144	108	177	339	169	113	85
102	588	294	196	147	140	429	214	143	207	178	337	169	112	84
103	583	291	194	146	141	426	213	142	106	179	335	168	112	84
104	577	288	192	144	142	423	211	141	106	180	333	168	111	83
105	571	286	190	143	143	420	210	140	105	181	331	167	110	83
106	566	283	189	142	144	417	208	139	104	182	299	166	110	82
107	561	280	187	140	145	414	207	138	103	183	297	165	109	82
108	556	278	185	139	146	411	205	137	103	184	295	164	108	81
109	550	275	183	138	147	408	302	136	102	185	293	164	108	81
110	545	273	182	136	148	405	203	135	101	186	291	163	107	80
111	541	270	180	135	149	403	201	134	101	187	289	162	106	80
112	536	268	179	134	150	400	200	133	100	188	287	161	106	79
113	531	265	177	133	151	397	199	132	99	190	285	161	105	79
114	526	263	175	132	152	395	197	132	99	200	283	160	104	78
115	522	261	174	130	153	392	196	131	98	201	281	159	104	78
116	517	259	172	129	154	390	195	130	97	202	279	158	103	77
117	513	256	171	128	155	387	194	129	97	203	277	157	102	77

1/8 = eighth-note delay, 1/8T = eighth-note triplet delay, 1/16 = sixteenth-note delay

Appendix G

Musical note to MIDI and frequencies

Note	MIDI no.	Frequency	MIDI no.	Frequency
C	0	8.1757989156	12	16.3515978313
D♭	1	8.6619572180	13	17.3239144361
D	2	9.1770239974	14	18.3540479948
E♭	3	9.7227182413	15	19.4454364826
E	4	10.3008611535	16	20.6017223071
F	5	10.9133822323	17	21.8267644646
G♭	6	11.5623257097	18	23.1246514195
G	7	12.2498573744	19	24.4997147489
A♭	8	12.9782717994	20	25.9565435987
A	9	13.7500000000	21	27.5000000000
B♭	10	14.5676175474	22	29.1352350949
B	11	15.4338531643	23	30.8677063285

Note	MIDI no.	Frequency	MIDI no.	Frequency
C	24	32.7031956626	36	65.4063913251
D♭	25	34.6478288721	37	69.2956577442
D	26	36.7080959897	38	73.4161919794
E♭	27	38.8908729653	39	77.7817459305
E	28	41.2034446141	40	82.4068892282
F	29	43.6535289291	41	87.3070578583
G♭	30	46.2493028390	42	92.4986056779
G	31	48.9994294977	43	97.9988589954
A♭	32	51.9130871975	44	103.8261743950
A	33	55.0000000000	45	110.0000000000
B♭	34	58.2704701898	46	116.5409403795
B	35	61.7354126570	47	123.4708253140

Note	MIDI no.	Frequency	MIDI no.	Frequency
C	48	130.8127826503	60	261.6255653006
D♭	49	138.5913154884	61	277.1826309769
D	50	146.8323839587	62	293.6647679174
E♭	51	155.5634918610	63	311.1269837221
E	52	164.8137784564	64	329.6275569129
F	53	174.6141157165	65	349.2282314330
G♭	54	184.9972113558	66	369.9944227116
G	55	195.9977179909	67	391.9954359817
A♭	56	207.6523487900	68	415.3046975799
A	57	220.0000000000	69	440.0000000000
B♭	58	233.0818807590	70	466.1637615181
B	59	246.9416506281	71	493.8833012561

Note	MIDI no.	Frequency	MIDI no.	Frequency
C	72	523.2511306012	84	1046.5022612024
D♭	73	554.3652619537	85	1108.7305239075
D	74	587.3295358348	86	1174.6590716696
E♭	75	622.2539674442	87	1244.5079348883
E	76	659.2551138257	88	1318.5102276515
F	77	698.4564628660	89	1396.9129257320
G♭	78	739.9888454233	90	1479.9776908465
G	79	783.9908719635	91	1567.9817439270
A♭	80	830.6093951599	92	1661.2187903198
A	81	880.0000000000	93	1760.0000000000
B♭	82	932.3275230362	94	1864.6550460724
B	83	987.7666025122	95	1975.5332050245

Note	MIDI no.	Frequency	MIDI no.	Frequency
C	96	2093.0045224048	108	4186.0090448096
D♭	97	2217.4610478150	109	4434.9220956300
D	98	2349.3181433393	110	4698.6362866785
E♭	99	2489.0158697766	111	4978.0317395533
E	100	2637.0204553030	112	5274.0409106059
F	101	2793.8258514640	113	5587.6517029281
G♭	102	2959.9553816931	114	5919.9107633862
G	103	3135.9634878540	115	6271.9269757080
A♭	104	3322.4375806396	116	6644.8751612791
A	105	3520.0000000000	117	7040.0000000000
B♭	106	3729.3100921447	118	7458.6201842894
B	107	3951.0664100490	119	7902.1328200980

Note	MIDI no.	Frequency
C	120	8372.0180896192
D♭	121	8869.8441912599
D	122	9397.2725733570
E♭	123	9956.0634791066
E	124	10548.0818212118
F	125	11175.3034058561
G♭	126	11839.8215267723
G	127	12543.8539514160

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