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RECORDING on a Budget

How to Make Great Audio Recordings Without Breaking the Bank

BRENT EDSTROM

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Printed in the United States of America
on acid-free paper

This book is dedicated to my son, Evan.

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About the Companion Web Site

www.oup.com/us/recordingonabudget

Oxford has created a password-protected Web site to accompany *Recording on a Budget*. The site features over 240 audio excerpts that demonstrate concepts from the book including microphone placement, audio processing, active listening, and mixing and mastering. Many categories of instruments and ensembles are represented by the recordings.

The online excerpts provide an essential dimension to the book and readers are encouraged to listen to the recordings in conjunction with chapter readings in order to assimilate the concepts from both a conceptual and an aural perspective.

Throughout the text, online examples are indicated with Oxford's symbol.

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Introduction

In 2008, I gave a presentation on the topic of audio recording at the International Association for Jazz Education convention in Toronto. My goal for the session was to talk, musician-to-musician, about the process of recording and to discuss the types of equipment that are required to record music. The topic seemed to resonate with many of the attendees, and over the next few weeks, I received a number of e-mails from fellow musicians and educators who were in attendance at the session. It was clear that many musicians share an interest in recording but lack the technical background to know where to begin. It was also evident that questions concerning the cost and selection of recording equipment prevent many musicians from exploring this rewarding aspect of the music-making process.

This book is not a technical book geared to audio professionals—it is written by a musician and is geared to other musicians, songwriters, composers, band and choir directors, hobbyists, educators, and anyone else who has a desire to learn about recording from a practical and budget-conscious perspective. While technical knowledge is always an advantage, my intent is to focus on the concepts that will help you record better music. The recording arts *is* a complex subject that requires a unique blend of technical knowledge and artistic acumen, but recording need not be a black art relegated to individuals with a degree in audio engineering. With a basic technical knowledge, an understanding of the recording process, and a few tools, it is possible to make good recordings. That is not to say that you will never need to hire a recording or mastering engineer—as with all facets of music there are times where it makes sense to utilize the skills of someone with appropriate experience. However, in many cases, you will be able to do the work yourself and you will find that the development of skills in the recording arts is an immensely satisfying experience that can be enriching on both a professional and an artistic level.

From a professional standpoint, there are innumerable benefits to the development of skills in the recording arts. The industry is competitive and recordings are the primary calling card that enables musicians to find work, so musicians who can produce a quality recording will be at a distinct advantage. In many cases, budget constraints, timelines, or location may limit access to a professional studio. For example, although I live in a fairly large city, there is not a *single* studio that owns a grand piano. This has been true in other locations as well, so practically speaking, it was essential for me to learn about recording in order to maintain my professional career, as it would be impractical to drive hundreds of miles just to record a demo or mix a commercial project.

Perhaps the biggest benefit in developing a knowledge of recording comes in the form of artistic freedom. Home recording provides the opportunity to work whenever the muse strikes, and knowledge of recording concepts is useful when working with audio professionals. For example, many years ago I recorded a trio album at a "name" studio in Seattle. Unfortunately, the young engineer did not have a background in trio music, and I was disappointed with the results of the session. In hindsight, knowledge of recording concepts would have helped me to communicate my artistic goals in a more proactive way as the session unfolded.

Knowledge of concepts such as microphone placement and signal processing is also useful in the realm of live performance. I do occasional work as a pianist and musical director for Motown singer Freda Payne. As a musical director, it is my job to make sure Ms. Payne sounds great when she performs at a given venue, so it is natural for me to assess the front-of-house mix during our soundchecks. In most cases, the live engineer is a professional and I need not worry about the sound, but there have been occasions when it has been necessary to speak with the engineer or ask the bassist to make an adjustment to an amplifier. My interest in the *sound* of a recording or live mix has made it possible to be more attentive to these details.

Voltaire stated that "the most useful books are those in which the reader does half the work himself. He develops the author's thoughts if given the germ of an idea."[1] Voltaire's statement is perfectly suited to the tone of this book. While technical knowledge is an advantage in the realm of recording, I would stress that, as with other aspects of musicianship, practice and experimentation will enable you to develop skills in this exciting discipline. My intent is to focus on the concepts that will help you to make good recordings and to provide practical advice regarding the tools that are necessary to do the job. In doing half the work you will be rewarded with a new set of skills and a different perspective on the music-making process.

Recording on a Budget

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1

Setting Up a Project Studio

In this chapter we will consider the primary components of a project studio. Before we begin, I want to emphasize that there is no one-size-fits-all solution: the selection of equipment involves an array of variables that include cost and performance, as well as more subjective choices such as the selection of microphones and speakers. With that said, by the end of this chapter you will have a good understanding of the various components of a project studio. You will understand how the components work together in a cohesive unit and will be able to narrow the options for your unique situation.

How Good Is Good Enough?

One of the biggest challenges you will face when purchasing equipment for a project studio is balancing cost with performance. In 2008, I had the pleasure of hearing recording engineers Al Schmitt and Phil Ramone speak in Toronto. One of the attendees asked Mr. Schmitt to describe his approach to recording acoustic bass. The equipment he described would cost several thousand dollars—well out of the reach of most home recordists. Although the example might seem extreme, it does illustrate the fact that high-end facilities use high-end gear, and it is unreasonable to expect a $200 signal chain to compete with a $5000 signal chain. That type of assumption is missing the point of this book: audio recording can be an exceedingly rewarding experience at any level, and it is possible to make good recordings with a modest investment in equipment. It is also important to remember that equipment is only one part of the recording equation. The performance of the musicians and the sound of their instruments and voices are key components, and the best gear in the world will not make a mediocre performance sound better. On the flip side, it is important to understand that your ears and engineering skill will have a big impact on the final product. I have no doubt that Mr. Schmitt or Mr. Ramone would produce better recordings in my modest project studio, given their obvious mastery and many years of experience in the recording arts.

When purchasing equipment, consider a price point that is reasonable for the work you intend to do and purchase the best tools at that price point. I see an analog with the purchase of woodworking tools: as an avocational woodworker, I find it does not make sense for me to purchase the high-end tools that would be found in a professional shop. However, it does make sense for me to purchase quality tools that will keep an edge and allow me to produce professional-quality work. Similarly, as a professional musician I need to be able to produce professional-sounding recordings for commercial projects, demos, and CDs, but my project studio will never be used to produce an album for a star like Norah Jones or Sting. Practically speaking, this means that my microphone cabinet contains solid microphones in the $100 to $500 price range but no $3000 microphones. On the other hand, I would never consider wasting money on a microphone that doesn't provide excellent (if not exceptional) sound quality.

As you work through the chapter, consider the type of recording you intend to do. If your goal is to record a choir, you will likely want to focus your resources on good microphones and preamps. In contrast, a mixing console with plenty of channels may be necessary if your goal is to record a band.

Purchasing Equipment

This is a wonderful time to be involved with digital audio recording. There is a wide selection of tools available at various price points that make it possible to get started without breaking the bank. Trade magazines such as *Electronic Musician, Mix, Sound on Sound,* and others provide helpful reviews of products and full-color advertisements of new products. A downside is that it can be easy to lose sight of the objective—creating great recordings on a budget. While high sample rates and boutique preamps are attractive options, I sometimes remind myself that many of the recordings that have withstood the test of time were produced with relatively primitive gear. While I certainly recommend tools that provide a suitable level of quality for a given task, you will find that equipment represents just one facet of recording. The quality of the performance is a crucial aspect, as are the ears and vision of the engineer.

All of us work within some financial constraints, and it can be a challenge to purchase gear that provides an appropriate level of quality without a large outlay of cash. With that said, with some forethought it is possible to piece together a quality project studio that will provide years of enjoyment. Although equipment is obviously a necessary part of audio recording, this is not a "gear" book. This tome is about making the best possible recordings with modest equipment. You will find that, as you develop your skills in the recording arts, you will be rewarded with a new appreciation for the sonic nuances in professional recordings.

Think Long Term

A first step is to consider the possible ways you intend to use your project studio. As much as possible, try to anticipate future uses, as well as immediate use. For example, a singer/songwriter will likely need the capability to record just a few tracks at a time. However, additional tracks may be necessary to effectively record multiple instruments if the same musician decides to record a live version of a song with other musicians.

Oftentimes, physical space will determine track count. For example, it was easy to calculate the track count for my project studio: I have room to simultaneously record piano (2 tracks), bass (1–2 tracks), and drums (4–5 tracks) and perhaps a guitar, vocal, or auxiliary percussion track. I simply don't have room to handle more than twelve tracks so a twenty-four track setup would be excessive. Almost all recording software provides the capability to playback twenty-four (or more) tracks so the number of simultaneous inputs available on an audio interface is usually a key consideration. (We will discuss these concepts in more detail in the chapters that follow.)

Another long-term consideration involves the way you work with other musicians. In some cases, it may make sense to rent a professional facility to do tracking and utilize a home studio for mixing. In this instance, it is helpful to consider software compatibility. Professional facilities use any number of tools, but applications such as Pro Tools, Logic, and Cubase are very common. It may make sense to start with a scaled-down version of software and upgrade as your needs and skills develop. Many of the common software applications have upgrade paths available that will enable you to move to a more advanced version of the product without paying full price.

The style of music you intend to record is also a consideration. Classical recordings are usually made with fewer microphones than rock or pop recordings and are typically recorded without overdubs. For this reason, classical recordists usually prioritize the purchase of just a few high-quality microphones and preamps in order to best capture a live performance. In contrast, most recordists of rock music will utilize sixteen or more tracks so an audio interface with numerous inputs would be appropriate.

Purchase Quality Tools

Throughout this book we will consider ways in which you can save money, but I always encourage friends and colleagues to purchase equipment that provides a suitable level of quality. A good microphone will be useful for decades, as will other components such as reference monitors and mixer. In fact, I own a mixer that is nearly twenty years old and it still gets occasional use when I need additional inputs.

In terms of quality, there are two rather subjective applications of the term. *Build quality* refers to how a product is constructed and how it will hold up over time. The term may also refer to defect tolerance in the manufacturing stage. A more subjective application of the term refers to *audio quality*. In this book you will learn many concepts that will help you evaluate the specifications of a given product, but ultimately, we are concerned with how a product sounds. There is often a correlation between cost and audio quality; after all, high-end studios would use less-expensive microphones and preamps if the tools provided the best level of performance. The key is to purchase a suitable level of quality. Is it possible to make a good recording with a $100 to $200 microphone? Absolutely! Will it sound as good as a $4000 microphone and preamp? Probably not, but that is missing the point—reasonably priced equipment is used every day to capture performances that are heard on CDs, commercials, and television. Even professional studios will likely have just a few channels of "the good stuff" and will rely on more modest equipment for basic tracking. As you will see in chapter 4, there are also times when less expensive equipment is the right choice. My best advice is to purchase the best and most suitable tools you can afford, with the thought that a quality tool will last a lifetime.

Use a Modular Approach

Although all-in-one solutions are a good choice for some people (and we will discuss these options in the pages that follow), I prefer to use a modular approach in my project studio. With a modular approach it is possible to upgrade single components without the need to purchase an entirely new system. With a modular system (computer, digital audio interface, and software), it is easy to add additional inputs, effects plug-ins, or other components as necessary. For example, it may make sense to upgrade your system with new software and plug-ins geared to mastering if you find yourself doing that type of work.

All-in-one solutions do make some sense in some situations. Such a system might be a good choice for a high school director who wants a reliable and portable system for live recording or a songwriter who doesn't want a complex modular system to interfere with the elusive muse.

Arm Yourself with Knowledge

Knowledge is the key to finding good value in equipment. Throughout the book I will discuss options for various types of recording equipment, and this knowledge will enable you to narrow your choices. Owing to the ever-changing nature of the audio industry, I will for the most part avoid making specific recommendations. Even if I knew the industry were to stop developing new gear, I would still avoid making specific recommendations since the selection of gear is such a personal process. However, once you know how the various components of a project studio fit together, you will have the essential knowledge that will enable you to refine your selections.

I am in the habit of visiting online forums to see how a particular product is holding up in the trenches before I purchase a new item. Be aware, though, that online user reviews can provide a skewed picture of a product. I have found instances in which reviews were posted by users who had not actually used the product. My guard is also up when I read blanket statements (either pro or con) that do not include detailed evidence relating to how the product does, or does not, stack up.

User manuals are another helpful source of information that can inform a purchase. Manufacturers typically provide product manuals as a free download, and this information can be useful in determining the suitability of a product for your intended application.

Shopping for Equipment

For a budget-conscious musician, used gear can provide attractive options. After all, if a studio was willing to spend top dollar for high-end equipment just a few years ago, it is likely that the same equipment could be very useful to a musician who is looking for a good ratio of value to quality. Two of the most popular options are eBay and Craigslist. In some cases, it is possible to trade an item that you are no longer using for a product that you need. Bear in mind that crooks frequent online services so be cautious when buying, selling, or bartering online.

Avoid Gear Lust

I mentioned previously that trade magazines are useful in expanding your knowledge of recording and recording products. Be aware that it is easy to fall into the trap of "gear lust." This is the feeling that a particular product will magically make your music sound better and therefore would be an essential part of your recording toolkit. While it is true that specific products such as a great microphone or preamp can make your music sound better, it is important to give yourself a reality check. For example, if your recording environment has severe acoustic problems, then it is likely an exercise in futility to purchase high-end microphones and preamps. In contrast, inferior gear can turn an excellent performance into a mediocre one if the gear is not up to the task or not accurately capturing the performance. The trick, and it's not easy, is to find a suitable level of quality when purchasing gear. If your goal is to enjoy making music and learning about recording, then likely you will not need the highest fidelity. On the other hand, if you intend to create professional projects, then likely you will need to look at "prosumer," or higher quality, in the gear that you purchase. In many cases, spending a bit more money upfront to purchase a professional tool will pay big dividends over the long term.

Where to Cut Corners

You can save a great deal of money by making your own studio furniture, cables, and accessories. While top-of-the-line microphone cables can cost more than $5 per foot, it is relatively easy to make quality cables for around fifty cents per foot. Racks, stands, and many other items can also be made at home for a reasonable cost.

Throughout the book I will discuss strategies and concepts that will help you evaluate and purchase tools that are just right for what you need to do. Avoid the temptation that you need to wait for the latest and greatest in order to begin your exploration of audio recording. After all, people have been making records for nearly 100 years, and though the technology has advanced to an astonishing level, the bottom line is that good performances are at the heart of good recordings. Of course, we want to capture performances with sonic clarity, but your ears and musical sense are just as important (if not more so) than esoteric gear. I once heard it phrased that a top engineer/producer could produce a hit on mediocre equipment, and I think that comment is not far from the mark. You will find that, as your audio recording skills grow and your ears become more accustomed to hearing the nuances of a recording, you will be able to get better results with modest gear.

Anatomy of a Project Studio

In the following pages you will learn about the primary components of a project studio. The discussion is geared to familiarizing you with common categories of equipment and how the equipment is connected to form a studio. In most cases, topics such as the selection and use of specific types of microphones will be developed in other chapters.

I want to emphasize that this chapter is *not* meant to be a shopping list. Each individual will have different priorities based on budget, the type of music he or she wishes to record, workspace, and many other variables. However, by the end of the chapter you should have a good sense of why you might want to purchase a digital audio interface with an ADAT Optical interface or a mixing console with phantom power and direct outputs on each channel.

Microphones

A transducer is a device "that changes one form of energy into another, corresponding form of energy."[1] In the case of a microphone, the microphone responds to incoming sound waves and converts them into an electrical signal. The selection and use of microphones is a big topic that will be addressed in more detail in chapter 4. However, it will be helpful to consider the primary categories at this point.

Dynamic Microphones (Moving Coil)

Dynamic microphones consist of a diaphragm that is connected to a moving coil suspended in a magnetic field.[2] Fluctuations in sound pressure cause the diaphragm to move in and out, creating a small electrical current. Dynamic microphones such as the Shure SM57 (see figure 1.1) and SM58 are inexpensive and durable. They are used to record a wide variety of sources, including guitar amplifiers, snare or toms, or auxiliary percussion. You can hear these microphones and many others at the companion Web site.

FIGURE 1.1

Three relatively inexpensive microphones: Shure SM57, Rode NT1-A, Cascade Fat Head II

Condenser Microphones

While a dynamic microphone does not require the application of voltage to operate, a condenser microphone contains active components that do require a small amount of power. A mixing console or audio interface is typically used to provide the voltage, known as *phantom power*, to a condenser microphone via a microphone cable.

Condenser microphones are frequently used on voice and acoustic instruments such as bass, guitar, and piano. As you will learn, some condenser microphones provide a switch to enable the selection of a variety of response patterns. A quality condenser microphone will typically cost between $200 and $700. (The Rode NT1-A shown in figure 1.1 is at the lower end of the price range). Higher-end microphones such as the AKG 414 (found in most studios) cost about $1000, while a Neumann can run several thousand dollars.

Ribbon Microphones

Although ribbon microphones were, for a time, an uncommon relic of the past, these microphones seem to be making a comeback. A ribbon microphone is a type of dynamic microphone that contains a thin strip of metal that responds to sound waves. Ribbon microphones tend to sound very natural, although some are fragile and can be damaged by the application on phantom power or even bursts of air. Ribbon microphones are sometimes used on saxophone or other woodwinds, as overheads for drums, as room microphones, or for many other applications including acoustic or electric guitar. Ribbon microphones can be very expensive, although less expensive options such as those made by Cascade are now available (see figure 1.1).

Microphone Stands

Although microphone stands are one of the more mundane purchases for a project studio, they are important and a few words of advice are in order. While inexpensive stands can be used for many tasks, you will want to purchase quality stands for your better microphones. A boom stand will be the best choice for most tasks as it will allow you to position a microphone close to a performer while keeping the stand out of the way of the musician.

In general, a heavier stand will provide better stability, but also consider the dimensions and weight of the base, as well as the quality of the components. Also be sure to consider the length of the boom arm: you will need a fairly long boom arm in order to place a microphone over the bass strings of a piano or over a drum set.

It is easy to underestimate the amount of weight a stand will need to support. For example, some microphone manufacturers make special adapters so that two microphones can be attached to the same stand. A stand would need to be robust in order to support that much weight.

Smaller boom stands can be convenient when placing a microphone close to a kick drum or guitar cabinet, and a large boom stand, such as On Stage's Studio Boom Stand, can be very convenient when setting up an overhead drum microphone or for orchestral recording, when it is desirable to position a microphone above the orchestra.

Preamplifier

A preamplifier is used to boost the relatively weak signal of a microphone and will, at the very least, provide a control to adjust input gain. Most preamps also provide phantom power, as well as other options such as phase reversal and variable input impedance. Preamplifiers are a component of mixing consoles and most audio interfaces so a dedicated preamp is not always necessary. However, preamps can impart a unique color so some engineers use additional preamps in addition to the "stock pres" found in a console or audio interface. Boutique preamplifiers can cost a thousand dollars or more for a single channel and may be utilized by engineers on key tracks such as a lead vocal to impart a special color or warmth. In my estimation, boutique preamps are too often seen as a magic bullet. For example, in my project studio it was first necessary to sonically treat my recording space and develop my skills as an engineer before more expensive preamps were even a consideration—and I still haven't found it necessary to purchase a boutique preamp. You will likely find that the preamps found in a quality audio interface will be perfectly suitable for most applications.

Recording Options

There are several ways to record a signal and the approach you take will be determined by many factors such as portability, stability, flexibility, technical savvy, and preferred work style. The following paragraphs detail four common approaches.

Audio Interface and Computer

A computer-based recording system generally consists of a computer that runs multitrack audio recording software and one or more audio interfaces. An audio interface will typically provide two to eight microphone preamps, as well as any number of instrument inputs. Quarter-inch analog outputs are common, although some audio interfaces may provide the capability to receive and transmit signals digitally. The Mark of the Unicorn (MOTU) interface shown in figure 1.2 is an example of a popular multichannel audio interface.

FIGURE 1.2
MOTU Audio interface

Audio interfaces are most often connected to a computer via FireWire (IEEE 1394), although some interfaces connect via USB. The bandwidth of USB 1 is sufficient for just a few audio tracks, but the specifications for USB 2 are similar to FireWire. Some audio interfaces are connected to the computer via a proprietary cable that connects to a *peripheral component interconnect* (PCI) card installed in an available expansion slot inside the computer.

There are numerous factors to consider if you plan to purchase an audio interface: First, consider the type and number of inputs and outputs (both analog and digital). You will need an interface with multiple microphone preamps if you plan to record several musicians simultaneously. Also, consider if you will need phantom power on each channel. As mentioned previously, phantom power is required to power condenser microphones. Some mixers and audio interfaces provide a single phantom power switch for several channels, which can be problematic if you plan to use condenser microphones and ribbon microphones simultaneously.

It is also helpful to consider future expansion when selecting an audio interface. For this reason, the inclusion of an ADAT Optical input and output is often desirable since a single ADAT Optical port can provide an additional eight channels of input or output via a digital connection.

A primary job of an audio interface is to convert analog signals to digital (AD conversion) and digital to analog (DA), so it is also important to evaluate the quality of the converters. This is obviously a subjective assessment, but there is no question that some interfaces sound better than other. Magazine reviews and online forums can be useful in narrowing choices, but ultimately, your ears should be the primary tool with which you evaluate a given audio interface.

The terms *bit depth* and *sample rate* come into play in several aspects of digital audio recording, and this is a good place to present these terms. When music is digitized, the analog-to-digital converter (ADC) takes a periodic snapshot of an incoming signal. The rate at which data are sampled is the *sample rate*. For example, the sample rate of a CD recording is 44.1 kHz, or 44.1 thousand samples per second. It might be helpful to think of a computer monitor when visualizing sample rate: a line that is drawn on a computer screen is represented by individual pixels. The more pixels that are used, the more realistic or less jaggy the line looks. This is akin to sample rate, as higher sample rates can help a signal to sound better.

The term *bit* stands for *binary digit*—the ones and zeros that form the basis of digital music. Common sense dictates that larger numbers can be represented with more ones and zeroes. *Bit depth* is a related term that refers to the number of bits that make up a number and generally relates to the concept of dynamic range. Consider if just two bits (0 and 1) were used to represent the dynamic range given audio sample—dynamic range would be nonexistent as the samples would either be full volume (1) or off (0). However, if sixteen bits are used (e.g., 1111111111111111), numbers in the range of 0 to 65,535 can be used to represent the dynamic range of a sample of music. Similarly, if twenty-four bits are used, the range is 0 to 16,777,215. Although sixteen bits are used for CD recordings, studios typically operate at a bit depth of twenty-four bits or higher. Using the analogy of a computer monitor, bit depth relates to how realistic colors will look. If just eight bits are used to represent the entire color spectrum, subtle changes between adjacent colors will not be accurately reflected since it is possible to represent only a total of 255 colors with eight bits. Hence, a sixteen- or twenty-four-bit color palette will look more natural. Similarly, higher bit depth can help music to sound more natural.

Although CD recordings have a sample rate of 44.1 kHz, most studios use higher sampling rates, such as 96 kHz or 192 kHz. Although higher sample rates don't necessarily equate with better recordings, I would certainly consider sample rate and bit depth when contemplating the purchase of an audio interface.

A less tangible assessment of an audio interface relates to the quality of the software drivers provided by the company. A *driver* is a type of specialized software that is installed to provide a connection between hardware (such as an audio interface) and your operating system (such as Windows 7 or OS X). Buggy drivers can bring your computer system to its knees, and while it's impossible to tell how reliable a given set of drivers will be, you can often glean insights by visiting user forums prior to making a purchase. For this reason, I only purchase computer hardware from vendors who have a track record of releasing reliable drivers. A quick visit to the manufacturer's download site will help you determine if they are proactive about releasing and updating drivers for their products.

One of the obvious benefits of the combination of an audio interface and computer is the ease with which recordings can be edited, overdubbed, and otherwise processed. For this reason, most project studios use a computer-based recording system. A disadvantage is that a computer may be more unstable than a dedicated recorder (although this is not always the case). We will explore computer optimization in the next chapter.

Mixer and Recorder

Another way to make audio recordings is to use the combination of a mixing console and stand-alone recorder. In this scenario, one or more microphones are connected to a mixer and these signals are passed to a dedicated hardware recorder, such as the Alesis HD24 shown in figure 1.3.

FIGURE 1.3

Alesis HD24

This setup is appealing for a couple of reasons: For some individuals, the tactile sense of using real knobs and sliders can provide a more intuitive experience than controlling virtual knobs and sliders on a computer screen. A dedicated hardware recorder can provide less fuss and better reliability, which can be particularly attractive when doing live recordings. A downside of this system is that punch-ins and other types of editing can be tricky if not impossible to execute. It may be necessary to purchase *outboard gear* in order to do things such as add reverb, delay, or compression. Alternatively, some stand-alone recorders provide the option to export data to a computer, which makes it very convenient to mix and edit tracks.

As with audio interfaces, it is important to consider the number and type of connections, as well as the quality of the microphone preamps, when considering the purchase of a mixer. It is particularly important to note the number of *direct outs,* as the outputs will be used to connect each channel to a corresponding channel on the recording device (see figure 1.4).

FIGURE 1.4

Direct outs on Allen and Heath console

Some mixers do not provide direct outputs on each channel and such a mixer would be at an obvious disadvantage when used for multitrack recording. Although there are several ways to configure a mixer and recorder, one of the most convenient methods is to dedicate half the channels to monitoring input sources and use the other channels to monitor the output of the recording device. Some mixers provide a toggle to switch between a line input and microphone preamp, which makes it feasible to record sixteen tracks on a sixteen-track board. In this instance, the microphone inputs are used when recording and line inputs are used for playback. Note that additional suggestions relating to the purchase of mixing consoles are provided at the end of chapter 6, "Using a Mixer."

Hybrid Mixer/Audio Interface

Several manufacturers, including Mackie and PreSonus, sell hybrid consoles that offer the benefits of an analog mixer with the convenience of a FireWire connection. With this type of system, sound sources are connected to the mixer and signals are digitized and sent to a computer via a FireWire cable.

All-in-One Solutions

There are several all-in-one solutions for digital audio recording. Systems such as the Korg D3200 provide an onboard mixer with multiple inputs, internal hard drive, CD burner, and effects that enable a musician to complete an entire production without using a computer or external processors. Some of the advantages of this type of system include portability and hands-on operation. These systems are at a disadvantage when it comes to editing, as it can be difficult to perform complex tasks in a small editing window and signal processing options are typically less powerful than those found on computer or hybrid systems. Zoom's R16 provides a unique approach to all-in-one recording. The unit can function as a USB audio interface or it can be used as a stand-alone recorder. Recordings are stored on inexpensive SD or SDHC memory cards.

Monitors

Up to this point we have concerned ourselves with capturing an audio signal. The other side of the equation involves monitoring audio. The term *reference monitor* is typically used to describe the speakers that are employed for this task. In an ideal scenario, the monitors will provide an accurate representation of the music you are recording and mixing, and your mixes will translate well to other reference systems. Unfortunately, a variety of acoustical issues will impact the accuracy of your reference system. Unless you are building a control room from scratch and have the expertise to acoustically tune a room, built-in speakers will not be an option. For most of us, near-field monitors are the best choice. Near-field monitors are designed to be placed fairly close to the engineer and can help to minimize acoustic problems in the listening environment.

Reference speakers come in two broad categories: passive and active. A *passive speaker* requires a separate power amplifier, which is used to drive the speaker. In this scenario, the stereo output of your (unpowered) mixer, audio interface, or all-in-one recorder is connected to the input of the power amplifier. The output of the power amplifier is connected via a speaker cable to the input of the speaker, as shown in figure 1.5.

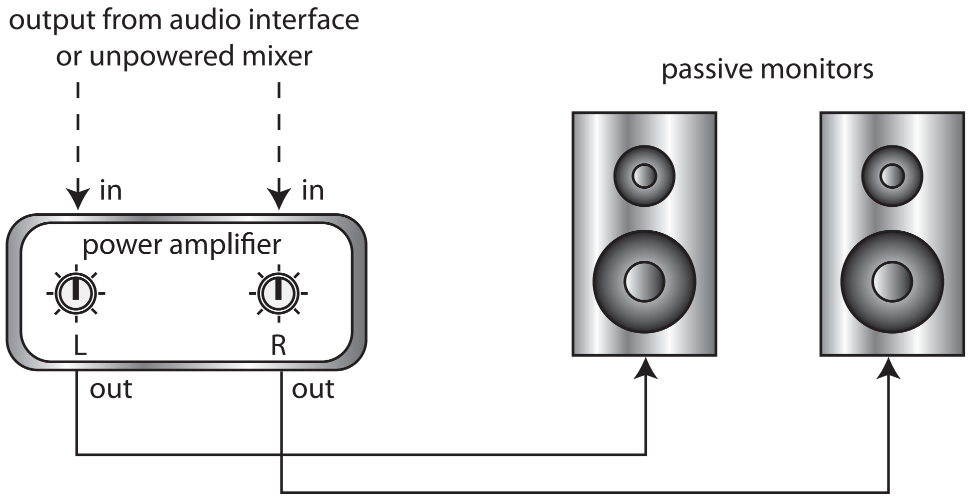


FIGURE 1.5

Connecting a power amplifier to a passive monitor

Although passive monitor systems can perform well and tend to be less expensive than an active setup, most project studio owners opt for *active reference monitors,* as there are several advantages to an active system. One advantage is that each monitor contains a built-in amplifier that has been engineered to produce the best possible results with the given monitor. Some active monitors have a built-in active crossover that can help the speakers work more efficiently and with better clarity. With a bi-amp crossover, one amplifier is used to drive the bass speaker, called a *woofer*, and another amplifier drives the high-frequency speaker, called a *tweeter*. The crossover splits the incoming signal and passes lower frequencies to the woofer and higher frequencies to the tweeter (see figure 1.6). This allows each speaker to function more efficiently and should provide a more accurate representation of a given audio signal.

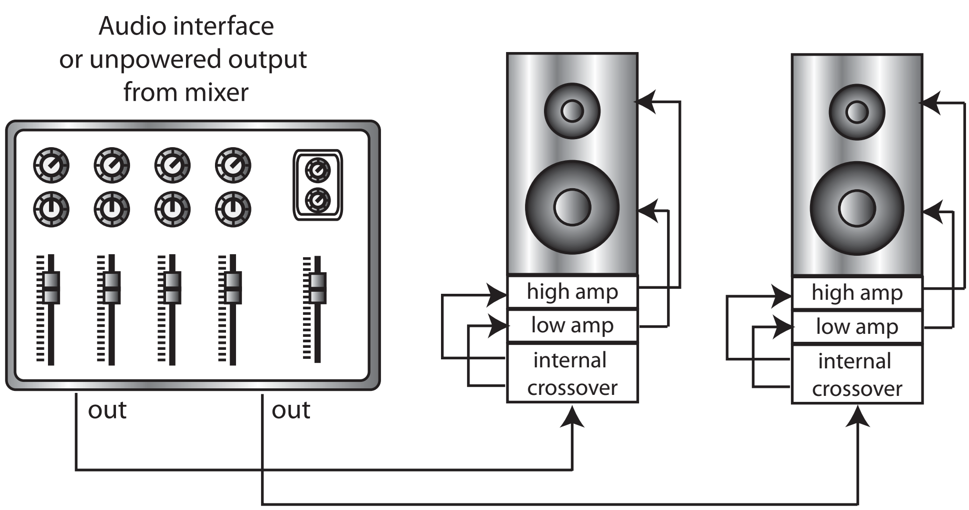


FIGURE 1.6

Active speakers with bi-amp crossover

It is helpful to understand the concept of a frequency response curve when evaluating speakers and other types of devices such as microphones and equalizers. To create a frequency response graph, a signal of constant amplitude but varying frequency is applied to a device.[3] When the output of the device is measured, changes in amplitude for a given frequency (shown on the X-axis) are reflected on the Y-axis. A device that does not boost or reduce specific frequencies is said to have a *flat* frequency response. Figure 1.7 shows a frequency response curve for a Mackie 824, a popular active monitor.

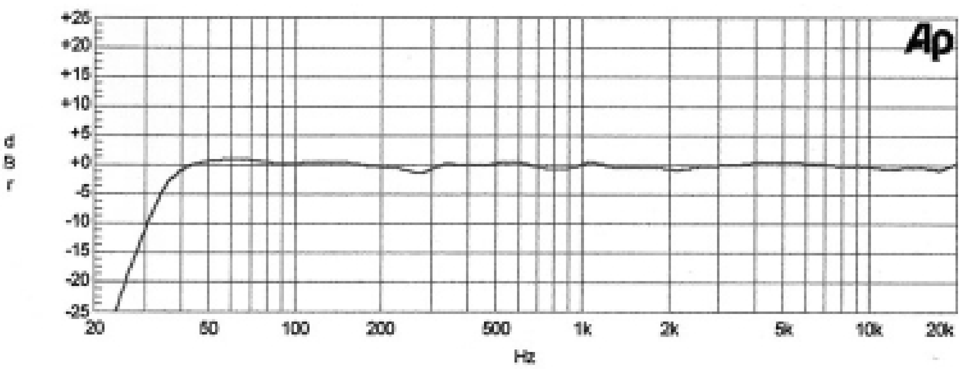


FIGURE 1.7

Frequency response curve: Mackie 824

When selecting a reference system, it is important to distinguish between monitors designed for recreational listening and those that are intended for professional use. Generally speaking, speakers designed for home use will not accurately reproduce a given input—they tend to artificially enhance a signal. While this can be great for recreational listening, it is a bad idea to use such a system for critical listening. A professional reference system is designed to highlight every audio flaw and imbalance so that you can make good mixing decisions.

A subwoofer, or "sub," is a speaker that is designed to reproduce low frequencies down to around 25 or 30 Hz. Although it is not an essential component of a project studio, a subwoofer is required for surround mixing and can be beneficial when mixing some genres of music.

You will want to give forethought when selecting monitors if you think you may eventually do surround mixing: The monitors should be "THX pm3 Approved," and you will want to consider the future cost of purchasing additional monitors. The most common surround format, 5.1, refers to a speaker system consisting of five speakers—left, center, right, left rear, right rear—and one subwoofer called the *low-frequency effects* channel. While it might be reasonable for a project studio owner to spend something in the range of $500 to $800 per monitor for stereo mixing, the cost of adding three additional monitors and a subwoofer for a surround system is likely cost prohibitive. For many musicians, a less expensive pair of reference monitors will be perfectly suitable, and many of the budget offerings are actually quite good. Also keep in mind that, although clarity and accuracy are desirable in a reference system, it is also important to get to know your monitors. For example, even engineers at high-end mastering facilities must learn how a song will translate to a variety of playback systems.

Some manufacturers make reference monitors that purport to adapt to the audio characteristics of a given room. Unless your control room is built from the ground up and designed for monitoring, there will likely be acoustical problems in your listening space that will cause your playback system to emphasize or de-emphasize certain frequencies. The idea behind adaptive monitors is that they can be calibrated to adapt to a given room. Monitors such as the JBL LSR4328P come with a calibration microphone and are able to self-calibrate to a given room. The speakers emit test tones that are picked up by a calibration microphone. The speakers then analyze the incoming signal and utilize filters to adapt themselves to the given room characteristics. In this way, the speakers are able to tailor themselves to a variety of listening environments. Although these systems are still relatively expensive, the concept is certainly attractive and worth further exploration if you are in the market for reference monitors.

One final caveat is in order: it is usually a good idea to purchase isolation pads for your reference monitors. Isolation pads can help to minimize coupling that can occur when the speakers vibrate and transfer the vibration to your speaker stands or mixing desk. My anecdotal experience has been that isolation pads are worth the additional expense.

Headphones

While it is not possible to rely solely on headphones when mixing, I find them to be helpful when used in conjunction with near-field monitors. I tend to use headphones when checking the position of instruments in a mix (pan), as well as when checking the amount of reverb or other effects. In some cases, it is possible to catch problems with a reverb tail or other subtle noises when checking a mix with headphones.

In addition to a pair of quality headphones dedicated to mixing, one or more pair of closed-ear headphones will be necessary if you plan to have musicians overdub or play to a click track. Closed-ear headphones surround the ears and help prevent leakage of the headphone feed into microphones. While it is important for the musicians to be comfortable and hear a pleasing mix in their headphones, my experience has been that tracking headphones receive hard use so I typically purchase tracking headphones that are relatively inexpensive.

Headphone Distribution Amplifier

A headphone distribution amplifier (see figure 1.8) is essential when musicians are acoustically isolated in a studio. To use a headphone distribution amplifier, connect the output of a headphone mix (or main mix) from your audio interface or console to the inputs of the headphone amplifier. Then connect one or more headphones to the amplifier so that the musicians can set individual levels. I always set each headphone to a low level and play a loop of some comfortable music through the headphone amplifier. This makes it possible for the musicians to adjust their headphones at any time when they arrive and start setting up for a session.

FIGURE 1.8
Headphone distribution amplifier

Analog Connections

Equipment such as microphones and audio interface are only part of the equation. The type and quality of cables that are used to connect the components of a project studio are also very important, and it's easy to underestimate the cost of outfitting your studio with quality cables. For example, a five-foot cable snake consisting of eight balanced cables may cost around $125 dollars. Depending on configuration, a sixteen-channel studio might require four sets (16 inputs and 16 outputs) in order to connect to and from a hard-disk recorder.

Balanced Signals

In general, you will want to purchase gear with balanced inputs and outputs and use balanced cables to connect the components of your system. Balanced cabling will help you maintain a pristine signal chain and avoid hum and other problems in your studio.

A balanced cable consists of two signal wires and a shield. In a balanced system, the source device transmits a signal down both signal wires, but the polarity of one of the signals is reversed: one wire is "hot" (+) and the other is "cold" (−). When the signal reaches the destination device, the cold signal is inverted and summed with the hot signal. Noise that enters the cable has the same polarity for each wire and, thus, is removed when the cold signal is inverted.

Although this concept might seem counterintuitive at first, the application of some arbitrary terms may help. Consider a balanced signal called "music." "Music" is transmitted down the hot cable as +music and down the cold cable as

−music. At its destination, the negative signal is inverted and −music becomes +music and the two signals are combined: +music + (+music) = 2 × music. Now consider unwanted external noise (we will call this "noise"), which enters the cable while "music" is being transmitted. Since "noise" doesn't have the benefit of being generated by a balanced source, it enters both signal wires *with the same polarity* (e.g., +noise and +noise). When the cold signal is inverted, one copy of noise becomes −noise while the other remains +noise. When the signal is summed, "noise" cancels itself out: +noise + (−noise) = 0 noise.

XLR

XLR cables (figure 1.9) are typically used to connect a microphone to a preamp, but they may also be used to make a balanced connection between a mixer and active speakers or between a microphone preamp, audio interface, or mixing console.

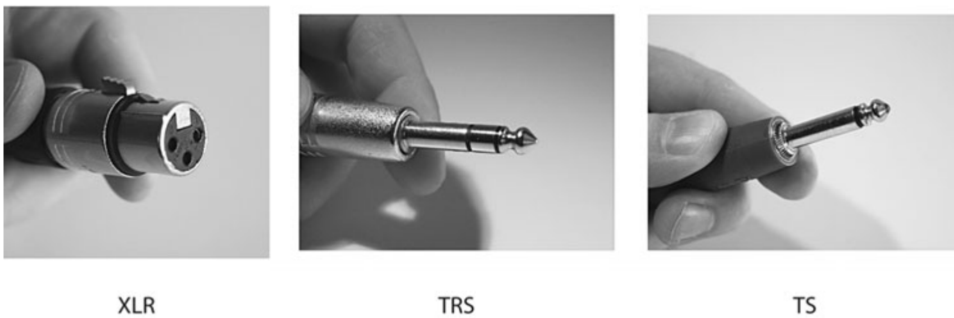


FIGURE 1.9
XLR, TRS, and TS plugs

TRS

The "tip-ring-sleeve" TRS plug (alternatively, TSR) is similar to an XLR plug in that it is often used to make a balanced connection between components, although TRS plugs can also carry an unbalanced stereo signal. TRS cables consist of three conductors: two of the conductors carry a signal and the third conductor is the ground or sleeve. It's easy to spot a TRS cable because the plugs have two insulators, as can be seen in figure 1.9.

TS cable

A "tip-sleeve" cable (figure 1.8) is an unbalanced cable consisting of a two conductors (signal and ground or sleeve). TS cables are typically used to connect unbalanced gear such as an electric guitar or bass to an amplifier.

DI

A DI box, called a direct injection or direct interface, is used when it is necessary to connect a high-impedance source such as an electric bass or keyboard to a low-impedance balanced microphone input on a mixing console. Direct boxes come in two primary flavors: active and passive. An active direct box (see figure 1.10) requires the application of power (via a battery or phantom power) to function while the circuits of a passive direct box do not require power. DIs help to prevent signal coloration that can occur when a high-impedance source is connected to a low-impedance destination, and they are also necessary to prevent hum and other extraneous noise that can be introduced in longer unbalanced cable runs.

FIGURE 1.10

Active direct box

Snake

The term *snake* simply refers to a collection of cables that have been banded together. Cable snakes such as a DB25 to TRS or ¼-inch TRS to ¼-inch TRS are useful in minimizing clutter when patching multiple cables to and from a console or interface.

While many studios utilize in-wall microphone sockets, a microphone snake can be very convenient, as it will allow you to plug multiple microphones to a box at one end of the snake. At the other end, multiple ¼-inch or XLR plugs can be conveniently connected to an audio interface or console keeping cable clutter to a minimum.

Digital Connections

Whereas analog cables transmit fluctuating voltages that are *analogous* to sound pressure waves, digital cables transmit the ones and zeros that constitute a digital signal. Most project studios will utilize a variety of analog and digital connections. Note that a digital source should never be connected to an analog destination and vice versa, even though the connectors are compatible in some cases.

S/PDIF

S/PDIF (usually pronounced as "Spee Diff") stands for Sony/Phillips Digital Interface and is used to transmit two channels of digital audio information via an unbalanced coaxial or optical cable (see figure 1.11). Although S/PDIF is generally considered to be a consumer interface, S/PDIF is utilized on some professional equipment.

S/PDIF AES-EBU Optical

FIGURE 1.11
S/PDIF, AES/EBU, and Optical cables

AES/EBU

AES/EBU is the "pro" version of two-track digital audio (see figure 1.11). This interface, specified by the Audio Engineering Society and European Broadcast Union, transmits digital data via a balanced line with XLR connectors.

ADAT Optical Interface

The ADAT Optical Interface, also called Lightpipe, utilizes an optical cable (see figure 1.11) to transmit eight channels of digital audio. Although Lightpipe was developed by Alesis, manufacturer of a popular series of multitrack recorders called the ADAT, many manufacturers provide ADAT connectivity. For example, a digital audio interface may provide an ADAT Optical interface, which makes it easy to connect to a second interface or digital preamplifier for additional inputs. You will need to configure your audio interface to synchronize to the external ADAT device in order to avoid clicks, pops, and other problems.

Other

As you can see, there are many items to consider when planning a project studio. Other topics such as acoustical treatments will be presented in the chapters that follow. Although most of the previous items are found in a typical project studio, the following items represent optional equipment that may or may not be germane to your particular situation.

Patch Bay

An analog patch bay may be useful if you frequently change signal paths, as it will allow you to easily connect the output of one device to the input of another without rummaging around in the back of a rack or mixing console. Typically,

FIGURE 1.12
Patch bay

equipment outputs such as the output of a microphone preamp or compressor are connected to the top row of jacks in the back of the patch bay. Inputs such as a channel input, auxiliary return, or compressor are connected to the bottom row of jacks in the back of the patch bay.

With regard to patch bays, the term *normal* refers to an internal connection that is made when the output of a device such as a reverb unit is "normally" connected to the input of another device such as an auxiliary return. In a normaled patch bay, an internal connection is made when a source and destination are plugged into the top and bottom jack of a column in the back of the device. In this instance, it is not necessary to use a patch cable to connect the devices from the front of the patch bay.

In a *half-normaled* connection, the normaled connection is broken only if a plug is inserted into the lower jack in the front of the patch bay. This means that a copy of a signal is available by plugging into a jack in the top row, but when plugged into a destination jack on the bottom row, the normal of the destination is broken.

Power Conditioner

A power conditioner can help protect your equipment against power surges and may even provide some sonic benefit by filtering certain types of electrical noise. A similar device, the *uninterruptable power supply,* or UPS, can be attached to a computer or other device in order to provide a few minutes of battery power in the event of a power outage or brownout.

Music Instrument Digital Interface

While not normally needed for recording audio, a MIDI interface will be necessary if you wish to connect a MIDI keyboard or other types of MIDI equipment to your computer. Most digital audio interfaces will provide at least one MIDI in and out port, although a multiport MIDI interface will be useful if you intend to use more than one MIDI device. Note that some Digital Audio Workstation controllers utilize MIDI for communication with the computer (see below).

Control Surface

One of the disadvantages of using the combination of a computer and digital audio interface is that you lose some of the tactile benefits of working with real knobs and faders. A workstation controller, such as the Mackie Control Universal (see figure 1.13), features motorized faders and provides a tactile connection to your recording software.

FIGURE 1.13

Mackie Control Universal

Studio Furniture

A typical studio will include a mixing desk and one or more equipment racks. Near-field monitors may be mounted on a bridge above the mixing desk or placed on speaker stands. While studio furniture can cost thousands of dollars, it is not difficult to make simple studio furniture. The final chapter of the book features step-by-step instructions for creating the attractive mixing desk, speaker stands, and equipment rack shown in figure 1.14.

FIGURE 1.14

Custom studio furniture: Mixing desk, speaker stands, and rack

Studio Ergonomics

Of course, a studio will not be fun to use if it is not ergonomic. In general, reference monitors should sit at ear height and computer keyboard, mouse, and frequently used knobs and faders should be within easy reach. Level indicators for microphone preamps and audio interface should also be in the line of sight and the computer monitor should be adjusted to a comfortable height.

Practical Application: Four Budget Project Studios

In the pages that follow we will consider outfitting four hypothetical project studios. As I mentioned at the start of this chapter, this material should not be considered a shopping list since individual needs will vary and products and prices change. Rather, the discussion is meant to demonstrate a thought process and to highlight the categories of equipment that will be useful for each application.

Budget Songwriting Studio

This sample configuration will be built around a computer and an audio interface with at least two microphone preamps (see table 1.1). Two preamps will provide the opportunity to simultaneously record voice and an instrument such as guitar. Note that, for many musicians, a four-channel interface would be a

TABLE 1.1 Sample equipment list: Songwriting studio

Item	Cost	Notes
New or used PC or Mac computer with FireWire	\$0–\$800+	Most musicians already own a PC or Mac. See chapter 2 for a discussion of computers and software for audio recording.
Focusrite Saffire PRO 24, PreSonus FireStudio Mobile, or similar FireWire interface	~\$300	Many interfaces are available in this price range. Note that the FireStudio does not include an ADAT port for additional inputs.
Shure SM57 or SM58 dynamic microphone	\$100	Great for recording electric guitar amp, voice, snare, etc.
Rode NT1-A, Sterling ST51, Studio Projects B1 or similar large-diaphragm condenser microphone	\$100–\$200+	Many options available in the \$100–\$200 price range.
Headphones: Audio-Technica ATH-M50, Sennheiser HD 280 PRO, Sony MDR 7506, etc.	~\$100	Be sure to purchase closed headphones so sound doesn't bleed into the microphone.
Other: microphone stand, microphone cables	~\$150	

better choice in order to overdub a drum kit or record voice and acoustic guitar (which is sometimes recorded with two microphones).

Other Notes

Many songwriters will want to use prerecorded loops. Steinberg's Sequel 2 costs \$100, comes with many loops, and runs on PC and Mac. If you already own a Mac, Apple's Garage Band software (\$80 as part of the iLife suite) is a capable recording software and comes with a fair assortment of loops. Note that both of these products have limited track counts.

The selection of microphones can be challenging since there are so many variables involved. Dynamic mics, such as the SM57 and SM58, work well for voice and the "57" is often used to record guitar cabinets and snare drum. For acoustic instruments, one or two condenser microphones might be a better choice.

Upgrades

The addition of near-field monitors would be a good upgrade for some musicians. There are also many fine FireWire interfaces available in the range of \$200 to \$500. For example, the Apogee Duet is a highly regarded two-channel interface, but many multichannel interfaces are available in a similar price range.

Computer-Based Multitrack Studio

This system is built around an eight-channel audio interface and Mac or PC computer (see table 1.2).

TABLE 1.2 Sample equipment list: Computer-based multitrack studio

Item	Cost	Notes
New or used Mac or PC	$0–$800+	Most musicians already own a PC or Mac. See chapter 2 for a discussion of computers and software for audio recording.
*MOTU 8Pre, PreSonus FireStudio Project, Focusrite Saffire Pro 40, MOTU 828mk3 or similar audio interface	$500–$750	Many options are available from a variety of vendors in the $500–$800 range. Consider ADAT I/O for future expandability.
Active monitors: Yamaha HS80M Mackie HR624mk2, Alesis M1, etc.	$300–$1000 (pair)	Many options available in this price range.
Two Shure SM57 or SM58 dynamic microphones	$200	Great for recording electric guitar amp, voice, snare, etc.
Two AKG Perception 170, Rode NT5 or similar small diaphragm microphones	$275–$425	A good choice for drum overhead, acoustic guitar, and piano.
At least one large-diaphragm condenser microphone (AKG, Shure, Rode, etc.)	$200–$500	Large-diaphragm condensers are typically used on voice and acoustic instruments such as sax, violin, etc.
Primary headphones: Audio-Technica ATH-M50, Beyerdynamic DT 770 PRO, etc.	$150–$250	
ART HeadAmp4 or PreSonus HP4 headphone amplifier (four channel)	$65–$125	
1–4 tracking headphones: AKG K 77, Samson CH70, etc.	$40–$50 each	Be sure to purchase closed headphones so sound doesn't bleed into microphones.
Microphone stands and cables	~$300	
*Software: Multitrack recording software may not be necessary depending on software package that comes with audio interface.	$0–$500	See chapter 2 for a discussion of multitrack recording software.

Other Notes

The MOTU 8Pre comes with capable digital audio recording software called AudioDesk, which runs on Apple computers, and the FireStudio and Saffire each come with entry-level recording applications. Also consider a hybrid mixer/audio interface such as the Mackie Onyx series or PreSonus StudioLive, as these mixers are a good option for many musicians and bands.

The selection of reference speakers can be as daunting as purchasing a new microphone. I read a favorable review of the Yamaha HS80M monitors, which

TABLE 1.3 Sample equipment list: Hardware-based multitrack studio

Item	Cost	Notes
See table 1.2 for other primary components.		
Allen and Heath MixWizard3 (16 ch.), GL2400-24 (24 ch.) or similar.	$1000–$2000	
Alesis HD24 or HD24XR	$1600–$2000	
Cable snakes to connect mixing console to recorder.	$75–$100+ per 8 channels	Custom cables can be constructed more economically: see chapter 12.
Active direct box	~ $125 each	Used to connect a bass, electric guitar, or keyboard directly to the mixing console.

were designed to replace the ubiquitous NS10s, a staple of many recording studios. Mackie monitors are frequently seen in project studios, and I have generally been pleased with my Mackie HR824s. Less expensive alternatives like the Alesis M1 are viable options for many musicians. When possible, see if you can try a set of monitors prior to making a purchase and evaluate the monitors with a recording you are very familiar with.

Hardware-Based Multitrack Studio

A hardware-based studio is a variation on the computer-based studio listed above (see table 1.3). Consider an analog (or digital mixer) and stand-alone recorder in lieu of the computer and audio interface listed above.

This type of system is great if you plan to simultaneously record many instruments. In addition to reference monitors, you will need to purchase enough microphones, stands, and cables to support your desired simultaneous track count.

Music Educator's Portable Studio

This studio is geared to the needs of a band or choir director who intends to make quality recordings of a rehearsal or concert (see table 1.4). A key consideration is ease of use and portability. For these reasons, I would suggest an all-in-one solution along the lines of Zoom R16, Korg D888 or Tascam 2488neo, coupled with two quality condenser microphones. A pair of Oktava small-diaphragm microphones would be a good start for recording in stereo in an XY, ORTF, or spaced-pair configuration (see chapter 4), and two additional omnidirectional or multipattern microphones could be added to better capture a larger choir or orchestra. Oktava microphones are manufactured in Russia and are somewhat "under the radar," but they seem to have a good following among

TABLE 1.4 Sample equipment list: Music educator's portable studio

Item	Cost	Notes
See table 1.2 for other primary components, including headphones, reference monitors, (optional) and cables		
All-in-one recorder: Zoom R16, Tascam 2488neo, Korg D888 or quality stereo recorder	\$600–\$800	
Matched pair of small-diaphragm condenser microphones: Oktava MK012, Shure KSM137, Neumann KM 184 or similar	\$550–\$1700 (pair)	Less-expensive options are available but are not recom mended for critical recording.
Pair of on-stage SB9600 or similar tripod stands. (Only one stand will be required for XY or ORTF recording if a microphone bar is used.)	\$50–\$120 each	Some stands used for photography lighting have a compatible stud size and are generally less expensive than stands marketed to audio professionals.

some audio engineers. I own a pair and have been pleased with them on a variety of acoustic instruments.

Conclusion

We have covered a wide range of equipment in this chapter. As I have already stressed, your situation and budget are unique, and my intent in listing specific products is to provide insights into a thought process that will help you make good purchasing decisions. The rate of change is obviously fast in the realm of digital audio so I would encourage you to read trade magazines and user manuals and take advantage of the many online resources that provide information about the products you intend to purchase.

2

Computers and Software for Audio Recording

A computer is at the heart of most modern project studios. In this chapter, you will learn about the primary components of a computer and how they relate to digital audio recording, consider the variables that will help you to make good purchasing decisions, and learn how to optimize a computer for audio recording. This chapter will also provide the necessary background information for building a custom PC. Step-by-step instructions are provided in the final chapter of the book.

Primary Components

Computers are comprised of many interrelated hardware and software components, and it can be a challenge to know how to prioritize these components when purchasing a new computer or upgrading an old computer for audio recording. Just as with other facets of audio production, there will be tradeoffs when balancing price and performance. The first section of the chapter will focus on the function of the primary hardware components. The final section of the chapter is devoted to a discussion of audio recording software and system maintenance.

CPU

Computers are built around a microprocessor, a small integrated circuit that functions as the "brain," or central processing unit (CPU; see figure 2.1). Although computers are capable of doing any number of amazing things, the CPU is capable of only a limited number of tasks that are defined by its instruction set.

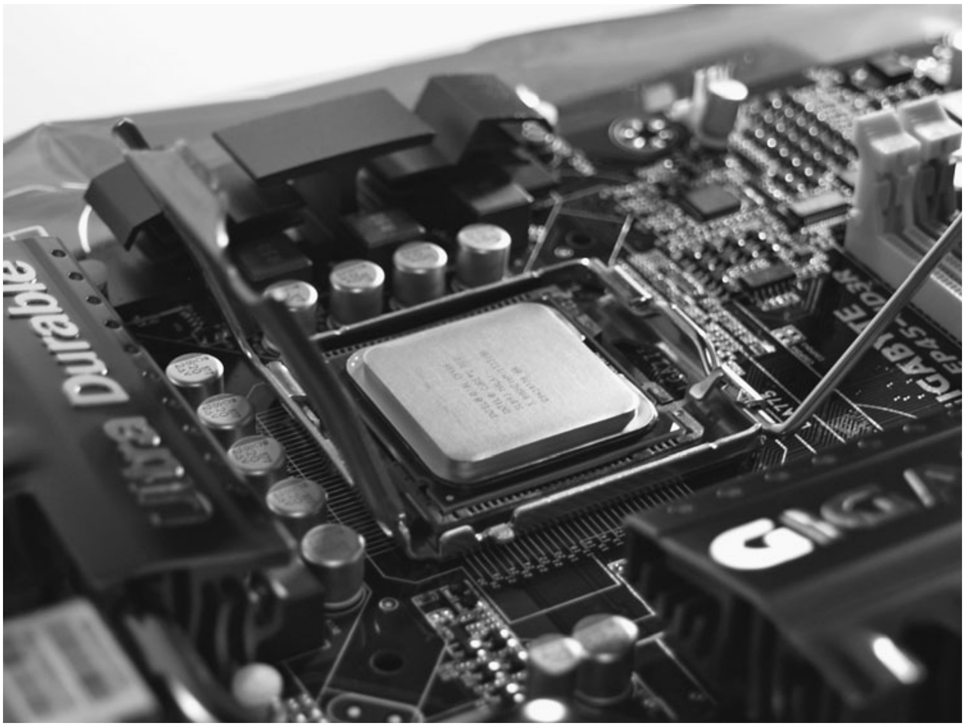


FIGURE 2.1

Central processing unit (CPU)

At a higher level, programmers use languages such as C++ to write code in a human-readable form, and a *compiler* converts these complex sets of instructions into the machine code that is capable of being executed by the CPU (see figure 2.2).

FIGURE 2.2

Computer code into machine code

With regard to digital audio, musicians are primarily concerned with the speed and efficiency of the processor. The term gigahertz (GHz) reflects the clock rate of a microprocessor and is often used as a general measure of speed. For example, a 2 GHz processor will be twice as fast as a 1 GHz processor, all things being equal. However, it is important to realize that the efficiency of the processor comes into play, as some instructions may be executed more or less efficiently depending on the design of the processor. It is also helpful to realize that other data bottlenecks will occur throughout a computer system so gigahertz should be seen only as a general guideline when evaluating a system.

With regard to recording music, fast processors are primarily useful when using multiple real-time plug-ins. As the name implies, a real-time plug-in such as a digital reverb or dynamic processor must process audio as data is recorded or played back from disk—usually within a few milliseconds. Since plug-ins require complex calculations, a faster processor may provide the ability to run a greater number of simultaneous plug-ins. For the same reason, a fast processor will also be helpful if you intend to utilize virtual synthesizers.

Some computers contain multiprocessors, or two or more processors that can work together to provide additional processing power. Related terms such as *multicore, dual-core*, and *quad-core* reflect multiple CPUs in a single unit. Multicore systems are only useful if programs are written to take advantage of the extra processing power. A related term, *distributed processing*, is also making inroads in digital audio recording. Through distributed processing, applications such as Apple's Logic Pro can utilize the processing power of other networked computers.

Motherboard

Using the analogy of a body, whereby the CPU represents the brain of a computer, the motherboard is akin to the nervous system. The motherboard houses the CPU and memory and provides a network of connections between the CPU and other parts of the system, such as the video card, hard-drive controllers, and expansion slots, as shown in figure 2.3.

FIGURE 2.3

Motherboard

Internal Hard Drive

A hard drive contains one or more rotating magnetic platters that can be used to store data, such as a digital audio recording. The drive controller provides an interface between the hard drive and the motherboard. Although there are many types of hard-drive interfaces, two primary categories are typically used: Parallel ATA (PATA), a category of drives also known as Enhanced integrated device electronics (EIDE); and Serial ATA (SATA). Most newer computer systems utilize SATA drives while IDE drives are commonly found in older systems, as well as some recording devices such as the Alesis HD24. For many years, SCSI (Small Computer System Interface) drives were used in recording studios, but IDE and SATA drives are more common in home recording studios today.

There are several things to consider when evaluating a hard drive: size, RPM, sustained transfer rate, seek time, and cache. You will also want to do some research to see if users find the drive to be quiet, as a noisy drive can be problematic in a studio environment.

Revolutions per Minute

The revolutions per minute, or rpm, refers to the rotational speed of the hard-drive platter and is one indication of how fast data can be written to and read from the disk. There is a direct correlation between rpm and track count, which is why musicians favor faster drives. I would currently consider 7200 rpm as a minimal speed for multitrack recording, and some musicians opt for faster, 10,000 rpm drives. Earlier in the chapter I mentioned system bottlenecks and hard drives are a prime example of these: some musicians will opt for a 10,000 rpm system drive (the system drive is the drive that hosts the operating system) as it will provide a performance boost that can help a computer feel more responsive.

Bandwidth and Transfer Rate

There are several factors that can affect the speed at which data are moved to and from the drive. Some manufacturers list a specification called the *sustained transfer rate*, a number that provides an indication of a drive's performance. Audio recording involves the continuous transfer of large amounts of data to and from a drive so the sustained transfer rate is a more important number to consider than the *burst rate*, which represents a best-case scenario.

Where larger numbers are desirable for data transfer, smaller numbers are desirable for seek time. *Seek time* refers to the number of milliseconds it takes for a drive to find data. Just like a person looking through a file cabinet, drives with faster seek times allow data to move more quickly to and from the disk.

Some hard drives have a *cache*, a small amount of volatile memory that can help the drive work more efficiently. You might think of the cache as a sort of holding tank for data. The cache (or buffer) can hold data in reserve if the hard drive has a hard time keeping up with intensive reading or writing operations.

Hard Drive Size

A single track of stereo digital audio requires about 10 MB (*megabytes*) of storage *per minute* at a sample rate of 44.1 kHz and bit depth of sixteen bits. (A megabyte is equal to about 1 million bytes). Today, most drives are measured in *gigabytes* (GB), which represents a billion bytes, or *terabytes* (TB), which represents an astounding million-million bytes.

How much is enough? In practical terms, modern operating systems will need 15 to 20 GB of space and an additional 50 GB of space for applications so a drive in the range of 250 to 500 GB is ample for a system drive. Most musicians will want to use a secondary hard drive (either internal or external) of at least 250 GB for recording. Although the system drive can usually be used for projects with just a few tracks, a dedicated recording hard drive will allow digital audio data to be transferred more efficiently and should yield higher track counts and better stability. Sample libraries such as drum loops, instrument samples, and virtual pianos can be quite large so plan on 500 GB to 1 TB if you intend to use lots of commercial sample libraries. Fortunately, hard drives are relatively inexpensive so it is economical to purchase plenty of space for recording and backups.

RAM

Random-access memory, or RAM, is another critical aspect of a computer. Although there are many different types of RAM, the function is the same: to provide a location to store temporary data. For example, a program might reserve a block of memory to temporarily store a segment of an audio recording so that calculations can be performed on the data and the result stored in another block of reserved memory. However, if the computer runs low on RAM, the blocks of data may be temporarily stored on disk in the form of a *swap file*. In general, swap files, or "paging," is a bad thing where digital audio is concerned since problems can arise when the operating system attempts to store data to disk at the same time as a program attempts to record multiple tracks of data. Although it's not always possible to avoid swap files in a modern operating system, using ample RAM can help. Be sure to check the RAM requirements for any software you intend to purchase. Today, 2 GB of RAM is adequate for many tasks, although many musicians will utilize 4 GB or more of RAM.

RAM can sometimes be the source of computer glitches. Be particularly attentive to system crashes and other anomalies if you upgrade RAM, as these problems may indicate a system incompatibility with the upgrade. It is also important to note that RAM is volatile; anything residing in RAM will be lost if there is a power loss, which is why it is important to frequently save your work to disk.

Ports

The motherboard will host a number of different ports to enable the connection of external hard drives, printers, DVD burners, and the like. The three most important ports for digital audio are FireWire (IEEE 1394), eSATA, and USB.

FireWire

FireWire ports are used to connect devices such as an audio interface or external hard drive to a computer. Two versions are currently in use: FireWire 400 provides a maximum transfer speed of 400 megabits per second while FireWire 800 can transfer 800 megabits or more per second. The majority of professional audio interfaces utilize a FireWire connection.

eSATA

An eSATA connection is typically used to attach an external hard drive. Transfer speed is in the range of 3000 megabits per second, which makes these devices desirable for audio work.

USB

The USB 1 specification is too slow for multitrack digital audio work, but the bandwidth is ample for a MIDI interface. USB 1 is primarily relegated to connecting peripherals such as a typing keyboard, mouse, and printer. USB 2 provides a faster transfer speed of 480 megabits per second, making it suitable for multitrack digital audio. Some manufacturers make USB 2 audio interfaces and USB 2 external hard drives are very common.

Expansion Slots

The motherboard of a desktop computer will provide several slots that can accept expansion cards. Cards might include a video card, modem, or wireless card, but several professional audio interfaces require the installation of a PCI card. We will cover expansion slots in more detail in the final chapter of the book.

System Software

Up to this point we have covered the primary hardware components of a computer system. However, without an operating system and drivers, the hardware components are just so much metal and plastic.

Firmware

When you turn a computer on, a special type of software is run. This software, which is called firmware, or BIOS (Basic Input Output System) depending on the context, is primarily responsible for booting the computer ("pull itself up by the bootstraps,"[1] as it were) and helping the computer find and load the operating system. Other types of hardware such as an audio interface also contain firmware. Firmware may periodically need to be upgraded, and it's a good idea to back up all of your data since a firmware update is considered an extreme update, as there is a higher potential for data loss.

Drivers

Drivers provide an interface between hardware such as an audio interface, MIDI keyboard, printer, or other device and the operating system. A conceptualization of this relationship can be found in figure 2.4.

FIGURE 2.4
Relationship among hardware, drivers, and OS

Drivers are particularly important in digital audio recording since ineffectual drivers can cause problems such as clicks and pops and system crashes. For this reason, it is very important to determine if new drivers are available for all of your hardware when updating to a new operating system. Many computer bugs result from driver problems so it is a good idea to hold off upgrading your operating system or other components if you are in the middle of a critical project. Many computer glitches can be resolved by updating the drivers, so if you experience problems with new hardware, check the manufacturer's site to see if they have posted new drivers. Sometimes driver problems may involve a seemingly unrelated system component. For example, a video driver may cause a crash during an audio recording session. For this reason, it's always a good idea to read any crash logs or error messages to try to determine the source of the problem.

Operating System

The operating system is responsible for the user interface, running the software applications, and coordinating the use of computer resources (see figure 2.5).

On the PC side, three operating systems are typically used for audio recording: Windows XP, Windows Vista, and Windows 7. Although many musicians still run XP, Microsoft will soon stop supporting the system so a transition to Windows 7 seems inevitable for most musicians. On the Mac side, musicians typically use a current version of OS X. It is important to note that some manufacturers such as Digidesign, makers of the industry standard Pro Tools application, require the use of specific versions of the PC or Mac operating systems.

Apple provides an interesting option for musicians who wish to dual-boot in Windows and OS X. Apple's Bootcamp application can be used to configure a hard drive to dual-boot in both operating systems. This effectively means that an Intel Macintosh can boot as either a full-fledged Windows or an Apple

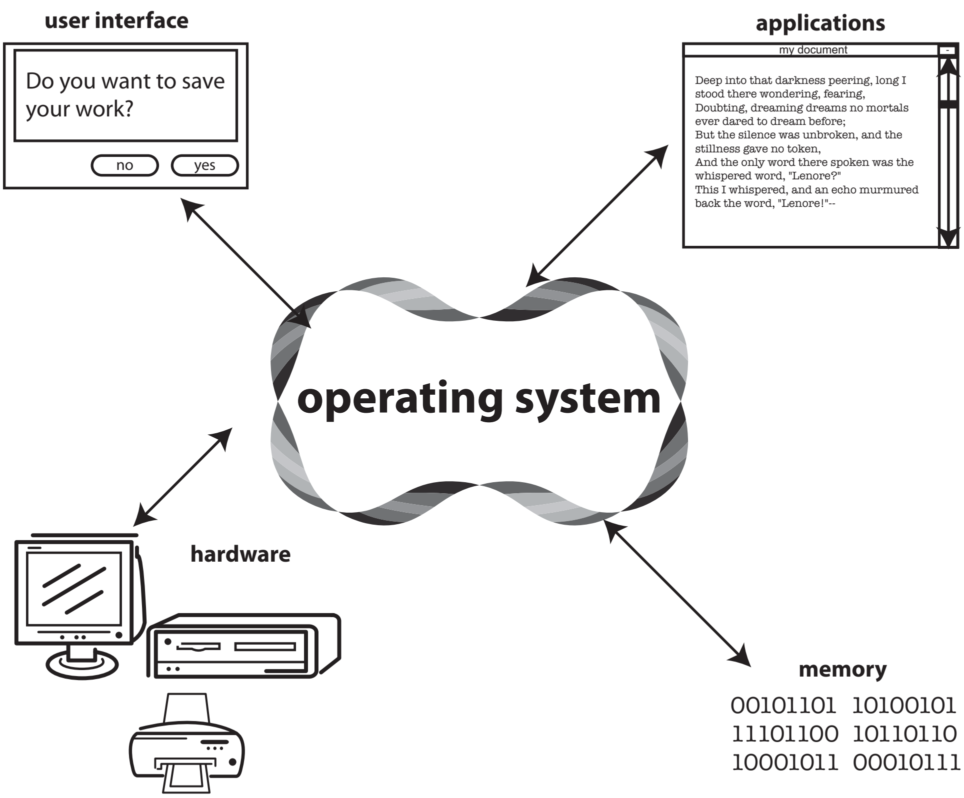


FIGURE 2.5

Conceptualization of an operating system

machine. Unlike virtualization software such as Parallels or VMware Fusion, a Windows installation via Bootcamp utilizes native computer hardware without a software virtualization layer and, thus, Bootcamp is typically the best approach for musicians who intend to run Mac and PC software nonconcurrently. With this approach, you select an operating system (OS X or Windows) at boot time.

Linux provides an interesting alternative for some musicians, but an inherent problem with Linux is the lack of device drivers for professional audio equipment. It is important to stress that, without drivers, a given hardware component will not work with an operating system. Many versions of Linux (called distributions, or "distros") are freely available for download, and some distros, such as Ubuntu Studio (http://ubuntustudio.org), are geared specifically for music and video.

One of the biggest benefits of Linux is that distributions and applications are freely available for download. Distributions typically come with a "live" version, which means that you can burn a disk image of Linux and boot your computer directly from the live CD. This is a terrific way to explore audio recording without any expenditure of cash. For educators, Linux can be an attractive option, given that the operating system and applications are free and Linux will typically run well on older computer systems. For a budget-conscious musician, Linux has an excellent potential *if* you match a distro with the appropriate hardware.

I spent many hours with Ubuntu Studio as part of the research for this book and was impressed with the many fine applications for audio production. Unfortunately, it was a challenge to configure the operating system to work with my hardware. I tested five audio interfaces but the operating system only recognized one device—an older M-Audio USB interface. There appear to be ongoing efforts to develop new open-source drivers, and I anticipate the operating system will be a good choice for musicians as more device drivers become available and the process for installing new drivers is streamlined. At present, I would only recommend Linux for musicians who have the time and aptitude to tinker with the operating system.

Purchasing a Computer

Consider the following tips when it comes time to purchase a new computer. In general, it is *not* necessary to purchase a top-tier "screamer" for digital audio work.

Chose Primary Applications

Students and colleagues frequently ask for recommendations when purchasing a new computer, and a common question is, "Should I buy a Mac or a PC?" I always advise to think in terms of software first: you will need to purchase a Mac if you intend to run MOTU's Digital Performer or Apple's Logic Pro. You will need to purchase a PC-compatible computer if you want to run programs like Cakewalk's SONAR or Fruity Loop Studio. Several vendors, such as Digidesign, Steinberg, and Ableton, offer products that run on both systems. Although you will find zealous proponents of each system in online forums, I can assure you that Macs and PCs are each capable of creating great music. Both systems have their strengths and, unfortunately, each system has the potential for crashes and other problems.

Evaluate System Requirements

I would suggest visiting software vendor Web sites as a second step in researching the purchase of a new computer. Vendors typically list the minimum and recommended requirements for their software. In general, steer clear of the minimum requirements, as these numbers generally represent a poor level of performance. The recommended requirements will provide the parameters for a computer purchase. For example, a visit to Cakewalk's Web site indicates that SONAR 8.5 will run well on a machine with Windows XP, Vista or Windows 7, a 2.8 GHz processor, and at least one GB of RAM.

Consider Portability

Think if your needs will be best served by a laptop or a desktop. While most modern laptops are capable of providing plenty of power for multitrack recording, they offer limited expansion capability and are not well suited for rigorous recording sessions. A primary concern is heat: electrical components are necessarily in close proximity in a laptop and heat buildup can be a problem when running a session with high track counts. On the other hand, a laptop will likely be the best choice for location and field recording.

Consider Future Expansion

As I mentioned previously, laptops provide fewer expansion options. If looking at a desktop, consider if you will need additional bays to house extra hard drives or expansion slots for audio and video cards. It is also important to look at the type and number of ports, such as FireWire, USB 2, and eSATA.

Evaluate the Vendor

Consider the track record of the vendor in terms of customer satisfaction, reliability, and support. One source of information can be found at the American Customer Satisfaction Index (http://theacsi.org). Some computer magazines conduct reliability polls, which can provide an additional source of information. Finally, I would encourage you to call a given vendor's customer support *prior* to making a purchase. Some vendors force you to wait an unreasonable amount of time to talk to technical support staff, and you will get a good sense of the vendor's level of support by making a pre-sales technical inquiry.

Select a System

Once you have taken the previous steps you will have a clear picture of your needs in terms of a music-making computer. In most cases, it is not necessary to purchase a top-of-the-line system for music production. Owing to the rapid rate of change in the technology sphere, computers do not retain their value, so in most cases it makes better financial sense to spend your money on a mid-line system.

You may even want to consider the purchase of a used computer since a system with an appropriate amount of power can be found on sites like Craigslist for a nominal cost. Some programs like Ableton Live LE have very modest system requirements, and you can purchase a compatible used system for very little money. An upgrade of RAM and a new hard drive may be all that is necessary, but be sure that the seller provides an installation disk for the operating system since the OS is an expensive item to purchase. Be particularly cautious about purchasing a used computer that supports RDRAM (e.g., PC 800 or PC 1066), as the memory is very expensive.

DAW Software

Most of the readers of this book will want to purchase multitrack Digital Audio Workstation (DAW) software. DAW software provides the capability to record and edit multiple tracks of digital audio, apply a variety of processors and effects, and mix the tracks to a two-track master file.

TABLE 2.1 Primary DAW vendors

Vendor	Product	OS	Web Site
Ableton	Live	Mac & PC	www.ableton.com
Apple	Logic	Mac	www.apple.com
Cakewalk	Sonar	PC	www.cakewalk.com
Digidesign	Pro Tools	Mac & PC	www.digidesign.com
Image-Line	FL Studio	PC	http:/flstudio.image-line.com
Mackie	Tracktion	PC & Mac	www.mackie.com
Mark of the Unicorn	Digital Performer	Mac	www.motu.com
Propellerhead	Record	Mac & PC	www.propellerheads.se
Sony	Acid	PC	www.sonycreativesoftware.com
Steinberg	Cubase	Mac & PC	www.steinberg.net

As with the selection of a computer, there are many variables involved with the selection of a DAW. Furthermore, the workflow, or "feel," of the system can have a direct impact on your productivity so it is important to consider more than the basic feature set. With that said, I have used DAWs from most of the primary vendors and can tell you that the current programs offer a wonderful amount of power and flexibility; in a sense, it's hard to make a bad choice. Table 2.1 can be used as a starting point in your research and, while not comprehensive, the primary vendors are listed.

Download Demos

Many DAW vendors provide time-limited demos of their products. When possible, always spend time with a product demo as this will yield insights into each of the following categories.

Quality and Variety of Effects and Processors

While most modern DAWs should be equally capable of recording tracks to disk, the variety and quality of effects and signal processors will vary from vendor to vendor. For example, some vendors include convolution reverbs and guitar amplifier simulators. Other vendors might include dynamic processors that emulate classic analog gear. Of course, needs will vary from user to user, but it makes sense that an electric guitarist would likely want to utilize amplifier simulations while an acoustic musician might be more interested in the type and variety of reverbs.

I would point out that this is the one category in which a listener will be aware of your choice. While there is no way for a listener to tell which DAW was used for a given production, effects and processing *do* impact the final product in an audible way and, thus, are one of the most important considerations when evaluating the features of a DAW.

Workflow

The intangible "feel" of a DAW can be an important consideration and some programs seem to do a better job than others at supporting a given work style. In my case, the two primary workflow considerations are extensible key commands and screen sets. I like to assign key commands to my most-used functions and be able to easily organize sets of windows that help me with various editing tasks. Although most vendors provide at least some customization of key commands, the implementation in Logic Pro is very deep, which is one of the reasons I enjoy using that program. However, other users might be more concerned with the implementation of track *comping* (the ability to easily make a composite track of multiple takes) or the integration of video files for film scoring.

Support for Loops

For some musicians, the integration of a loop browser will be important. Although loops can be imported into all of the major DAW programs, an integrated browser can help you to work more spontaneously if you routinely work with drum loops and other types of pre-recorded audio clips.

Support for Software Synthesis

While all of the major DAWs support the integration of third-party virtual synthesizers, the type and number of built-in synthesizers vary greatly among vendors. Software synthesizers from third-party vendors range from $100 to $400 (or more), so for some users it may make sense to spend more money upfront to get a larger palette of software synthesizers.

Collaboration

Consider the possibility that you might collaborate with other musicians. While it is generally very easy to transfer raw digital audio files between various DAW programs—it's simply a matter of exporting the tracks to an external drive and importing the tracks in another DAW—this procedure is less than ideal if you want to collaborate on a mix or composition because, in order to hear effects or virtual synthesizers, tracks must be rendered to disk with processing intact. This makes it impossible for a colleague to undo an effect or change a synthesizer sound or parameter. For this reason, it is usually best to purchase the same software a colleague has if you intend to do extensive collaboration in the realm of mixing, editing, or composition.

Purchasing Software on a Budget

Many vendors provide upgrade paths that enable users to purchase an entry-level product and upgrade to an expanded feature set when skills and budget warrant. Though only some of the products listed in table 2.2 offer an upgrade path, each of these applications costs less than $200 at the time of this writing.

TABLE 2.2 Budget commercial software

Product	Price
Ableton Live LE	$99
Acid Music Studio	$70
Apple Garage Band	$80 (part of iLife suite)
Apple Logic Express	$200
Cakewalk Sonar Home Studio	$100
Cubase Essential	$150
Mackie Tracktion 3 Project	$50
Magix Samplitude Music Studio	$80

TABLE 2.3 Other inexpensive, free, or shareware resources

Software	Description	Approximate Price
Amadeus Pro	Multitrack audio editor for OS X	$40
Audacity	Audio editor (Linux, Mac, and PC)	Free
Ardour	Multichannel recording for OS X and Linux	$60 suggested price
KRISTAL audio	Multichannel recording for Windows	Free personal and educational use
Mixcraft 4	Multichannel recording for Windows	$65
Reaper	Multichannel recording for OS X and Windows	$60 discounted, $225 commercial
Fission (Rogue Amoeba)	Audio editor for OS X	$32
Rosegarden	Multichannel recording for Linux	Free
Sound Studio	Audio editor for OS X	$80

Note that special academic pricing is available for many DAW products from vendors such as Academic Superstore, so be sure to research academic pricing if you are a student or teacher.

Other Resources

There are a number of applications that are inexpensive or even free. For example, Reaper is an inexpensive yet capable multichannel audio application that appears to have a strong following (see table 2.3). The development of these and other applications tends to be very fluid so you will want to spend some time online to learn more about these applications and similar programs.

Fine-tuning a Computer for Audio Recording

Computers are capable of working for long stretches without crashes, but problems are more likely if you don't run a tight ship. The following tips come from personal experience, as well as the installation documentation for Pro Tools, Logic, Digital Performer, and other industry-standard DAW applications.

Close Unnecessary Applications

In general, you will want to keep your recording computer free from any unnecessary applications and utilities. Each running program or utility will compete with your DAW for systems resources, including RAM, CPU processing, or disk access, and can create a potential for system conflicts and crashes. For this reason, many audio professionals keep a clean system devoted solely to audio production and use a secondary computer for e-mail, business programs, and other applications. Another option is to keep a streamlined version of the operating system on a secondary boot drive that is devoted solely to audio recording.

Disable Utilities

Avoid running any type of disk utility while recording. It's also a good idea to disable utilities such as instant messaging since they use additional resources and can interfere with a recording application.

Use a Secondary Disk for Audio Recording

One way to maximize disk access for your DAW is to dedicate one hard drive to the operating system and music applications and a secondary hard drive (either internal or external) for recording.

Disable Power Management

When recording, you obviously want to avoid a situation in which the computer tries to spin down a drive or blank the computer screen. I even avoid the use of screen savers as they can put a momentary strain on the computer during a critical recording operation.

Defragment Drives as Necessary

Over time, your disk may become fragmented. Fragmentation occurs when many files are stored and read from disk and files are broken up into noncontiguous chunks of data. Disk access slows when drives are fragmented so it may be necessary to periodically defragment your computer using a defragmentation utility. Note that Apple generally does not advise the use of third-party defragmentation utilities. They make a convincing argument against defragmentation in article titled "HT1375," which can be read at www.apple.com.

Disable Automatic System Updates

While updates can sometimes be beneficial, don't allow the computer to disrupt a recording session with an automatic update. Also, some software such as Pro Tools requires a specific version of the operating system to run and an automatic update may cause your software to become buggy or inoperable.

Turn Off Unnecessary USB Devices

Device drivers can be a source of crashes and they also compete for system resources so it is a good idea to unplug scanners and other USB devices when not in use.

Disable Time Machine (Mac)

While backups are very important, it's inadvisable to run a disk-intensive program like Time Machine while recording to disk.

Enable Journaling (Mac)

Journaling provides a sort of safety valve that can help to prevent data loss in the event of a power outage or other problem. The following information comes from article "HT2355" at support.apple.com:

> Journaling is a feature that helps protect the file system against power outages or hardware component failures, reducing the need for repairs. . . . When you enable journaling on a disk, a continuous record of changes to files on the disk is maintained in the journal. If your computer stops because of a power failure or some other issue, the journal is used to restore the disk to a known-good state when the server restarts.[2]

Adjust the Size of the Audio Buffer

Your DAW software will provide the ability to change the size of the audio buffer. (An audio buffer acts like a temporary storage space to keep data flowing smoothly in and out of the interface.) If the size of the buffer is too small, clicks and pops can occur as the computer may not be able to keep up with audio input and output. If the buffer is too big, unacceptable *latency* (the amount of time it takes an application to process and output a given input signal) may occur. Finding the sweet spot in terms of the size of the audio buffer may take some experimentation. In general, use a larger buffer for larger track counts and a smaller buffer for real-time activities such as virtual synthesis.

Installing a Clean Operating System

Over time, your operating system may slow down. This can be the result of disk fragmentation, an overflowing hard drive, extraneous system utilities, spyware or virus, or any number of other factors. If you experience these problems or other system anomalies it may be time to perform a clean install. Follow these steps to perform a clean install:

- Backup all of your data.
- Unregister programs that require a software key.

- Make a note of important settings such as your network settings, settings for audio interfaces, etc.
- Note important information such as software serial numbers and authorization keys.
- Save special project templates, custom key command sets, and other data to another disk.
- Remove hard drive and replace with a new blank drive.*
- Format the new drive (this is usually handled by the OS installer).
- Reinstall the operating system and applications.
- Copy necessary data from the old drive to the new drive. (This is usually accomplished by purchasing an inexpensive external hard drive enclosure.)

*Although it might seem extreme to completely replace a drive, hard drives are relatively inexpensive and I have found that swapping an old drive with a new one provides the peace of mind that I won't inadvertently erase important data.

Case Study: Computing on an Extreme Budget

I thought it would be interesting to see how an older computer would fare in a modern project studio. I happen to own a Dell Dimension 8250 which I used for the test. The computer has a reasonable amount of processing power with a 2.4 GHz processor, but memory and other features are limited and a computer with similar specifications could be purchased on craigslist for less than $100. Unfortunately, the computer uses RDRAM so it was impractical to upgrade RAM beyond the 768 MB that was already installed in the computer.

A first step was to use the original equipment manufacturer (OEM) disk to format the hard drive and install a fresh copy of Windows XP. I used another computer to download the latest service pack from Microsoft and used a flash drive to install the update on the Dell. The next step involved installing a driver for a Focusrite Saffire PRO 24 audio interface and installing a copy Cockos's Reaper 3 Digital Audio Workstation software.

I used the computer for several hours of recording and was very satisfied with the performance. Reaper felt responsive and the system was stable throughout the recording process. I connected an external interface to the Focusrite via ADAT optical and recorded ten simultaneous tracks without a problem. Reaper showed that 65 percent of the CPU's processing power was utilized to play back ten tracks when an effect was inserted into each channel.

Although I would recommend a more powerful computer for projects involving virtual synthesizers or large numbers of audio tracks and effects, the test demonstrated that it is possible to record and mix using a modest computer. The test also highlighted that preamps and microphones are more important than raw computing power when it comes to making great recordings.

Other Considerations

As you can see, there are many things to consider in the selection and use of a computer for digital audio recording. We will conclude the chapter with three items that are essential for all computer users.

Backups

Your most important day-to-day computer maintenance task should be to make regular backups of your work. Ideally, an extra copy of your data should be stored off-site to protect against a catastrophic event such as a fire, flood, or other calamity. Enclosure docking stations (see figure 2.6) can be very useful for backups. These products provide a quick method for connecting a stock 2.5-inch or 3.5-inch SATA or IDE drive to your computer, thereby saving the cost of a traditional external hard drive. For example, a 500 GB external hard drive will likely cost around $75–100 while a similar size internal hard drive (which can be connected to a dock) may cost as little as $50.

FIGURE 2.6

Thermaltake BlacX External Hard Drive Enclosure

Ventilation

Always ensure your computer has proper ventilation. Over time, dust will find its way inside the computer so you may want to periodically (and gently) vacuum the heat ducts. Also check that drapes, cables, or other items do not impede the flow of air around the computer.

UPS

An uninterruptable power supply can keep you from losing data in the event of a loss of power. Keep in mind, though, that the computer may crash if power is removed from peripherals. For this reason you should run your computer, monitor, hard drives, and other connected equipment such as an audio interface from the UPS. Speakers and other noncritical hardware can be connected to another circuit. This should enable you to save work that is in progress in the event of a power outage.

3

Introduction to Acoustics

Acoustics can be defined as "the science of the production, propagation, and perception of sound."[1] Acoustics is based on the larger discipline of physics, but it also involves psychoacoustics—a branch of psychology concerned with the perception of sound. For these reasons, the study of acoustics involves an interesting blend of the objective and the subjective. While acoustics is obviously a deep subject that cannot be covered with any detail in a single chapter, my intent is to focus on the primary concepts and terms that will help you record better music. The chapter will also provide a foundation for further study.

An understanding of acoustical concepts can aid in many facets of music production, such as the selection and use of microphones or placement of reference monitors. This knowledge can help you to visualize how to utilize baffles to provide acoustic separation between instruments and to avoid the problems that occur when multiple microphones are simultaneously used in a recording space. These concepts will also help you to evaluate the sonic properties of a room for recording or mixing and will provide a starting point in terms of the many "gotchas" that may occur in a typical home recording environment.

Knowledge of acoustics is also helpful to the recording musician. An interesting example of the relationships among performer, engineer, and acoustics comes from a book by pianist Charles Rosen:

> Finding the balance between too much resonance and too little is not only difficult, but obstructed by the aesthetic taste of some

> sound engineers. The first record I made for CBS was Ravel's *Gaspard de la Nuit*, and the sound engineer started by placing one of the microphones so close as to be almost inside the piano. The opening piece, *Ondine*, begins with a soft irregular *tremolo*, representing the shimmering of light on water. Placing a microphone very close to the instrument emphasizes the initial percussive impact of each note as it is struck and removes the liquid blending together of the total sonority that was Ravel's clear intention. With the microphone so close to the strings at the upper part of the piano, the sound was considerably more brittle than it was sitting at the keyboard. When I said I thought that the microphone was too close, the sound engineer protested that if it were moved farther away, we would lose fidelity. . . . Reluctantly the sound engineer pretended to move the microphone away, and was at last persuaded to make the distance perceptible to eye and ear (an inch can make an extraordinary difference).[2]

Of course, the most obvious application of acoustics is in the design and treatment of recording rooms and control rooms, and it is in this application that I would stress the necessity of further research. While various acoustic treatments will be presented that can be useful in fixing a variety of sonic issues, acoustical treatment is a complex subject and the solutions are often expensive, so the reader is advised to consult the many resources listed in the bibliography. I would also stress that it is advisable to hire an acoustician prior to any construction project, renovation, or purchase and application of any substantial acoustical treatment.

Another reason a discussion of acoustics is appropriate to a book devoted to budget recording involves the purchase of equipment. Too often musicians purchase new equipment in the hopes of making better recordings. In many instances, new microphones, preamplifier, or reference monitors are not the answer—acoustical phenomena may be causing your recordings to sound dull, woody, or lacking in warmth, so the treatment of acoustic issues in a recording or engineering space is often money well spent.

Sound Waves

What musicians generally refer to as *sound waves* are variations in atmospheric pressure. In order for vibrations to be transmitted from one point to another, there must be a medium—some material that provides a connection between the source and the destination. It is also necessary that the medium be elastic and normally in a state of equilibrium. When a disturbance is created, excited molecules propagate the disturbance through the medium and the rate of propagation is determined by the density of the medium, as well as its elasticity.

When you clap your hands, a disturbance or *impulse* is created in the medium of air. The excited air molecules move slightly and create a momentary region of high pressure called a *compression* or *condensation*. Since the medium is elastic, the molecules create an alternating region of low pressure called a *rarefaction* before returning to a state of equilibrium.

In order to visualize the concepts of compression, rarefaction, and propagation, consider a large bowl of chilled Jell-O. A disturbance is created if you gently press your finger down on the top of the Jell-O. When you release your finger, the Jell-O returns to a state of equilibrium but, for that to happen, the disturbance must be propagated through the rather dense and elastic medium of the Jell-O. Clearly, the individual Jell-O molecules don't radiate to the edge of the bowl and back. You can test this by placing a drop of food coloring on the surface of the Jell-O near the middle of the bowl. The food coloring does not move toward the edge of the bowl when a disturbance is created. Instead, the dot moves slightly owing to the disturbance but returns to its original position at the point of equilibrium. For this reason, it's easy to see that the disturbance or *impulse* is transferred from one molecule to another—the molecules move slightly when they are excited but they are not permanently displaced.

In his book *Measured Tones: The Interplay of Physics and Music*, Ian Johnston uses the example of a Newton's Cradle to describe the concept of wave propagation. As a ball is dropped on one end, energy is transferred from one ball to the next, eventually causing the final ball to move. Energy is transferred from the first ball to the last, but the intermediate balls do not move.

With regard to a medium, it is helpful to note that the medium may be one-, two-, or three-dimensional. A guitar string is, essentially, a one-dimensional medium while the head of a drum is two-dimensional. A key concept to remember is that the medium of air is three-dimensional. While this might seem like an obvious statement, it is easy to forget that sound waves propagate in three dimensions. This concept will be useful when you consider the placement of microphones, speakers, and acoustical treatments in a studio environment.

Waveforms

A waveform is a visual representation of a sound's shape over time, and a waveform view is the primary means with which recording engineers manipulate sound in the digital domain (see figure 3.1). There are several waveform characteristics that are of interest, as detailed in the following paragraphs.

Amplitude

In figure 3.2, a sinusoidal curve represents fluctuations of air pressure on the Y-axis and passage of time on the X-axis. This simple waveform represents two essential wave qualities: *amplitude* and *frequency*.

The change in sound pressure reflected on the Y-axis is termed amplitude. Higher values reflect greater intensity, which is analogous to loudness. How-

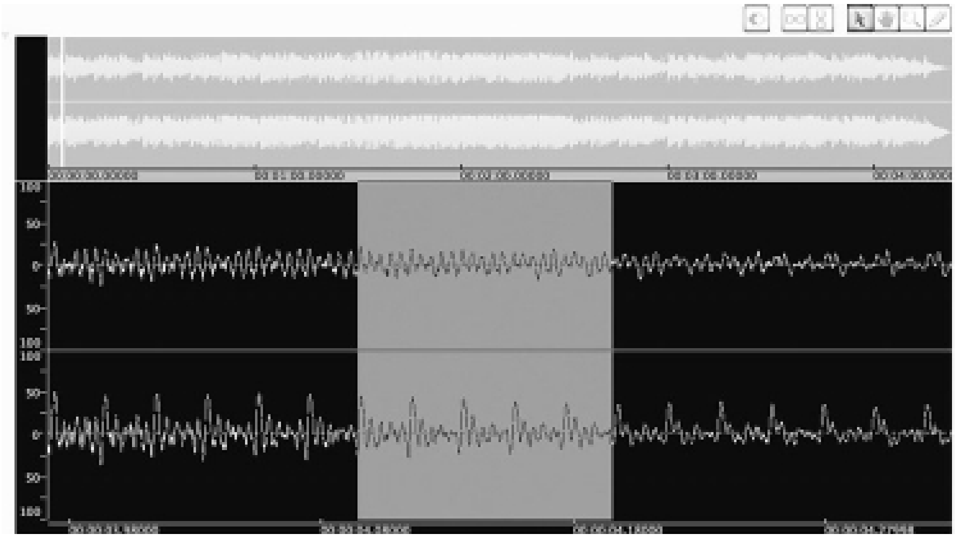


FIGURE 3.1
Digitally editing a waveform

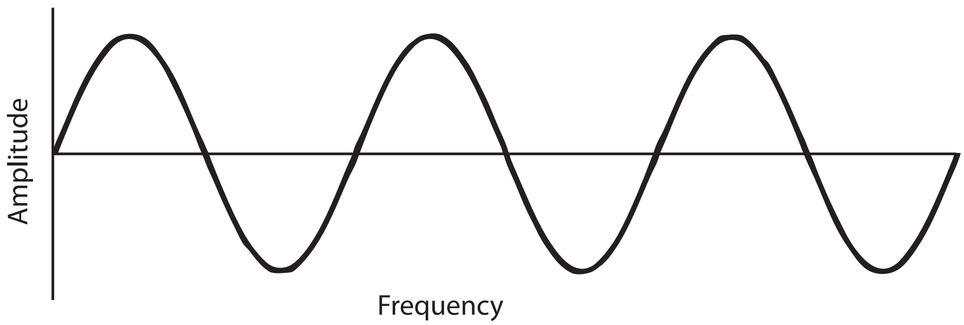


FIGURE 3.2
Sinusoidal waveform

ever, it is interesting to note that our perception of loudness is somewhat dependent on frequency: middle frequencies will sound louder than high or low tones given a signal of equal amplitude.[3]

Frequency

Frequency is represented on the X-axis. This is analogous to the strings of a guitar: thicker strings will oscillate more slowly, producing a lower pitch, while thinner strings produce a higher pitch. Similarly, a guitar string that is plucked vigorously will create oscillations of higher amplitude (but the same frequency) than the same string plucked with a gentle touch.

Frequency is measured in *hertz* (Hz), a figure that represents the number of oscillations per second. A related term, *kilohertz* (kHz) indicates a thousand oscillations per second. The generally accepted range of human hearing is 20 Hz to 20 kHz, but it should be noted that few adults can hear as high as 20 kHz. As an interesting aside, some students use this knowledge to advantage by creating cell phone ring tones that ring at very high frequencies that most teachers are unable to detect.

Frequency relates to our perception of pitch. For example, the note A (880 Hz) is an octave higher than the common tuning note A (440 Hz) since its frequency is twice that of the lower note. Interestingly, amplitude can also have an effect on our perception of pitch. In his excellent book *Master Handbook of Acoustics,* F. Alton Everest describes how, given a 100 Hz tone, the ear will perceive a different pitch depending on the amplitude of the tone. You can verify the experiment with your recording software by playing a test tone at 100 Hz and varying the output level. You will hear a subtle change in pitch. As Everest points out, "We cannot equate frequency and pitch, but they are analogous."[4]

Phase

You might think of the *phase* of a waveform as its time offset. The idea is that a waveform of a given frequency and amplitude can start at any arbitrary point in time. Phase is an important concept in audio recording since the relationship of the phase of two waveforms can be constructive or destructive. For example, in figure 3.3, two sine waves of the same frequency and amplitude have the same phase—the oscillations start at the same time. It's easy to see that when the values of the two waves are combined, a twofold increase of amplitude occurs.

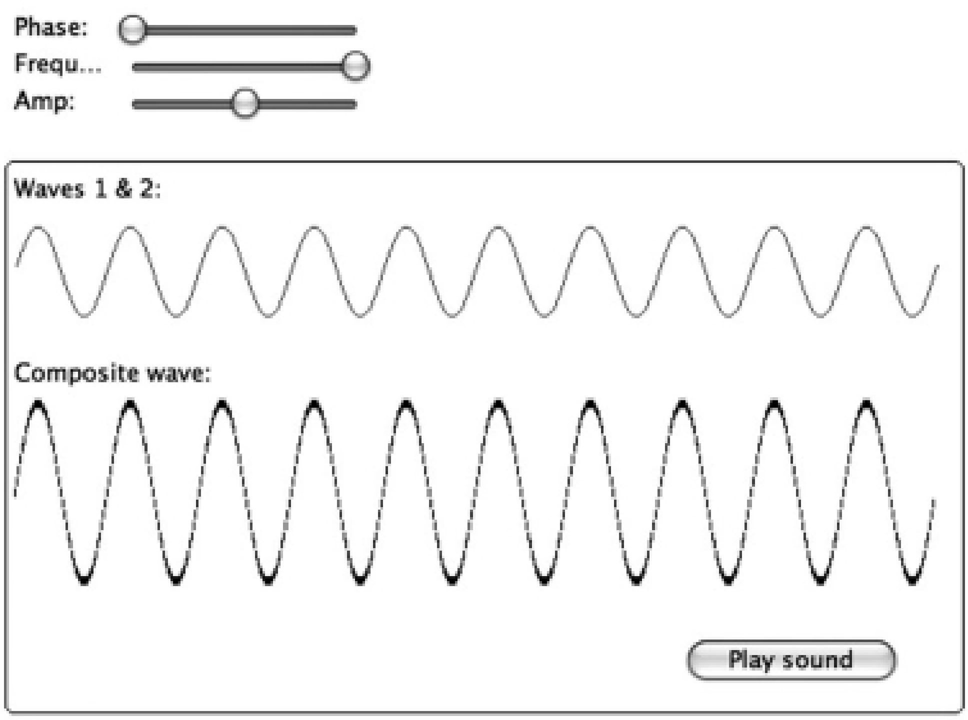


FIGURE 3.3
Sine waves in phase

Notice how sine waves cancel each other out when they are 180 degrees out of phase, as in figure 3.4.

Although the previous example is simplistic, it represents an important acoustic phenomenon of which recording engineers must be keenly aware: signals that are out of phase can interact in a destructive way and may cause a recording to sound weak or "phasey." The previous screenshot was taken from a simple Java applet that I developed to explore the concepts of frequency, amplitude, and phase. I would encourage you to experiment with the applet, which is available at the companion Web site, in order to develop a more intuitive understanding of waveforms.

EXAMPLE 3.1

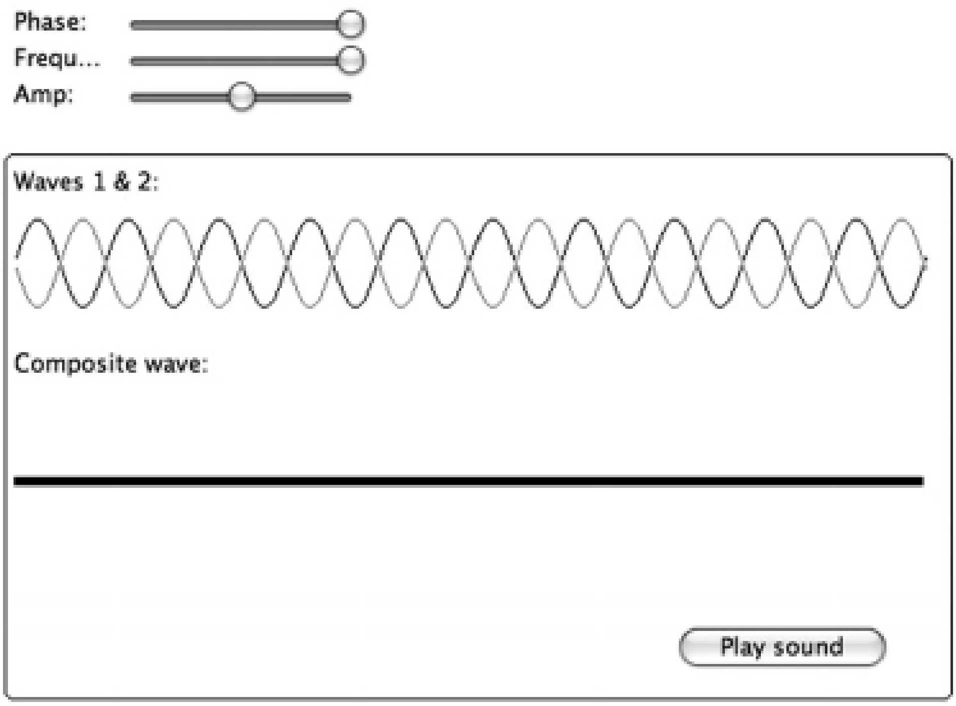


FIGURE 3.4

Sine waves out of phase

Velocity

The velocity or speed of a waveform is dependent on the medium—air molecules in the case of sound waves. At room temperature, the speed of sound in air is about 1,130 feet per second, and it is interesting to note that changes in temperature affect the velocity of a sound wave: the speed of sound changes 1.1 feet per second for a change of 1 degree Fahrenheit.[5] The speed of a sound wave is related to its wavelength and frequency as follows:

$$\text{speed} = \text{wavelength} \times \text{frequency}$$

Wavelength

The calculation of *wavelength* can be useful in determining modal resonances that occur in an enclosed space. Where frequency refers to the number of oscillations in a given unit of time (one second), wavelength refers to the distance a wave travels in a variable amount of time. In this case, the length of time is determined by the time it takes to complete a cycle. Given the previous formula for velocity, it is easy to see the relationship among wavelength, frequency, and velocity:

$$\text{wavelength} = \frac{\text{speed of sound}}{\text{frequency}}$$

By way of example, consider a sound wave at 440 Hz. To calculate its wavelength, divide the speed of sound (1,130 ft/sec) by the frequency (440 Hz). Hence, the wavelength of A-440 is 2.57 feet while the wavelength of its higher octave is 1.28 feet (see figure 3.5).

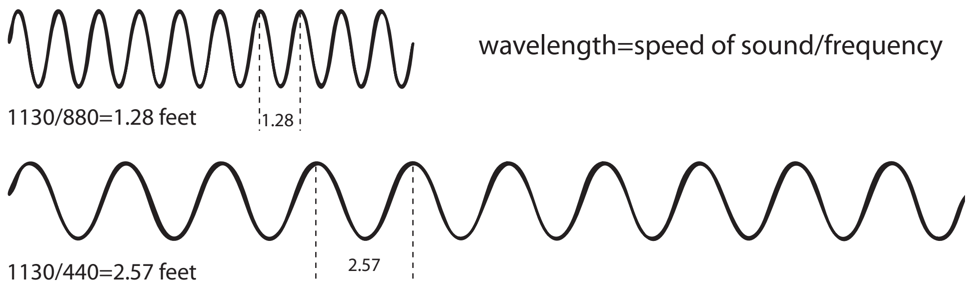


FIGURE 3.5
Calculating wavelength

Another application of the concept of wavelength has to do with sound *diffusion*. As you will see in the pages that follow, it is often desirable to diffuse or scatter sound waves in a small room. Given that the wavelengths of low frequencies are so large—the wavelength of the lowest note on the piano, A0 (27.5 Hz) is 41 feet—it's easy to see why it is so difficult to diffuse low frequencies. The wavelengths are simply too big to be affected by the small irregularities that are used to scatter higher frequencies.

Sound Waves in the Real World

Music would not be very interesting to listen to if it consisted solely of pure sine waves. In the real world, waveforms are much more complex and the complexity is what provides the pleasurable sensation of musical color or timbre. One way to visualize this concept is to consider our earlier discussion of frequency, amplitude, and phase. If two sine waves of differing frequency and amplitude are combined, a more complex wave is formed that might be said to have characteristics of both (see figure 3.6). Very complex waveforms can be created in

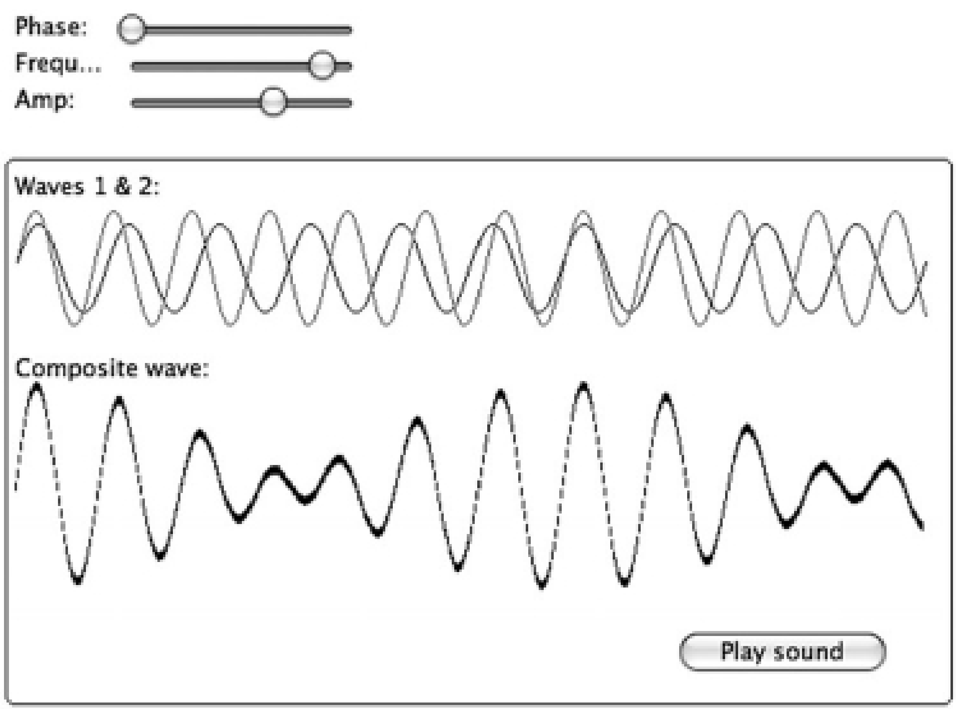


FIGURE 3.6
Combining two contrasting sine waves

this way by combining multiple sine waves of varying amplitude, frequency, and phase. This is the essence of additive synthesis, and you can explore the concept with the Java phase applet at the companion Web site.

EXAMPLE 3.2

Overtones

When you pluck a string on a guitar or other string instrument, the string vibrates with a primary frequency that we associate with a particular pitch. The string also vibrates in halves, and this secondary vibration produces another tone, called the first overtone or second partial, and its frequency is twice that of the fundamental. The string will also vibrate in thirds to produce the second overtone (third partial), and the process continues with other integral multiples to form the overtone series (see figure 3.7). The amplitude of overtones tends to decrease through the overtone series so we perceive of these partials as tone color or timbre, not as specific pitches.[6]

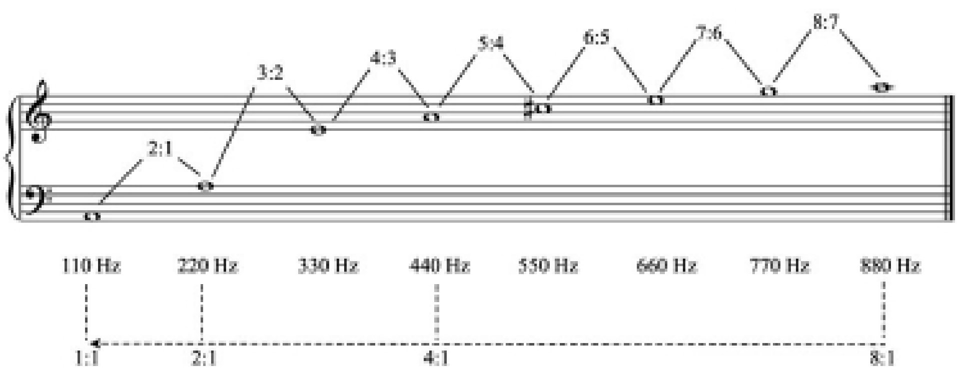


FIGURE 3.7

Overtone series

Harmonics

For purposes of our discussion, you may think of overtones and harmonics as largely synonymous. The primary difference is that an overtone is not necessarily an integral multiple of the fundamental. Musicians often refer to overtones as simultaneous "tones" that are a component of a fundamental tone while scientists use the term *harmonic* to refer to a component wave of a complex waveform. Harmonics have an integral relationship with the fundamental: the second harmonic is twice the frequency of the fundamental and the third harmonic is three times the frequency and so on. The variation in amplitude of harmonics is what determines the tone or timbre of a sound. Depending on the sound source, some harmonics might be absent while others are emphasized.

Harmonics are of particular interest in recording. Not only are harmonics responsible for the color or timbre of a sound, they contain acoustic energy that can affect our perception of loudness. For example, using EQ to emphasize a specific range of frequencies may cause a musical track to sound louder. Conversely, attenuation of a range of frequencies can cause a signal to sound softer.

Nonharmonics

The sound of some instruments, such as bells and piano, may contain partials that are *nonharmonic*. However, it is interesting to point out that *periodic* waveforms—those waveforms that repeat and might be described as "steady"—will always have harmonic overtones. As Ian Johnston states: "If a note can be kept perfectly steady (by blowing or bowing) so that it is genuinely periodic, then the Fourier theorem says that its overtones must be harmonic (whole number multiples of the fundamental). It cannot be otherwise."[7]

Spectrum

The term *spectrum* can be defined as "the distribution of energy across the range of audio frequencies at any given moment."[8] With regard to the discussion of harmonics, we are concerned with the audible range of frequencies of around 20 Hz to 20 kHz. This brings up an interesting point: If the range of hearing is between 20 Hz and 20 kHz, why is the sample rate for a digital audio CD 44.1 kHz? The reason for this is that, in order to accurately sample a signal, the sample rate must be "at least twice the frequency of the highest frequency component."[9] This makes sense when you consider that a sample must be taken of both the positive and the negative portion of an oscillation so, minimally, at least two samples are required during each cycle. However, since an audible tone may include inaudible frequencies above our frequency threshold, it is necessary that the sample rate be somewhat higher so that inaudible partials are not quantized or rounded down to produce harmonic distortion.

Earlier in the book we touched on the concept of frequency response, and it bears repeating here. Microphones and other devices that do not have a flat frequency response color the essential harmonic essence of a given sound source. While it is oftentimes desirable to select a microphone to emphasize a certain range of frequencies, if a microphone or other device does not capture harmonic content, the harmonic color or timbre is essentially lost during the recording process. This is one of the reasons engineers generally advise against committing equalization to disk when recording.

A bass woofer may be necessary to accurately monitor the low end of the frequency spectrum when recording or mixing some types of music. This is particularly true for some genres of popular music featuring sub-bass patches that approach the lower threshold of human hearing. While extremely low tones may not be heard as distinct pitches, they are often felt.

Loudness Levels

Acoustic energy spreads out in ordinary space so it is necessary to consider both the amplitude of a wave and the area in which the waveform is traveling. This number, known as *wave intensity*, is calculated in watts/square meter. It is stunning to consider our sensitivity to sound-wave intensity. As Doug Coulter notes in his programming text *Digital Audio Processing*, "an eardrum vibration 1/10th size of a hydrogen molecule at 1 kHz can be heard!"[10] Along these lines, the human ear is capable of responding to intensity of approximately 10^{-12} watts/meter^2.

Given our astounding sensitivity to loudness, a logarithmic scale was devised to provide a more natural way of referring to sound levels. Without such a scale, engineers might have to say something like "please pull down 2 kHz about 5,125,000."

A quick word about logarithms is in order. A logarithm provides another way of looking at exponents. For example, it is easy to see that 10^3 is equal to 1000 ($10 \times 10 \times 10 = 1000$). We could look at this relationship another way and say that $3 = \log(1000)$. Hence, the *log* of a number is the exponent that, when placed with 10, will yield the given number. In this context, the math is less important than realizing that, on a logarithmic scale, values increase exponentially, not linearly. This type of system is better suited to the way our perceptual organs work. The *decibel* (dB) provides a useful way to measure levels of loudness and its range is based on our threshold of hearing (barely audible) to a pain threshold of 1.0 W/m^2. Sound pressure levels can be calculated as:

$$\text{loudness (in dB)} = 120 + 10 \times \log(\text{intensity})$$

Dynamic markings can provide an intuitive way for musicians to visualize decibels. For example, where a *mezzo forte* represents a loudness of approximately 70 dB (.00001 W/m^2), a *forte* would equal approximately 80 dB (.0001 W/m^2) and a *fortissimo* would equal approximately 90 dB (.001 W/m^2).

Sound Wave Interactions

Of primary concern to a recording musician is the way a sound source interacts acoustically with the recording environment or performance hall. Ideally, the harmonic energy of a sound source will be evenly distributed so that no range of frequencies is emphasized or attenuated. In other words, it is desirable that the room have a flat frequency response and that the sound is evenly dispersed. Unfortunately, a variety of acoustical phenomena will come into play that will affect the distribution of harmonic energy.

Standing Waves

Standing waves are caused when sound waves interact after reflecting off of two parallel surfaces. You can visualize this interaction by thinking about what happens when a disruption occurs in a medium. Consider a bassist who bows a long note in a small room: the disturbance causes sound-pressure waves to emanate in all directions. When the waves reach a reflective surface, they reflect off the surface at the *angle of reflectance*, which happens to be the same angle as the initial *angle of incidence* (see figure 3.8).

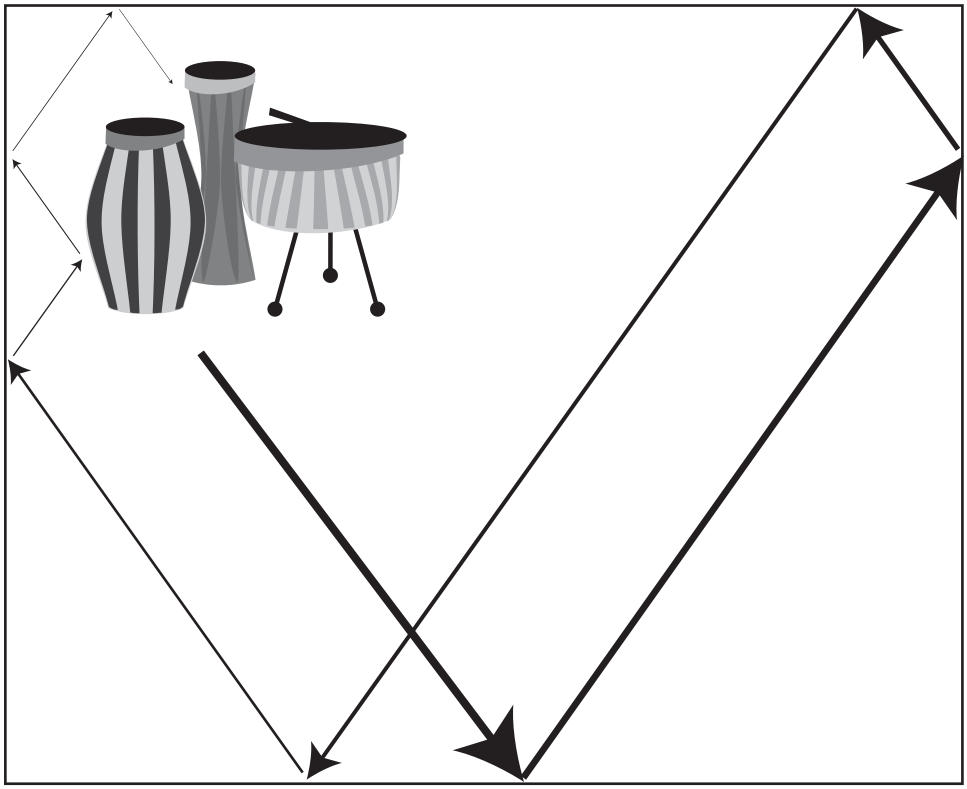


FIGURE 3.8
Angle of incidence and reflection

Now consider what happens when a sound-pressure wave reflects off of two parallel surfaces: those waves that are reflected perpendicularly off of each wall will interact when they return along their original path. This causes a phenomenon known as a *standing wave.* You can visualize this phenomenon by using a common child's toy, the Slinky. Lay the Slinky flat on the floor and ask a friend to hold one end at a fixed position. Move the other end to create a disturbance and you will see a wave travel down the Slinky in much the same way that a sound wave propagates through the air. Next, repeatedly move the free end in and out and adjust the frequency until the waves bounce off of each end and meet in the center at the same time. Providing you find the correct frequency, you will see the formation of a standing wave in the middle of the Slinky. A simple Java animation is available at the companion Web site that will be useful in providing a visualization of this phenomenon.

EXAMPLE 3.3

Sound-pressure waves may also be diffracted by objects in a recording space. *Diffraction* is easy to visualize if you consider a partially submerged rock in the middle of a flowing stream. The rock produces a disturbance in the water that alters the flow of surface waves. In a similar way, an obstruction in an auditorium can interact with sound-pressure waves causing the formation of acoustic shadows.

Room Modes

The term *room mode* is often used to describe the specific resonances created by standing waves. The problem with such modes is that they alter the frequency response of the room. A given frequency may be emphasized in one part of the room and reduced in another. Not only that, the fundamental frequency and its harmonics will be affected. For example, if the resonant frequency or mode is equal to 55 Hz, each of the harmonics (110, 165, 220, 275, etc.) may also be problematic. You can use the following formula to calculate room resonance:

$$\text{resonant frequency} = \frac{\text{room mode} \times 565}{\text{room dimension in feet}}$$

The variable called *room mode* refers to the fundamental frequency (1) or its harmonic (2, 3, 4, etc.). Although 565 might seem like an arbitrary term, it is half the speed of sound at room temperature. Using this formula, it's easy to see that a room that is ten feet in length will have modal resonances at 56.5 Hz, 113 Hz, 226 Hz, and so on.

Now, consider a room with a depth of ten feet *and* a width of ten feet: modal resonance will occur *at the same frequency* in each dimension. The problem will be even worse with room that is a ten-foot cube since the modal frequencies will be emphasized in three dimensions. For this reason, acousticians advise against building rooms with dimensions that are the same or a multiple of one another. Of course, most of us do not have the luxury of constructing a recording room with ideal dimensions, but there are ways to combat modal resonance. As you will learn later in the chapter, sound absorbers and diffusers can be useful in combating these resonances.

Reverberation

In order to visualize how reverberation works, it is helpful to consider the way a sound wave travels to our ears. As the name implies, *direct sound* reaches our ears in a straight line—the shortest possible path. Direct sound helps us to identify the location of a sound source. Since sound emanates in three dimensions, a portion of the sound may be reflected off of a surface, such as a mixing console or ceiling, and reach our ears slightly later than direct sound. These secondary reflections, known as *early reflections*, merge with direct sound and provide a sense of fullness, as well as an indication of the size of the room. However, early reflections can be problematic in a control room where accuracy is the primary concern.

Reverberation occurs when multiple reflections of sound are blended together. In an enclosed and reflective space, sound energy is reflected back into the room in myriad ways, which helps provide a sense of what might be described as depth or dimension. A key consideration is reverberation time, which is dependent on the size of the room and the absorptive qualities of the surfaces. When recording in a "live" room, reverberation can, to an extent, be controlled by the selection and placement of microphones. For example, moving a microphone closer to a sound source will create a recording with more direct sound and less room sound. Similarly, artificial reverb may be added after a recording and similar control can be achieved by varying the amount of dry (unprocessed) and wet (processed) signals.

Echo

Where reverb results from the combination of multiple indiscernible reflections, echo is the result of discrete reflections. Although electronic echo effects are sometimes useful for special effects, naturally occurring echo is generally undesirable when recording or performing, as it can be disconcerting for a listener to perceive a secondary or delayed copy of a sound source. A special type of echo, called *flutter echo*, can occur in smaller rooms with parallel surfaces. It is generally advisable to treat a room with absorbers and diffusers in order to avoid flutter echo.

Controlling Sound-Pressure Waves

To this point, the primary focus of the chapter has been on the propagation and interaction of sound waves. We will now consider the control of these pressure waves in a studio. The two primary considerations are isolation of the recording environment and treatment of the surfaces.

External Isolation

A key consideration in the design and construction of any recording room is the control of sounds emanating from outside the structure, as well as those coming from within the structure. Construction methods are beyond the scope of this book, but it will be helpful to consider several key concepts in order to dispel several myths. First, absorption is different from isolation. Where absorption will attenuate certain frequencies, isolation involves the implementation of barriers to keep sound waves from entering or leaving a recording space. The use of floating walls and floors are common. In this instance, spring or rubber supports can be used to isolate a structure so that vibrations are unable to move from the surrounding structure to the room (or vice versa).

Mass and decoupling are the keys to minimizing sound transmission. For example, some project studios utilize double-wall construction to provide airspace between the walls, as well as additional mass in the form of acoustic insulation and one or two layers of Sheetrock. In some cases, Sheetrock may be hung from resilient channel to minimize the transmission of vibrations through the framing. The combination of mass and airspace is very effective in keeping sounds in (or out) of a recording space.

Surface Treatment

Ideally, acoustic energy should be evenly distributed throughout the frequency spectrum in a recording room. Reflections should be controlled to avoid standing waves and other types of interference that affect the frequency response of the room, yet the room should reflect enough energy so that the sound is not overly dry. Absorbers and diffusers are the primary tools that are used to control sound waves.

Absorption

Absorptive material is rated with a number called the *sound absorption coefficient*. This number, which is on a scale of from 0 to 1, indicates the ability of a given material to absorb sound. Higher values indicate an absorptive material and lower values indicate reflectivity. However, it is important to note that the absorptive qualities of most material vary depending on the frequency range. For example, heavy carpet on a foam pad on concrete has a sound absorption coefficient of 0.60 at 2,000 Hz but only 0.02 at 125 Hz.[11] It can be a challenge to introduce materials with absorptive qualities into a recording room as specific frequencies may be absorbed while other frequencies are unaffected, thereby changing the frequency response of the room. For this reason, it may be necessary to utilize multiple absorbers to handle sound waves at different frequency ranges.

Treble Absorption

In general, surfaces that are rough or porous and covered with small protrusions or indentations will absorb higher frequency sound waves. Waves with smaller wavelengths enter the holes and energy is dissipated as the waves bounce off of the small protrusions. However, treble absorbers have no effect on the longer wavelengths of low frequencies.

Bass Absorption

In contrast to treble absorbers, bass absorbers, or *bass traps,* may utilize a number of techniques, such as a vibrating membrane over a cavity or absorptive porous materials. By way of example, consider 1-inch fiberglass tile with a hard backing. Its sound absorption coefficient is only 0.06 at 125 Hz when mounted flush on a wall. In contrast, the *same material* has a sound absorption coefficient of 0.69 at 125 Hz when suspended. Clearly, the use of an airspace cavity is an important consideration when attempting to absorb lower frequencies. In a small room, an emphasis is usually placed on treating lower frequency standing waves, although high-frequency absorption may also be desirable. Numerous professional products are available from companies such as Auralex, Primacoustic, and RealTraps, and plans for building a simple treble absorber can be found in the final chapter of the book.

Diffusion

The goal of diffusion is "the uniform distribution of sound energy in a room so that its intensity throughout the room is approximately equal."[12] During the 1960s and 1970s, it was common for studios to utilize an extreme amount of absorption, creating what might be described as a dead or "sound sucker" room. With this approach, microphones were typically placed close to instruments and artificial reverb was added to create a sense of warmth and spaciousness. Today, it is more common to utilize a combination of absorptive and diffusive materials to create a recording space that breathes.

Ideally, acoustic treatments will be implemented in such a way that the sound space is adaptable and can be changed to better suit a particular style of music. For example, a more diffuse sound might be appropriate for acoustic music, including classical or jazz, while a somewhat drier room may be desirable for music such as rock and pop, both of which place more reliance on artificial signal processing.

Diffusers work by spreading out sound energy so that sound is less direct. Where a distinct reflection such as an echo is generally undesirable, a diffuse sound will create a sense of spaciousness or ambience. It might be helpful to think of it this way: absorption removes sound energy from a room whereas diffusion reflects energy back into the room. In most cases, a combination of diffusion and absorption will be useful in treating an acoustical space.

Control Room

A great deal of attention in books and magazines is focused on the acoustics of the control room. The reason for this emphasis is that acoustical problems in a control room will manifest themselves so that mixes created in such a room will not translate well to other listening environments. For example, modal resonances may cause an engineer to reduce or boost a particular band of frequencies, creating a mix that sounds thin or boomy on other systems. In addition to the treatments listed in the last section, consider the following concepts when setting up a room for critical listening.

Symmetry

A primary consideration is the symmetry of the mixing environment. Try to place reference monitors so that they are equidistant from side walls in order to prevent sound waves from reflecting off the walls at different times, as this can create problems with the stereo image at the listening position.

Early Reflections

Be aware that sound waves may bounce off of reflective surfaces such as a tabletop, console, or keyboard before reaching your ears. Early reflections (see figure 3.9) mix with direct sound and can cause the cancellation of frequencies or a special condition called *comb filtering* that occurs when a delayed and direct sound are combined. In this case, some frequencies will be emphasized while others are cancelled out. Absorption is typically used in places where early reflections are an issue, but careful placement of monitors can also be helpful in minimizing these reflections.

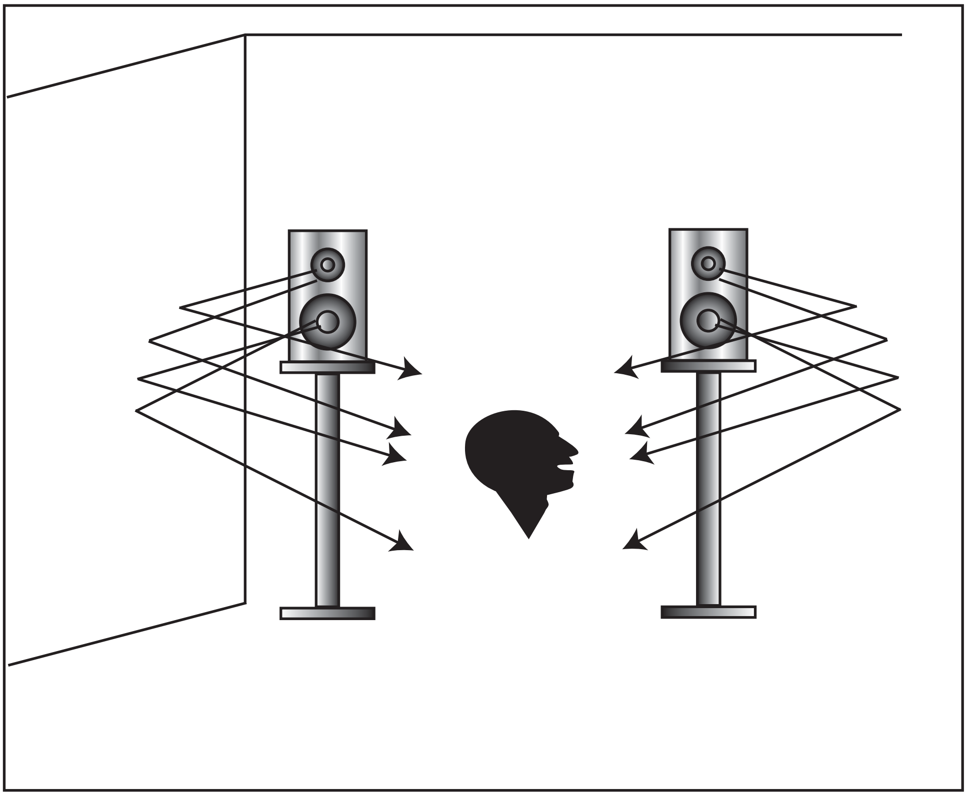


FIGURE 3.9
Early reflections in a control room

Practical Application: Building a Project Studio

In the following paragraphs I will detail the steps and thought process that went into the construction of my project studio. I would stress that this is *not* a "how to" tutorial. Your situation and goals are unique, and you will likely need to follow different steps. I simply want to illustrate the application of acoustic concepts in a real-world situation where it was necessary to strike a practical balance between goals and budget.

Although the construction of a home studio might sound expensive, raw materials such as 2 × 4s, Sheetrock, rockwool insulation, and resilient channel are reasonably priced and you can save a great deal of money by doing the work yourself. Of course, you will need to check local building codes and procure the necessary knowledge and permits, but basic framing is not hard to learn and you will likely find the process to be very satisfying.

Evaluating the Environment

Sound leakage is a key consideration for any studio. You obviously want to keep extraneous noise from entering the studio, but it is also important to contain sound so that the recording process does not disturb neighbors or your family. I was very fortunate in this regard as my basement is built against the side of a hill and the nearest neighbors are nearly a block away from the studio side of the house. The recording room was rather quiet to begin with, which allowed me to cut a few corners in order to save money on the project.

Evaluating the Room

The next step was to consider the size of the room and its intended use. I am fortunate to have a recording room of suitable size to incorporate a piano and one or two additional instruments. The room is in an L shape, and I realized that it would be convenient to divide the room into two sections. A wall was constructed between the two sections of the room and French doors were installed. This configuration affords the option of utilizing the space as two independent "iso" rooms or a single large recording room (see figure 3.10).

FIGURE 3.10

Glass wall between two sections of a recording room

Isolation

Acoustic isolation was an important consideration since I wanted to be able to practice or record without disturbing my family. I also wanted to minimize the potential for outside vibrations to enter the studio. To accomplish this, *double-wall construction* was utilized to provided airspace between the inner and outer walls. *Rockwool insulation* was installed to provide mass to the inner walls and the seams were sealed with acoustical sealant. Further, ⅝-inch Sheetrock was attached to *resilient channel*, a special type of metal spacer that is used to attach Sheetrock to the studs. The Sheetrock provided additional density and the resilient channel helped to decouple the Sheetrock from the framing. I elected not to purchase SheetBlok, a popular product that is used as a barrier layer. While the product would likely be needed in a typical studio construction, it is very expensive and I determined that the location of my recording room and my goals for the room didn't warrant the extra expense. My father, a master crafts-

TABLE 3.1 Absorption coefficients for Owens Corning 703 insulation

Freq	125 Hz	250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz	NRC
OC-703	0.11	0.28	0.68	0.90	0.93	0.96	0.70

man, designed a hanging ceiling that we filled with rockwool insulation. The resulting isolation is impressive—certainly acceptable for a project studio.

Absorption

It was easy to hear that reflections needed to be controlled so a first step was to build a number of absorbers utilizing Owens Corning 703 insulation. As you can see from table 3.1, OC 703 insulation does a particularly good job of absorbing frequencies above 1000 Hz but is less effective in absorbing lower frequencies.

I experimented with the number and placement of the absorbers until I was happy with the results. Although the room exhibited a slight amount of boxiness owing to the lack of bass absorption, it was perfectly suitable for a number of recording sessions.

Diffusion

The next step was to install diffusers. My goal was to tone down some of the boxiness of the recording space so that the room would be suitable for acoustic ensembles such as a piano trio or string quartet. I decided to purchase a box of Auralex Q'Fusor Sound Diffusors, which were used in conjunction with existing absorbers (see figure 3.11). While the installation of the Q'Fusors did not

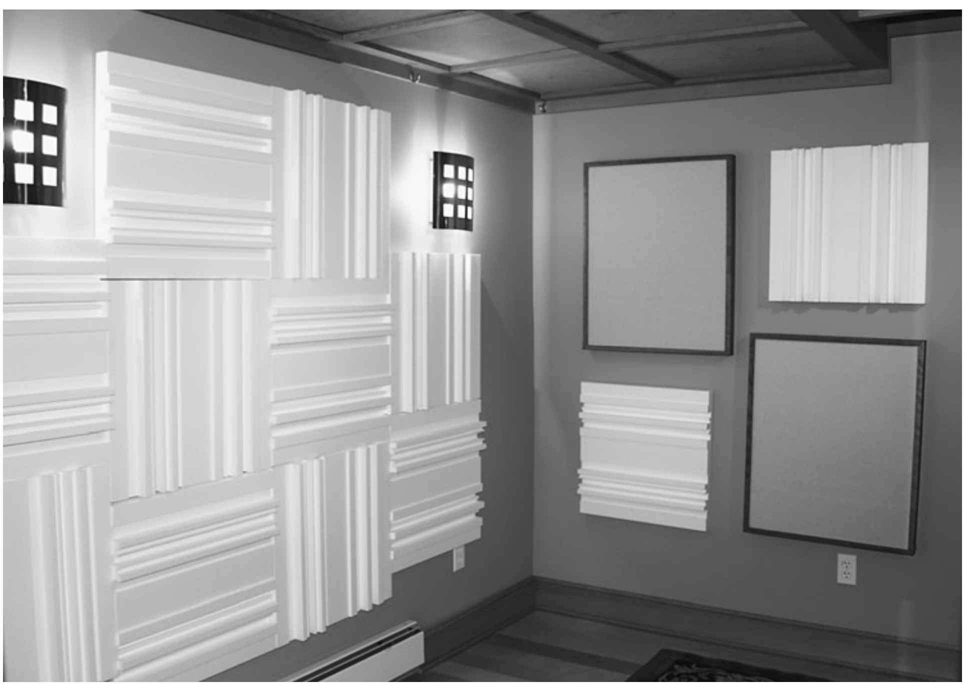


FIGURE 3.11

Diffusers and absorbers in recording room provide a drastic change, I did notice an improvement in the room and plan to add additional diffusers and bass traps in the future.

Control Room

The control room presented a different set of problems. The room is also used as a composition room and study so I elected not to utilize double-wall construction. In hindsight this was a mistake. There have been several occasions where an additional quiet room would have been helpful, and as my ears have improved with regard to mixing and mastering, external sounds and other acoustic issues have been distracting.

The installation of a number of sound absorbers was a good first step. Future treatments will include bass traps, ceiling treatment, and additional absorbers and diffusers. Although a professional engineer would need to be more rigorous in the treatment of the listening space, it is still possible for me do good work and to enjoy the process of making music. I would emphasize that, while proper acoustical treatment is always desirable, acoustic products can be expensive and may not be practical in every situation. In many cases you can minimize some of these problems by considering speaker placement, utilizing headphones, and adding some inexpensive absorption.

A product like IK Multimedia's Advanced Room Correction System would be a good solution for some individuals. The software comes with a measurement microphone and room-analysis software. The software calculates the frequency response of the room at the mix position and creates an EQ to compensate for acoustical problems. Mixing is done with the EQ inserted into the main bus, but counter intuitively, it is necessary to disable the plug-in prior to bouncing to disk.

Conclusion

We have covered a number of concepts in this chapter. While acoustics is obviously a large topic, my goal is that you understand the basic principles and are aware of potential problems that can occur in a project studio, as well as treatments that can help your recording space sound better. In some instances you may be able to improve acoustics by utilizing existing furniture. For example, a bookshelf can function as a primitive diffuser and an old couch might just have some beneficial absorptive qualities.

There are a number of excellent books and online resources on the topic of acoustics and studio construction. One helpful resource is provided by Auralex, a company that manufacturers acoustic products. Auralex provides a document called "Acoustics 101" that is available at http://www.acoustics101.com. The document has many tips that are useful for the do-it-yourself recording engineer. I would also recommend books such as "Sound Studio Construction on a

Budget" or "Master Handbook of Acoustics" (both by F. Alton Everest) if you are contemplating the construction of a studio, the installation of acoustic treatments, or if you would simply like to learn more about acoustics. A number of other books relating to acoustics and studio construction are listed in the bibliography.

4

Using Microphones

In some ways, the selection of a microphone for a given recording is similar to picking a fine wine to complement a meal. Each microphone imparts a unique sonic color, and it's important to select a microphone that will contribute the most desirable characteristics for the intended application. This also means that forethought should be given when selecting microphones for purchase. Since there are very few one-size-fits-all solutions when it comes to microphones, it is advisable to invest in a cabinet of a least a few microphones that will provide options for a variety of recording applications. In this chapter we will explore common categories of microphones, examine their pickup characteristics, and learn how to use and place microphones.

Microphone Categories

Microphones are transducers—they respond to fluctuations in sound pressure and convert the fluctuations into an electrical signal (see figure 4.1). The way in which the signal is converted depends on the design of the microphone. Three categories of microphones are typically found in a recording studio: *dynamic*, *condenser*, and *ribbon*.

Dynamic Microphones

Most dynamic microphones do not require the application of power to operate. Inside the microphone, a coil of wire is attached to a diaphragm and surrounded by a magnetic field. Sound pressure waves cause the diaphragm to move, which causes the production of a small electrical signal when the attached coil moves in proximity to the magnet.

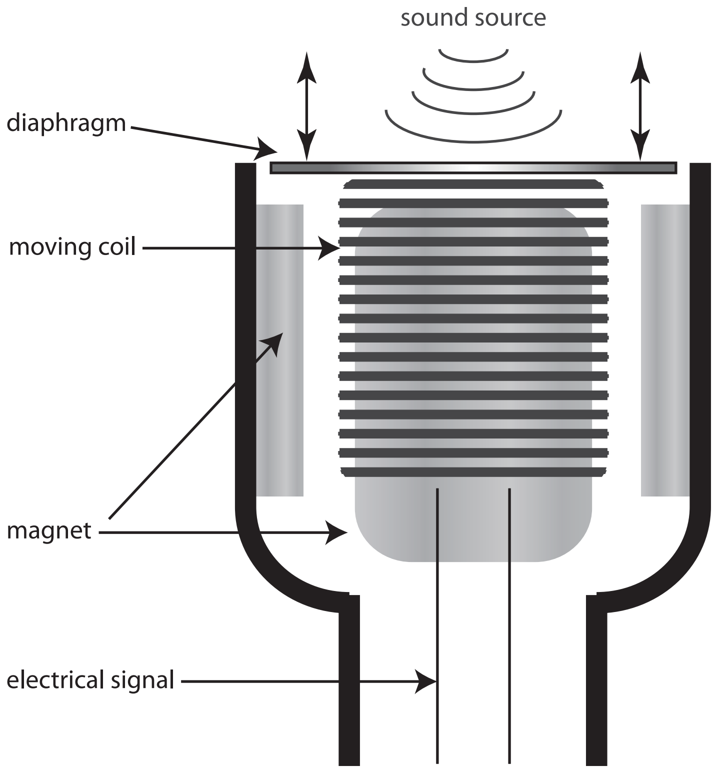


FIGURE 4.1

Transduction: converting sound pressure fluctuations to an electrical signal in a dynamic microphone

One characteristic of dynamic microphones is that they are less responsive to transients.[1] This makes sense when you consider the relatively large mass of a diaphragm and moving coil that must be set in motion in response to fluctuations in sound-pressure waves. This is why dynamic microphones tend to sound less crisp or transparent than other types of microphones. However, dynamic microphones can impart a unique color that can be wonderfully effective when recording a variety of sources, such as amplifiers, voice, drums, brass or auxiliary percussion. One advantage of dynamic microphones is that they can be useful where leakage is a concern. Given their relatively poor ability to capture weak signals, dynamic microphones can be a good choice when recording several instruments (such as a saxophone or brass section) where the players are in close proximity to one another.

The venerable Shure SM57 (see figure 4.2) and SM58 dynamic microphones are an excellent choice for many close-miking applications and the relatively inexpensive price point (around $100) is within reach of budget-conscious musicians. You can hear many audio excerpts recorded with the SM57 and SM58 at the companion Web site.

EXAMPLE 4.1

Ribbon Microphones

Ribbon microphones function in a similar way to dynamic microphones, but the diaphragm consists of a very thin strip of metal that is placed in a magnetic field. Ribbon microphones are more responsive to transients than dynamic

FIGURE 4.2

Dynamic, ribbon, and condenser microphones

microphones since the diaphragm has less mass and, thus, can respond more readily to changes in sound pressure.

Ribbon microphones can be a good choice for many applications, such as recording brass and woodwind instruments, over a drum set, or as a room or ambient microphone. The output of ribbon microphones tends to be low so the microphone will need to be paired with a preamplifier that provides enough gain.

A word of caution: some ribbon microphones can be damaged by the application of phantom power so it's important to ensure that phantom power is not applied to a ribbon microphone unless instructions from the manufacturer indicate that it is safe to do so.

Although professional ribbon microphones can be very expensive, some manufacturers such as Cascade offer ribbon microphones in the range of $150 to $300. You can hear samples of the Cascade Fat Head II (see figure 4.2) at the companion Web site.

EXAMPLE 4.2

Condenser Microphones

Condenser microphones are a mainstay of most professional recording studios. Unlike most dynamic and ribbon microphones, condenser microphones require a small amount of voltage to function. Voltage is typically applied via *phantom power,* which is carried over a standard XLR microphone cable from a recording console or audio interface. Some condenser microphones are powered by a battery or dedicated power supply.

The diaphragm of a condenser microphone is placed in close proximity to a fixed backplate and the side of the diaphragm facing the backplate is coated with gold or nickel.[2] A small amount of voltage is applied to create a capacitance, and as the distance between the movable diaphragm and fixed backplate changes, the capacitance changes, causing a change in output voltage.

Condenser microphones are typically described in terms of the size of the diaphragm: large diaphragm or small diaphragm. *Small-diaphragm* microphones typically have the best transient response and are well suited for recording acoustic instruments such as strings, guitar or piano. They can also be effective as a drum overhead or when recording an ensemble. These microphones often exhibit a crispness that works well when recording cymbals or other instruments with a great deal of high-frequency content. *Large-diaphragm* microphones are often described as exhibiting more warmth and are used on a wide variety of sources, such as voice, piano, and acoustic bass.

There is an astounding array of condenser microphones, both small and large diaphragm, available from a wide variety of manufacturers. Some microphones such as those manufactured by Neumann cost many thousands of dollars, but some very solid microphones such as the Rode NT1-A and Audio Technica 3035 (see figure 4.2) start at around $200. A recent microphone roundup in *Electronic Musician* magazine listed several viable options such as the Sterling ST51 and M-Audio Nova that cost as little as $100. Keep in mind that, while a $100 condenser microphone can be very usable, it is unreasonable to expect such a microphone to compare to a high-end microphone, so you will want to take time to carefully evaluate each microphone. As the reviewer points out, "the low prices make it easy for anyone who doesn't have a large diaphragm condenser to get one, and those who do will generally see improvements in the sound quality of their recordings."[3]

Stereo Microphones

A number of microphone manufacturers, including AEA, Audio Technica, Avantone, Rode, and Shure, offer stereo microphones. One benefit of a stereo microphone is ease of setup: you need only place the microphone at an advantageous position and can be assured that the capsules are engineered to be oriented in an optimal stereo configuration. The downside is a loss of flexibility. Some stereo microphones such as the Rode NT4 have a pair of fixed capsules in a permanent XY position, while microphones such as the Shure VP88 provide a bit more flexibility. It is a good idea to purchase a matched pair of condenser microphones, such as the Oktava MK012s, instead of a single stereo microphone if you plan to have just a few microphones. A matched pair will offer the flexibility to position the microphones in a number of ways, such as in a spaced pair, XY, or ORTF configuration.

Frequency Response, Equivalent Noise, Maximum SPL

Some microphones have a relatively flat frequency response, but other microphones are designed to emphasize certain frequencies. For example, the Electro-Voice N/D 868 is a microphone that is primarily designed for recording kick drum while the Shure SM58 is designed with brightened midrange to enhance vocals (see figure 4.3). It is advisable to purchase a microphone with a relatively flat frequency response if you plan to use the same microphone for multiple tracks because a frequency bump can be overemphasized in a mix if used on several tracks in the same song.

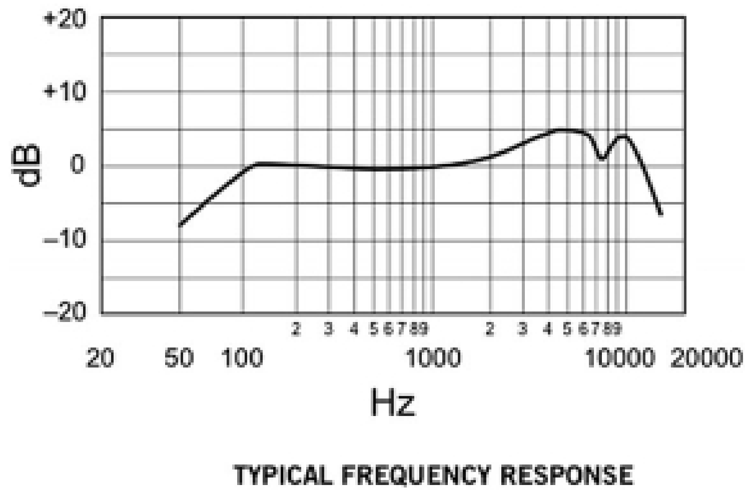


FIGURE 4.3
Frequency response of a Shure SM58 microphone

The frequency response of a microphone may change depending on the orientation and proximity of the microphone in relation to the sound source. For example, a directional microphone may emphasize bass frequencies when placed close to a sound source—a phenomenon known as *proximity effect*. Proximity effect is not necessarily a bad thing—the effect can be used to advantage when recording a vocalist or voice-over.

All microphones add a certain amount of noise. In most cases, this will not be a problem as the noise will be virtually imperceptible. However, you will want to select a microphone with a very low self-noise rating for critical applications such as recording an acoustic performance with a wide dynamic range, as self noise *can* be an issue in extremely soft passages with some microphones. An *equivalent noise* rating indicates the level of sound pressure that is required to create a voltage equal to the amount of noise produced by the microphone. A low number such as 5 dB SPL (A-weighted) indicates that a given microphone will be able to record very soft signals without adding a significant amount of self noise. The term *A-weighted* means that frequencies have been weighted to more closely match the way humans perceive of sound.

Another useful specification is the *maximum SPL*. This figure refers to a microphone's ability to respond to extreme sound-pressure levels. The maximum SPL indicates the sound-pressure level required to produce a certain amount of distortion (usually .5 or 1%). Some condenser microphones provide a pad that can be used to prevent the distortion that can occur when high sound-pressure levels cause the microphone capsule to overdrive its internal circuitry.

Polar Characteristics

The term *polar pattern* describes the way in which a microphone responds to the location of a sound source. Some microphones, such as the AKG C 3000, have a fixed pattern while multipattern microphones, such as the AKG 414, provide a switch to select from a number of pickup patterns. Some manufacturers provide interchangeable capsules so that the pickup pattern can be altered by swapping capsules.

Omnidirectional

Omnidirectional microphones (see figure 4.4) respond to sound from all directions, and they are a great choice when you want to include some room ambience in a recording. Omni microphones do not exhibit proximity effect so they can also be useful when placed close to a sound source, but they can also be effective when used in pairs for stereo recording.

I have found omnidirectional microphones to be useful as a sort of acid test when trying to improve the acoustics in my home studio. It is relatively easy to get a good sound when placing a directional microphone close to a sound source, but an omni microphone will reveal many of the undesirable characteristics of a room when placed at a distance.

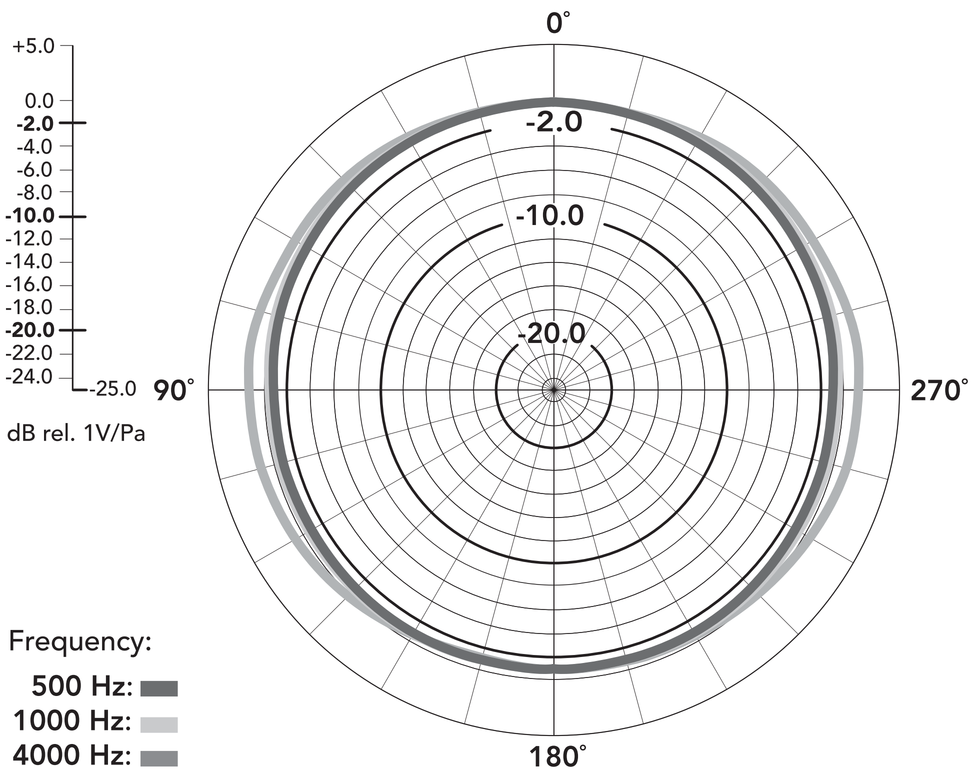


FIGURE 4.4

Omnidirectional pattern of an NT2-A (© RØDE Microphones. Used by permission.)

Bidirectional

A bidirectional or figure-eight pattern (see figure 4.5) refers to a pickup pattern where sounds are picked up equally well in front of and behind the microphone. Ribbon microphones are usually bidirectional and multipattern condenser microphones typically have a bidirectional setting. This pickup pattern can be useful for a number of situations, such as recording backup singers (one on each side of the mic) or as part of a mid-side or Blumlein array. It is interesting to note that in a recent series of microphone tests, classical musicians involved in three separate sessions preferred the sound of the bidirectional pattern over other patterns when I used the pattern to record their instruments in both mono and stereo.

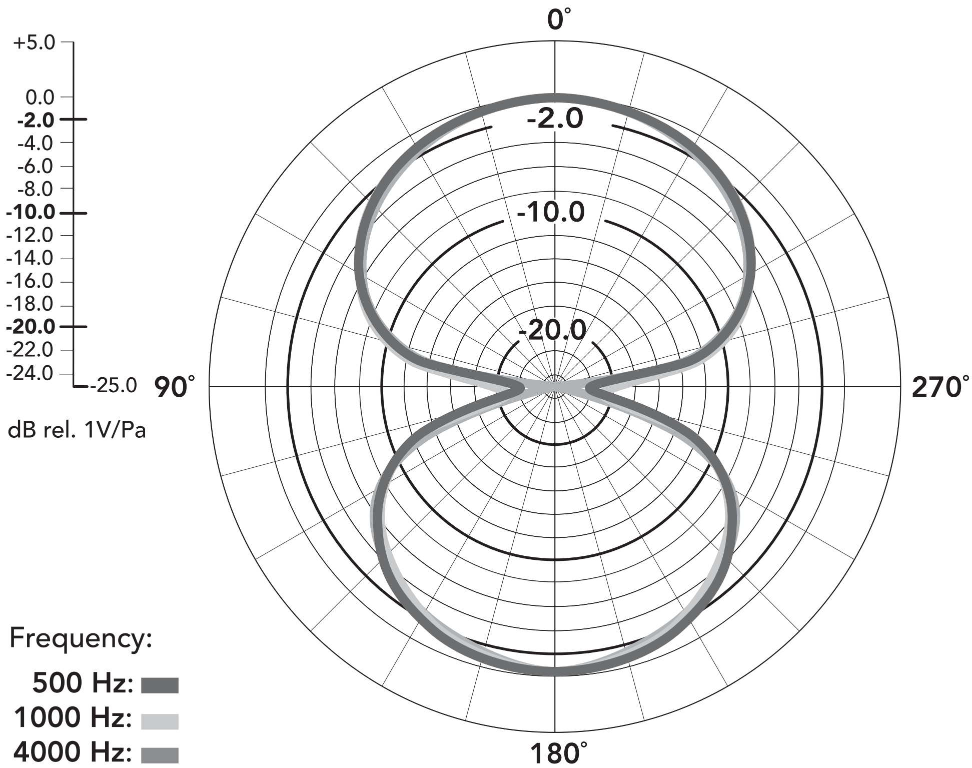


FIGURE 4.5

Bidirectional pattern of an NT2-A (© RØDE Microphones. Used by permission.)

Cardioid

The cardioid or unidirectional pattern (see figure 4.6) is a heart-shaped pattern that primarily picks up signals emanating in front, and to an extent the sides, of the microphone while minimizing signals from behind. Cardioid microphones are useful in the context of multitrack recording, where it is generally desirable to sonically isolate each track. Given their directional response, cardioid microphones tend to minimize reverberance and can help prevent leakage between instruments. Cardioid microphones are also useful in live recording since they can generally be placed at greater distances from a sound source than can omnidirectional microphones without adding an undue amount of room reverberance or audience noise.

Variations on the cardioid pattern include the supercardioid and hypercardioid. A *supercardioid* microphone is less sensitive to sounds emanating from the sides than a cardioid microphone and a *hypercardioid* microphone is even more extreme in rejecting sounds from the side. For this reason, super- and hypercardioid microphones can be a good choice when using multiple microphones concurrently when leakage from the side is a concern. These mi-

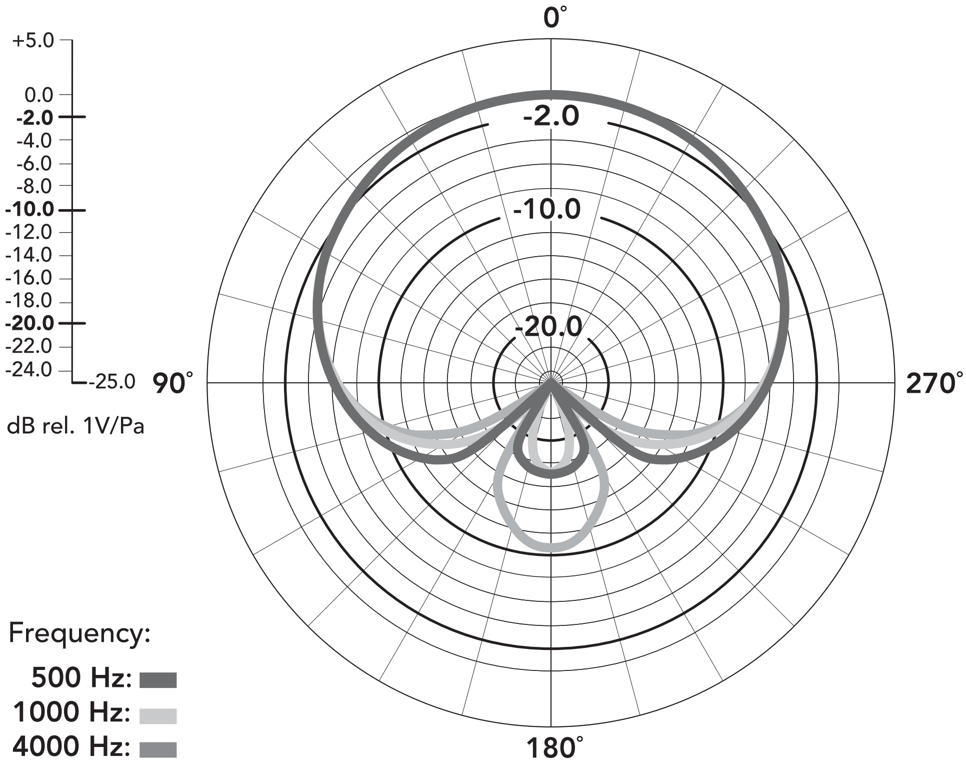


FIGURE 4.6

Cardioid pattern of an NT2-A (© RØDE Microphones. Used by permission.)

crophones could, for example, be used to simultaneously record hi-hats and snare on a drum set. It is interesting to note that supercardioid and hypercardioid microphones do pick up more sound from the rear than does a cardioid microphone (see figure 4.7).

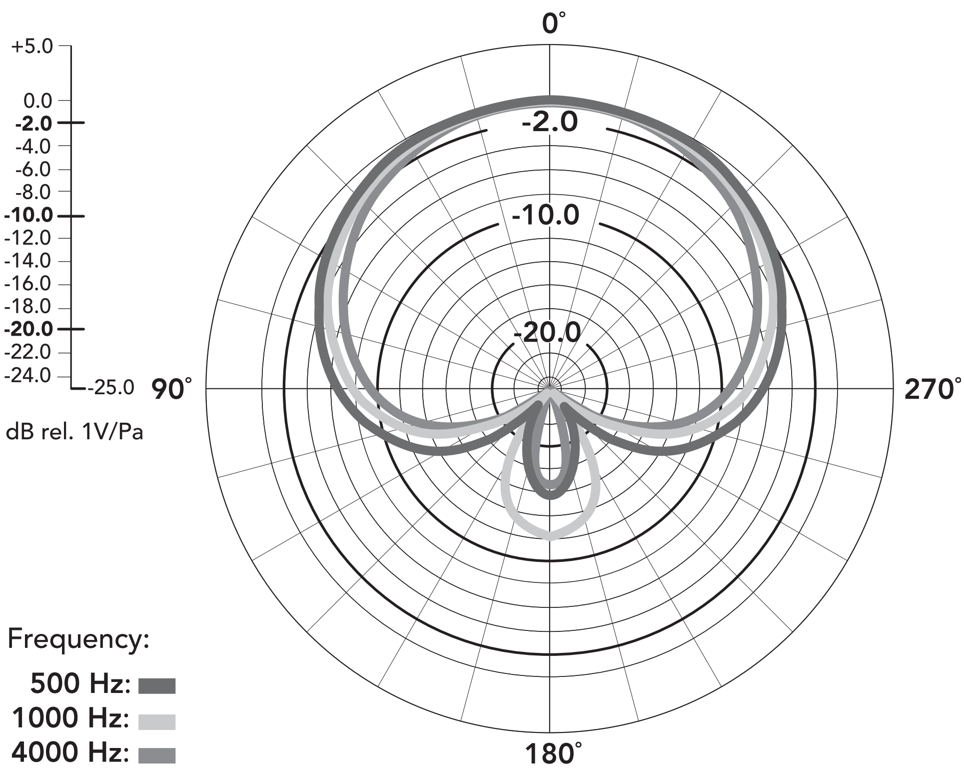


FIGURE 4.7

Supercardioid pattern of a R1 (© RØDE Microphones. Used by permission.)

Learning to Use a Microphone

In the following sections we will explore common microphone placement strategies. I would stress that, while the standard techniques are a good starting point, there are innumerable variations so I urge you to experiment to develop a sense of the parameters that will yield good results. While it's tempting to set up a pair of microphones in order to capture a big stereo image, it may be helpful to start with a single microphone in order to learn the unique sonic characteristics of your microphone and to explore how parameters such as distance, orientation, and polar pattern affect a recording.

Listen

Before recording, take a few minutes to objectively listen to the sound source. Walk around the recording room to see if there is a sweet spot. Listen closely to the instrument to see where a microphone could be placed to best advantage. I sometimes find it helpful to cover one ear so that my hearing is similar to how a microphone "hears."

Distance

Consider the type of music you intend to record. In general, classical music and some types of acoustic jazz are recorded with more ambient sound than styles such as pop and rock, where microphones are usually placed close to the individual instruments. Close microphone placement can yield a larger-than-life sound and minimizes ambience. This provides more flexibility in being able to add reverb and other artificial effects during the mixing process. On the other hand, placing microphones farther from the source can yield a sound that is more natural.

Select a Pickup Pattern

The selection of an appropriate polar pattern is an important decision. Cardioid microphones are frequently used for close miking and are often the best choice when simultaneously recording several instruments since the cardioid pattern is less sensitive to sounds that arrive at the back of the microphone. However, an omni microphone can also be used for close or distant miking and is often a good choice for classical or other types of acoustic music. A bidirectional pattern can provide some of the "direct" characteristics of a cardioid pattern while adding an element of ambience.

Place and Adjust

Once you have developed a sense for the appropriate distance and polar pattern, it's time to actually place the microphone. Consider how sound emanates from the instrument and aim the microphone so that it is in an advantageous position. It's often helpful to visualize secondary sounds as well as the primary sound. For example, the sounds emanating from the fret board are an intrinsic part of an acoustic guitar performance so you might want to place a microphone to get a balance of sound from the body and the frets (or utilize a second microphone).

At this point you will probably want to make some adjustments to the position of the microphone. Try moving the microphone closer to the source if the sound is indistinct or you hear too much ambience. Similarly, move the microphone away from the source if the sound is artificial or strident or you hear too many mechanical sounds, such as the hammers and dampers of an acoustic piano. Also listen to see if the sound can be improved by moving the microphone laterally. Often, a small lateral adjustment can make a big difference.

Selecting the Right Microphone for the Job

As with most things relating to music, practice can help you develop the skills that will enable you to select a microphone and place it to best advantage. With that said, the following suggestions can provide a helpful starting point.

Dynamic microphones like the Shure SM57 tend to have a rugged sound that works well on instruments such as snare, toms, hi-hat, and guitar cabinets. Small-diaphragm condenser microphones tend to have a crispness that sounds great on strings, drum overhead, piano, and acoustic guitar. Large-diaphragm condenser microphones often have a transparent yet warm sound that works great for vocals and most acoustic instruments such as bass, piano, guitar, violin, and saxophone. Ribbon microphones can sound very natural and may tame the stridency of instruments such as trumpet and saxophone. Ribbon microphones can also sound great when used as a drum overhead or when recording most acoustic instruments, and some engineers use them to record guitar cabinets. Be sure to experiment because many other combinations are possible. For example, in a recent session I tried a number of microphones on an acoustic bass and was surprised to find that an inexpensive dynamic microphone sounded very good when placed near the bottom of the instrument below the bridge.

Stereo Recording

There are a number of situations in which stereo recording makes sense, such as recording an ensemble or choir, and it is often advantageous to use two or more microphones when recording instruments such as a drum kit, piano, marimba, or horn section. A potential downside is that phase issues can be a problem when more than one microphone is used to record a sound source. When the same signal arrives at two microphones at a slightly different time, the variance in phase of the signals can cause a number of anomalies, such as comb filtering or a signal that sounds weak. To complicate matters, problems might not be evident in stereo so it's a good idea to periodically listen to stereo tracks in mono to ensure that your stereo tracks are translating well when combined into a mono signal.

While it is tempting to always record instruments such as piano or guitar in stereo, it is also helpful to consider how multiple instruments will be blended in a mix. In some cases, multiple mono sources will be more effective (and easier to control) in a mix than multiple stereo sources. A number of common techniques can be employed to capture a sound source in stereo, and you can listen to a wide variety of mono and stereo excerpts at the companion Web site.

EXAMPLE 4.3

Spaced Pair (3:1)

A pair of microphones can be spaced apart from one another and a stereo image is created by differences in the arrival time of the sound source, as well as differences in level. The spaced-pair technique can be useful when recording an ensemble from a distance, but it also works well in close-miking applications, such as recording a solo guitar or piano.

In general, the distance between microphones is three units for every unit away from the sound source, although this ratio is not always practical (see figure 4.8).

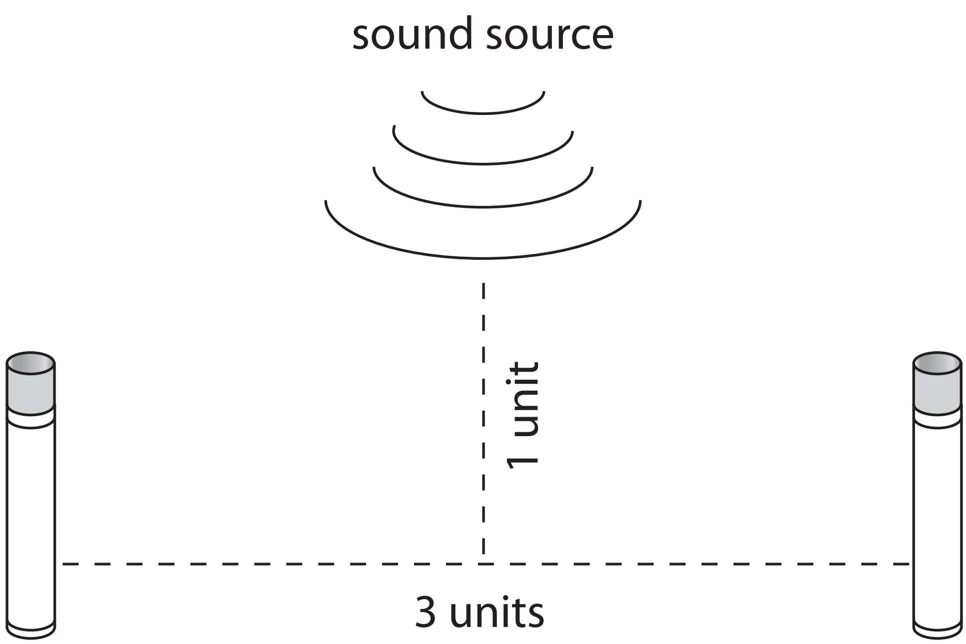


FIGURE 4.8
Spaced pair

A variety of pickup patterns can be employed when utilizing a spaced-pair configuration. A pair of omnidirectional microphones can yield a natural sound that includes both the sound source and the room (depending on the distance from the source), while a pair of cardioids are useful when a more direct sound is desired. Bidirectional microphones are also an option, although the bidirectional response may be problematic in some situations, such as when recording an ensemble in front of a noisy audience.

In addition to potential phase issues, sounds that are off-center may be unfocused when recording with a spaced pair.[4] Recording engineers sometimes add a third omnidirectional microphone in the center of a spaced pair, although this can create additional phase problems.

Coincident Pair (XY)

A coincident or XY setup offers good compatibility with mono and is achieved by placing the diaphragm of two cardioid microphones as close as possible to one another at an angle ranging from 90 degrees to as much as 150 degrees (see figure 4.9). Unlike a spaced pair, which relies on differences in time between signals, the stereo image of a coincident pair is primarily determined by differences in signal level.

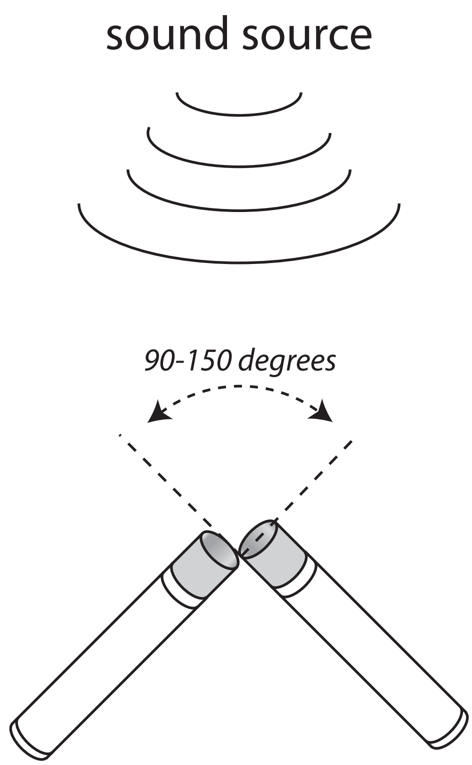


FIGURE 4.9

XY setup

Blumlein

The Blumlein configuration (see figure 4.10) consists of two bidirectional (figure eight) microphones positioned at 90 degrees. This technique offers a very natural sound since the microphones respond to sounds from both the front and back, offering a sort of "surround" coverage. Note that microphones are typically placed off-axis from the sound source in this configuration.

I have used this technique with a pair of inexpensive ribbon microphones and was pleased with the results, although the technique is not as useful in a recording space with poor acoustics or when audience noise is likely to be a problem.

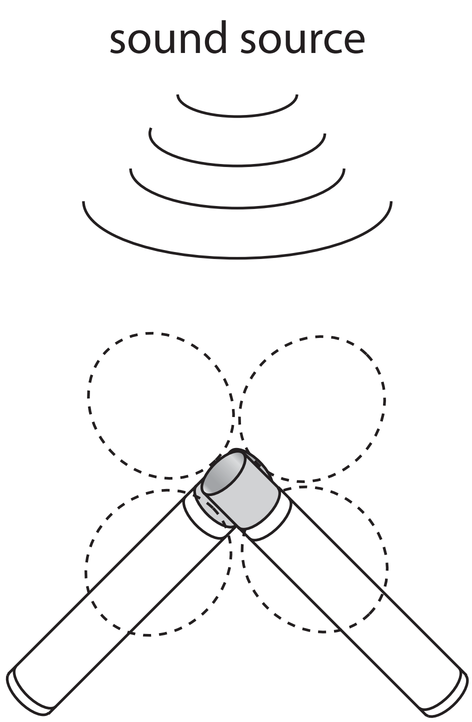


FIGURE 4.10

Blumlein setup

ORTF (Office de Radiodiffusion-Television Française)

The ORTF technique is a near-coincident technique similar to XY, but a pair of cardioid microphones are placed 17 cm apart and at an angle of approximately 110 degrees (see figure 4.11). As with XY technique, ORTF provides fair mono compatibility and works well when recording ensembles, as well as instruments such as piano or marimba. An ORTF template is provided in the appendix of this book. Simply trace the template onto a thin piece of wood or cardboard to easily configure your microphones for ORTF.

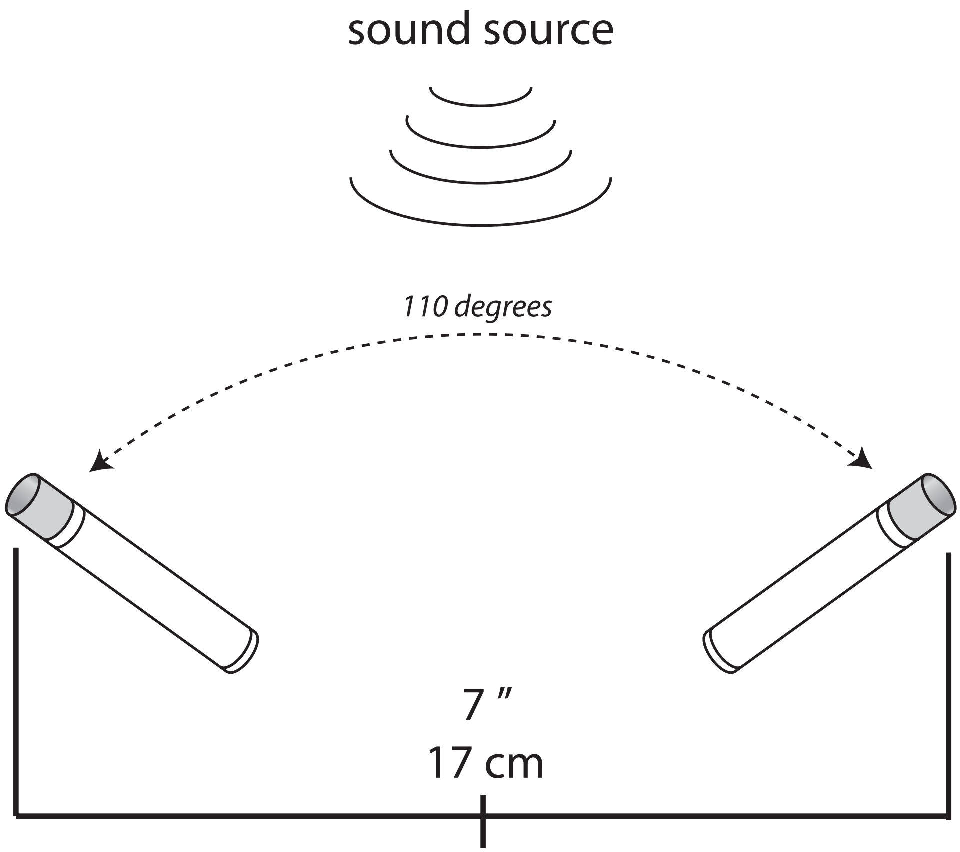


FIGURE 4.11
ORTF

Mid Side (MS)

Mid-side recording seems to be making a comeback. I have read several articles on the subject in trade magazines over the past few years, and I have found the technique to be useful in several applications. One microphone (usually cardioid or omni) is used as the mid or center microphone. A bidirectional (figure eight) microphone is placed at a right angle and designated as the side (see figure 4.12).

Unlike the other stereo techniques mentioned thus far, an MS signal must be decoded. Some Digital Audio Workstations provide MS decoding in the form of a virtual plug-in. However, no special equipment or software is required. To decode an MS signal, make a copy of the side (figure eight) track and paste it into a new track in your DAW. Next, invert the copy of the side signal (an inversion option should be available in your DAW) and pan each of the side tracks hard left and right. For convenience, it's a good idea to group the two side tracks in your DAW or route the output to a sub bus. At this point you can adjust the level of direct and ambient sound by altering the level of the center and side channels. MS technique offers the best mono compatibility since the side channels cancel each other out when combined in mono.

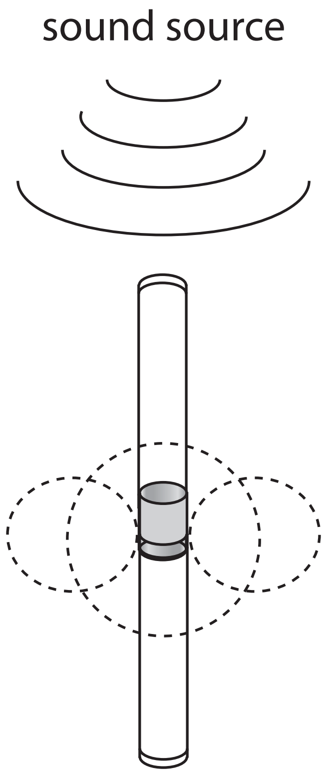


FIGURE 4.12
MS Technique

Ensemble Recording

Many music educators are interested in making good recordings of their orchestra, band, or choir, and the stereo techniques listed above are a good starting point. However, stereo configurations such as XY or ORTF can present some unique challenges. In order to capture the stereo field of a large orchestra or choir, it may be necessary to move the microphones back to a point where there is an undesirable ratio between the amount of direct and ambient or room sound. Off-axis coloration can also be a concern. For these reasons, a spaced pair of quality condenser microphones may provide better results. Start by listening carefully to determine the optimal distance from the ensemble and place two microphones an equal distance to the left and right of this point. Next, pan the microphones left and right and experiment with the distance between the microphones, as well as their height above the choir or orchestra. It is possible to use any polar pattern with this configuration, but the omnidirectional pattern is common. Depending on the distance between microphones, there might be a hole in the center of the stereo field. Some engineers will place a third microphone in the center to fill in this part of the stereo image (see figure 4.13).

A variation on the previous setup involves using a matched pair of microphones in the center (e.g., XY or ORTF configuration) in conjunction with a spaced pair of microphones. I recently used this configuration with a large orchestra and was pleased with the results. In this instance, an ORTF configuration worked well to capture the center of the orchestra and a spaced pair of omnidirectional microphones helped create a big sound by bringing the sides of the orchestra into focus. Not only did the side microphones help with the stereo image but also they essentially functioned as ambient microphones and added a sense of spaciousness and depth.

The combination of a spaced pair of microphones and one or two center microphones can also work well for *a cappella* choral music. Keep in mind that

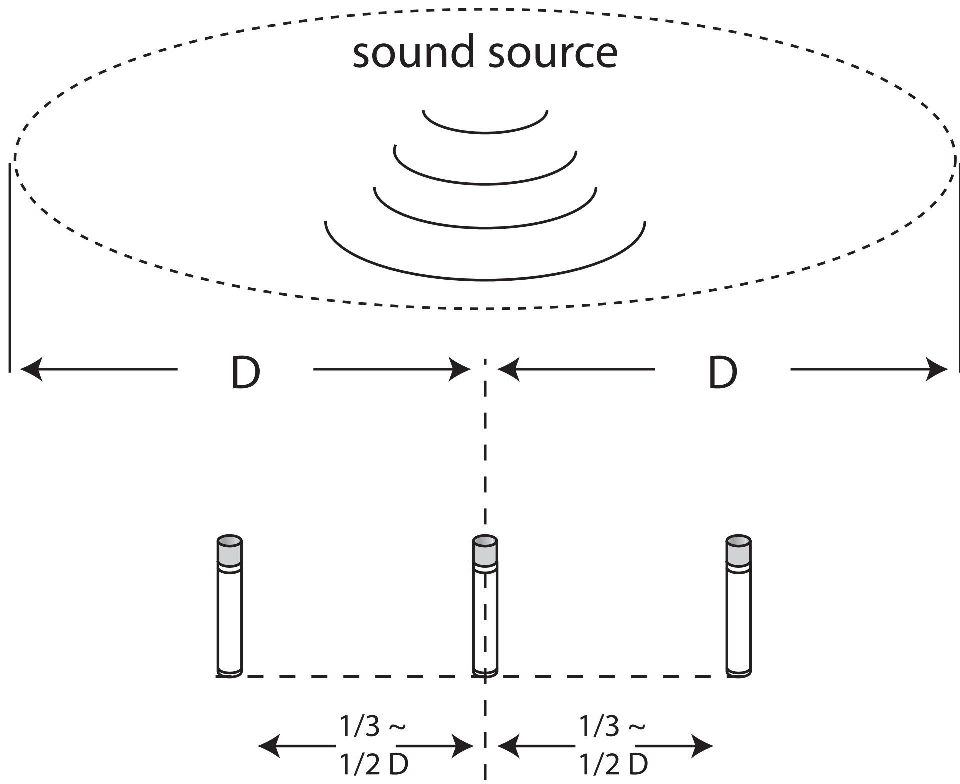


FIGURE 4.13

Spaced pair with center microphone

piano or other accompanimental instruments may sound indistinct when microphones are placed at a distance, so it may be helpful to shorten the distance between the microphones and choir or utilize accent microphones (see below) when recording instruments with a choir. You can listen to several choral and orchestral excerpts at the companion Web site.

EXAMPLE 4.4

In his book *The Recording Engineer's Handbook*, Bobby Owsinski describes another method for ensemble recording known as a Decca Tree—a technique that originated at Decca Records. With this technique, three omnidirectional microphones are placed in a triangle approximately ten to twelve feet above the conductor (see figure 4.14).

Accent Microphones

It is sometimes useful to add one or more accent microphones when recording a large ensemble or choir. For example, two or three microphones could be used to capture a choir or orchestra using the techniques illustrated in the last section and one or more additional microphones could be placed close to a vocal or instrumental soloist. Similarly, it can be effective to place an accent microphone near the piano when recording choral music. This will provide a measure of control over the presence of the piano and the balance between the piano and choir. One of the benefits of multitrack recording technology is that it can foster experimentation. After all, hard-drive space is inexpensive so it never hurts to record additional tracks if a decision regarding the use of accent microphones can be delayed until the mixing phase.

based on an illustration by Ron Streicher

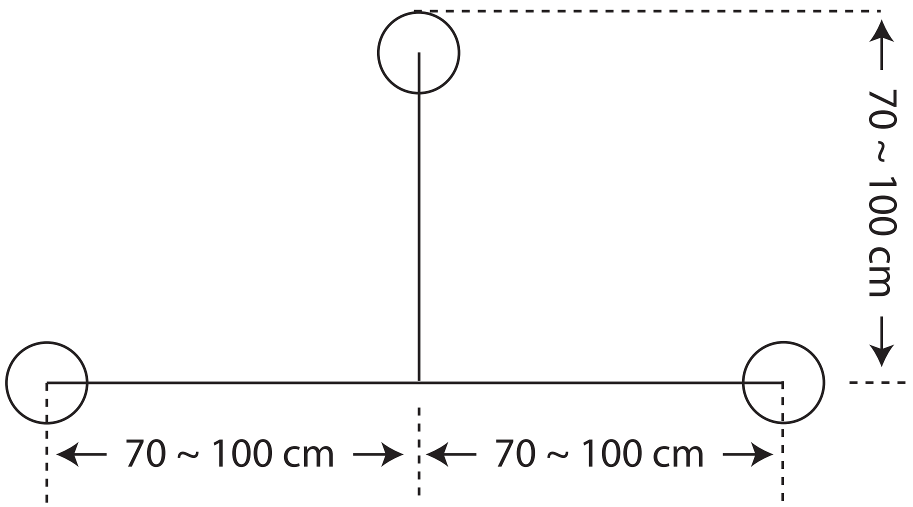


FIGURE 4.14
Decca Tree

Unmatched Microphones

In most situations it is desirable to use a matched pair of microphones when recording in stereo, but there are some instances when an unmatched pair of microphones can be useful. Although I wouldn't recommend this technique when recording an ensemble, unmatched microphones can be effective in some situations. For example, some recordists use large-diaphragm and small-diaphragm condensers when recording acoustic guitar. The large-diaphragm microphone is used to record the big, warm tone of the body of the guitar while the small-diaphragm microphone is placed near the fret board. I have also found that the use of an omnidirectional microphone and cardioid microphone can be effective when close-miking a piano. The omnidirectional microphone is placed over the bass strings and the cardioid is placed over the treble strings in a spaced pair. Although this technique doesn't translate well to mono, one student described the sound as being "like warm butter." Similarly, some engineers will use two matched microphones over the strings of a piano and place a third omnidirectional microphone at the end of the instrument just above the rim.

Preamps

Although the focus of this chapter is on microphones and their usage, microphone preamps are an important part of the equation. Just as different microphones impart a sonic color to a recording, preamps can have a neutral, positive, or negative effect on a signal. Some engineers have expensive boutique preamps at their disposal, and I have often wondered if high-end preamps are really necessary. I had the opportunity to test this question, as well as several others, on a recent trip to Western Carolina University. I was assisted by Dr. Bruce Frazier, Endowed Chair of Commercial Electronic Music, at the university, and Wells Gordon and Aaron Smith, two students in the recording arts program.

Case Study: Boutique Preamps and Microphones

In the first set of experiments, we ran a variety of microphones simultaneously through Focusright, PreSonus, and Mbox 2 preamps. The Focusright preamp is found in many high-end studios while the particular PreSonus preamp and the Mbox 2 represent a "prosumer" level of performance. I played the same ballad numerous times on a Yamaha grand piano to test a wide variety of combinations of microphones and preamps. (Numerous examples are available on the companion Web site.) My subjective assessment was that the top-of-the line-preamp did sound noticeably better than the less expensive preamps, owing to a subtle coloration or warmth that was imparted to the recording. Although it was somewhat disheartening to learn that the best-sounding preamp costs more than $3000, I would stress that PreSonus and Mbox preamps were certainly very usable.

EXAMPLE 4.5

In another set of experiments we compared the sound of a number of microphones through the same set of microphone preamps. It was very interesting to compare expensive products like the AKG C12 ($5000) and Royer R-121 ($1300) with more modest options such as an AKG C451 ($550). The results of this experiment were encouraging. All of the microphones performed admirably, but none of the microphones, even the expensive ones, exhibited a jaw-dropping level of performance. Each combination of microphones exhibited unique characteristics, and though many of the sounds I heard from the expensive microphones were appealing, to be sure, the less expensive microphones were very usable.

This brings up an interesting point: one of the things that was made vividly clear in this research is that the recording space, instrument, and performance are more important than microphones or preamps. Unfortunately, the piano in this high-end studio was out of tune on the day of the session. Although the piano wasn't in horrible shape, I believe that many of the subtleties of the more expensive microphones were masked by intonation problems. Phrased another way, high-end gear will not make an out-of-tune piano sound better. It is interesting to compare some of the piano recordings made at the high-end facility with the budget recordings that were done in my home studio; in my estimation, the tuning of the instrument was a more important factor than a particular combination of microphones or preamps.

I would not read too much into this research since our experiments represent an obviously narrow application; other subtleties would be evident if the microphones were used to capture another instrument, voice, or an ensemble performance. Still, many readers will question the price points of a variety of microphones and preamps, and the many online examples will provide a good starting point as you evaluate variables and prices. The online samples also provide an excellent opportunity to train your ears to hear the subtle sonic characteristics of a wide variety of microphones.

Microphone Technique

Much of the "art" in recording arts comes from microphone technique, and it is an aspect of recording that can be a challenge to those of us who are new to the discipline. In this next section we will consider some of the common microphone techniques that can be employed to capture a number of instruments. I would stress that these suggestions should be treated as a point of departure. For each of the techniques listed in this section, there are many other effective techniques utilized by engineers. Make a habit of experimenting because that is the best way to develop fluency and learn new techniques.

As with all of the techniques listed in this chapter, you will want to explore the effect of placing microphones at varying distances in order to balance the amount of ambient or direct sound. You will also want to experiment with the position and orientation of microphones in relation to the sound source since small adjustments can have a big impact on the sound.

Most of the examples in this section are geared to recording instruments as part of a multitrack session. These techniques will generally work well for popular music genres such as pop, rock, country, and some styles of jazz. In general, move a microphone farther away from the source for a natural sound that is suitable for classical music. However, distant microphone placement can be useful for many styles so be sure to experiment.

There are a number of books and articles that I have read over the years and as part of the research for this book. One of the many books I would recommend is *Professional Microphone Techniques* by David Miles Huber and Philip Williams. The book is concise and the authors provide many alternative techniques, as well as suggested methods for less common instruments, including accordion, banjo, harp, and marimba.

Solo Voice

Any number of microphones including dynamic, ribbon, or condenser can be used to good effect with a lead vocalist. A large diaphragm condenser microphone in a cardioid pattern is frequently used, and depending on the microphone, can impart a warm yet airy sound that works very well for popular styles. The microphone is typically placed fairly close to the singer—twelve inches is a good starting point—directly in front of the vocalist's mouth. For a different sound, it may be useful to place the microphone slightly off-axis. Off-axis orientation is also helpful in preventing plosives. I usually use a pop screen when placing a microphone close to a vocalist.

Many variations are possible. For example, an omnidirectional condenser microphone can work very well for a lead vocal, as can a dynamic microphone. A Shure SM58 can impart a robust sound and is typically placed very close to the singer. A number of vocal excerpts are available at the companion Web site.

EXAMPLE 4.6

Acoustic Guitar

One or two microphones are typically placed close to the guitar for a big sound that works well for popular styles. A common technique is to orient a cardioid condenser eight to twelve inches from the body of the guitar and a second cardioid microphone approximately halfway up the neck. In this spaced-pair technique, the body microphone provides fullness and warmth and can be blended with the secondary microphone to create a full-bodied sound.

For classical guitar, try moving the microphones back for more ambient sound. A coincident or near-coincident pair or mid-side configuration can be effective when used on acoustic guitar, but a single directional or omnidirectional microphone can also sound great. One technique that I particularly like for classical guitar is to place a matched pair of condenser microphones in an XY configuration about eighteen inches in front and slightly below the guitar. Angle the microphones up toward the top face of the guitar and place an omnidirectional ambient microphone above and about three feet back from the performer. You can listen to this and other examples at the companion Web site.

EXAMPLE 4.7

Electric Guitar

Although it is possible to use a DI to insert an electric guitar directly into a recording console, most engineers place a microphone near the amp. One common technique is to use a dynamic microphone such as an SM57 directly in front of the grill. Different tones can be achieved by varying the position and orientation of the microphone and, of course, the type of microphone itself has much to do with the sound. Be sure to listen to the examples on the companion Web site because these excerpts illustrate the importance of microphone placement. For example, the samples illustrating the placement of a microphone at the center and side of an amplifier cone sound like different amplifiers, yet the microphone was moved just a few inches.

EXAMPLE 4.8

I once had the pleasure of recording a few tracks for jazz guitarist John Stowell. He had a very specific vision for the tone of his guitar and suggested using one microphone on the amplifier and another on his fret board. I placed his amp in my “iso closet” and placed a condenser microphone near the fret board of the instrument. The resulting sound was a nice blend of the fullness of the amplifier with some of the natural sounds emanating from his instrument.

Electric Bass

As with an electric guitar, a microphone can be used to record a bass cabinet or a direct box can be used to connect the bass directly to a recording console. I particularly like the sound of a DI on fretless bass. You can listen to several excerpts at the companion Web site.

EXAMPLE 4.9

Acoustic Bass

Acoustic bass can be a challenge to record because different notes tend to speak at different levels and it is particularly hard to get a convincing bass sound when recording multiple instruments in the same room. As a backup, I usually use a DI box on the bass in addition to a dedicated bass microphone on a different channel. This provides the option to mix in a more present attack while retaining the natural warmth of the bass. A good starting position is to set a condenser microphone (small or large diaphragm) below the F holes and angled up slightly at the bridge. Some engineers will place a microphone below the neck joint but above the F holes and aimed directly at the body of the bass. Another technique is to place a directional microphone near the bottom of the bass just below the bridge. In my experience, each bass will require a fair amount of experimentation in order to find the optimal spot for microphone placement. You can listen to each of these bass recording techniques at the companion Web site.

EXAMPLE 4.10

Drums

Drum recording is a huge topic, but a good starting placement consists of a microphone positioned to capture the snare and hi-hat, a kick microphone, and one or two overhead microphones.

A dynamic microphone or cardioid condenser microphone is typically used to capture the snare and hi-hats. A common placement is to position the microphone slightly above the snare (see figure 4.15). Some engineers place a second microphone below the snare and angled up (although phase will need to be reversed) and an additional microphone can be utilized on the hi-hats. Many engineers will utilize a small diaphragm condenser for the hi-hats, but take care that the microphone doesn't get too close to the edges of the hi-hat because the whooshing of the hats does not sound good. I generally use a single microphone to capture the snare because overhead microphones will typically pick up plenty of hi-hat. However, individual microphones may yield more flexibility if your goal is to create a more produced sound.

A dynamic microphone is typically used on the kick drum and may be placed on either side of the drum. It generally works well to place the microphone on the front side of the drum opposite the beater or directly inside of the drum. With a bass drum, placement is often a balancing act between capturing some of the "thwack" of the beater and the full "boom" of the drum. Visit the

FIGURE 4.15

Placing a microphone on a snare

EXAMPLE 4.11

companion Web site to compare the placement of a microphone on the front, in back, and inside a kick drum.

Some engineers will use one or more microphones for tom-toms. A directional microphone can be placed above the head near the edge of each drum or a single microphone can be placed between two toms in order to capture the sound of both drums.

FIGURE 4.16

One approach to miking a drum set

It is common to use one or two microphones as an overhead above the drums. Small-diaphragm microphones are often a good choice as they typically enhance the brilliance of the cymbals. Many configurations are possible, but I have had good success utilizing a spaced pair of microphones (see figure 4.16). A coincident pair also works well, and this can be a good application for a stereo microphone. I own an inexpensive pair of ribbon microphones and those also work very well in a spaced pair configuration.

EXAMPLE 4.12

In some instances, an ambient microphone can be a good choice when recording drums. An ambient microphone is used in conjunction with the techniques described above, but the microphone is moved far enough from the kit so that it picks up the sound of the room. This ambience can then be blended with the closely placed microphones. You will find a number of drum excerpts, recorded with and without ambient microphones, at the companion Web site.

EXAMPLE 4.13

Piano

Piano might be one of the hardest instruments to record, given its large range, heft, mechanical noise, and dynamic range. A good starting position is to use two cardioid condensers placed in a spaced pair above the treble and bass strings (see figure 4.17).

FIGURE 4.17

Using a spaced pair of microphones on a grand piano

Mid-side technique can also work very well, and I frequently use that technique when I record my jazz trio. Place a figure-eight microphone inside the piano around middle C but oriented perpendicular to the player. The center microphone (cardioid or omni) is placed perpendicular to the figure-eight pattern. All of the common stereo techniques, such as coincident, near-coincident, and spaced-pair, can be successfully utilized to record a piano in stereo from a variety of distances. Each of these techniques can be heard at the companion Web site.

EXAMPLE 4.14

Saxophone

A large-diaphragm condenser microphone is often a good choice for saxophone. Start with the microphone about a foot away from and slightly higher than the bell, and angle the microphone so that it is aimed at or just above the bell. (Try placing the microphone below the bell of a soprano saxophone.) Ribbon microphones can also be effective when recording saxophone.

EXAMPLE 4.15

Flute

The tone of a flute comes partly from the sound of the player's breath moving into the mouthpiece, as well as the tone that emanates from the end of the instrument. For this reason, it often works well to place a microphone about twelve inches in front of and above the player in order to capture both elements. The microphone is typically placed at a distance for a classical recording in order to capture natural room ambience.

EXAMPLE 4.16

Trumpet and Trombone

A dynamic or condenser microphone can be a good choice for brass instruments including trumpet and trombone. A condenser microphone may yield a more brilliant tone while a dynamic microphone will tend to tone down the "brassy" tone of the instrument. Ribbon microphones often yield a smooth sound and may tone down some of the stridency inherent in brass instruments. A good starting microphone position is about eighteen inches from and slightly to the side of the bell. Visit the companion Web site to compare the sound of a trumpet recorded with large diaphragm condenser, ribbon, and dynamic microphones. As you will hear, each microphone imparts a distinct character to the instrument.

EXAMPLE 4.17

Violin

If you have ever attended a violin recital you have probably noticed that the violinist stood so that the face of the instrument was angled toward the audience in order for the sound to project. In a similar way, it often works well to position a microphone approximately two feet over the F holes but angled so that the diaphragm of the microphone is parallel to the face of the violin. Condenser microphones are typically used to record violin, although I recently used a ribbon microphone and was pleased with the sound.

Any of the stereo techniques can be effective when recording a string ensemble. Experiment with a variety of distances in order to find a good balance of direct and ambient sound. Several violin duo excerpts are available at the companion Web site.

EXAMPLE 4.18

Conclusion

There are obviously many other instruments and techniques to consider, but the techniques mentioned in this section are a good starting point for further exploration. One of the biggest joys of home recording is that you can take as many hours as you want to adjust the position of a microphone or experiment with polar patterns and stereo configurations. Keep in mind that the only goal of microphone placement is to capture the sound of an instrument or voice in a pleasing way, so *any* technique is viable if it sounds good to your ear. The only caution I would give is to be aware of issues such as phase and leakage, which can be a concern when multiple microphones are used. Other than that caveat, I would urge you to explore, and you will find that microphone technique will no longer be a lore that is known by recording engineers—microphone selection and placement will become a natural part of your musicianship.

You will find many audio excerpts utilizing a wide variety of microphones and configurations at the companion Web site. With the exception of the excerpts recorded at Western Carolina University, all were recorded in a home studio with relatively inexpensive microphones and preamps. You might also be interested to hear an excerpt of the Whitworth University Jazz Ensemble. For economic reasons, we used a number of inexpensive dynamic microphones running through an Allen and Heath console connected to an Alesis HD24 recorder. Although the sound does not rival a Rob McConnell and the Boss Brass recording, the results are reasonably convincing and we were able to complete a full-length CD by recording live in the school's performance venue. The project provided an excellent case study for budget recording, and you can read a description of the recording process used on the session in chapter 9.

EXAMPLE 4.19

5

Podcasting

Although podcasting is a relatively new concept, the Internet has fueled explosive growth in the type and variety of podcasts. Today, video and audio podcasts are available on a wide range of topics, from music and technology to politics and self-help. In this chapter we will consider the tools that are required to make a podcast, as well as techniques that will help your podcasts sound more professional. Although podcasting is not for everyone, the creation of a podcast provides a good introduction to audio recording, and many of the concepts that are explained in this chapter are directly applicable to music recording and editing.

Getting Started

Fortunately, very little equipment is needed to make an effective podcast. The basic requirements are a computer, microphone, and audio editing software. Freeware software such as Audacity will work very well for this application, so for most people, a microphone is the only item that has to be purchased to begin making podcasts.

Computer

Podcasts typically involve just a few tracks of audio data so very little computing horsepower is needed. However, you may want to consider a more powerful computer if you intend to make video podcasts since video editing and rendering can tax the resources of a computer.

Microphone

I typically advise against the purchase of USB microphones since they require a computer to operate and, thus, are rather inflexible. However, USB microphones such as the BLUE Snowball, Audio-Technica AT2020 USB, or Samson C01U offer convenience and are a good choice for this application.

In lieu of a USB microphone, consider a dynamic microphone such as the Shure SM58 or a condenser microphone along the lines of a Sterling Audio ST51 or Studio Projects B1. Dynamic and condenser microphones start at around $100 and have the obvious advantage of being able to do double duty to record an instrument or voice. A disadvantage of dynamic and condenser microphones is that an audio interface or mixer will be required since it is not possible to connect such a microphone directly to a computer. You might want to consider two microphones or a multipattern microphone if intend to conduct podcast interviews.

Audio Interface

You will need an audio interface to connect a non-USB microphone to your computer. Single-channel USB audio interfaces such as the M-Audio Fast Track USB start at around $100, but you will need to spend a bit more to get an audio interface with phantom power if you intend to record with a condenser microphone. (Refer to chapter 1, "Setting Up a Project Studio," for more information regarding the selection of microphones and audio interface.) Although it is possible to use a built-in computer soundcard to record audio, my experience has been that hum and other noises are typically a problem so you will likely achieve better results with an inexpensive audio interface.

Creating an Audio Podcast with Audacity

Audacity is an open-source audio recording and editing application that provides plenty of power for many tasks. It's an excellent tool to make audio podcasts, and the next several paragraphs will be devoted to this task. Audacity is available as a free download at http://audacity.sourceforge.net. Apple's Garage Band offers some nice features such as ducking that make it easy to create a polished podcast so that program is worth exploring if you happen to own a Mac. (Ducking is a technique that automates the process of turning down background music during a voice-over.)

Microphone Placement

In terms of microphone placement, it's generally a good idea to place the microphone close to your mouth in order to minimize any ambient noise. If you are using a directional microphone, you may want to experiment with very close placement to take advantage of a phenomenon called the *proximity effect*. Some announcers and vocalists use the proximity effect to create a big sound with an inordinate amount of bass. It is a good idea to use a pop screen when recording a podcast in order to avoid plosives—the burst of air that occurs when you say a word like "paper." (Instructions for building an inexpensive pop screen can be found in the final chapter of the book.)

You will want to consider several approaches if you intend to conduct podcast interviews. If two microphones are available, place the microphones so that the participants face each other. This is typical of a radio-station interview and allows more ease and flexibility in adjusting the relative signal level of each participant.

A multipattern condenser microphone can also be used to record a podcast interview. Set the microphone to a figure-eight or omnidirectional pattern and place the microphone midway between each person. You can listen to samples of each of these techniques at the companion Web site.

EXAMPLE 5.1

Setting Levels

Make sure your software is configured so that the audio interface or soundcard is selected. You can choose input and output settings in Audacity by selecting the Audio I/O tab in the Preferences menu (see figure 5.1).

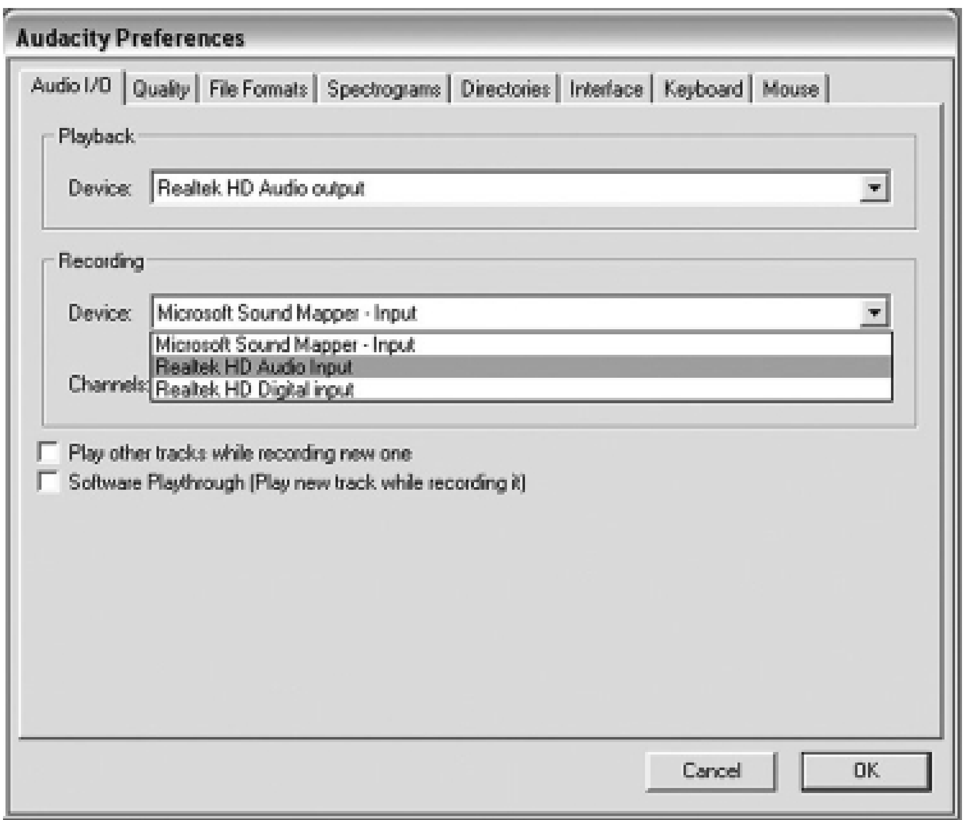


FIGURE 5.1

Selecting audio I/O devices in Audacity

You will want the hottest possible input signal without distortion in order to maximize the signal. Note that you may need to adjust the input trim of your microphone in any number of ways. If you are using a soundcard, use the Sound Speech and Audio Devices utility in Windows or the Audio MIDI Setup utility on a Mac. An external audio interface will provide microphone trim knobs and/or a software utility that can be used to adjust input levels.

Notice how the input level is visible in the top-right corner of the Audacity window (see figure 5.2). Click the dropdown arrow to select either vertical or horizontal modes.

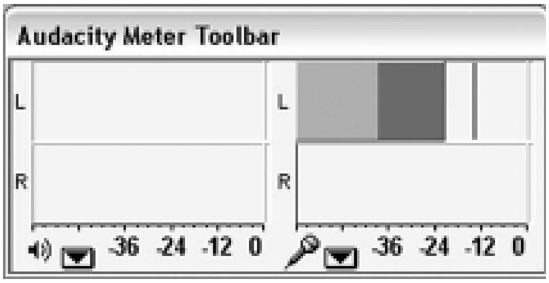


FIGURE 5.2

Viewing input level in Audacity

Recording

You will want to familiarize yourself with several helpful key commands prior to recording your podcast. Select the Preferences menu and then click the Keyboard tab to see all of the key commands or to assign your own. One quirk of Audacity is that it will create a new track each time you select the record button. I typically make frequent stops and starts when recording a podcast so I find it works well to use the "R" key to record and the "P" key to pause. Using the pause option will force Audacity to continue the recording on the current track.

Markers

I like to make frequent use of markers as it speeds up the editing process. Press Control + M (PC) or Command + M (Mac) to insert a marker. Note that this can be done "on the fly" as you record so it's an easy way to keep track of takes and your position in a script or outline. You can even type marker text without exiting Record mode in Audacity.

Editing

Audio editing is a straightforward process in Audacity. On the top left of the Audacity window you will see six tools that facilitate several common tasks (see figure 5.3).

FIGURE 5.3

Editing tools

Selection Tool

Use the selection tool (see figure 5.4) to select a region and press the delete key to delete a wrong word or to trim extra space from a segment of your podcast. You can also apply effects such as echo or fade-ins and fade-outs by selecting a region and selecting an option from the Effects menu.

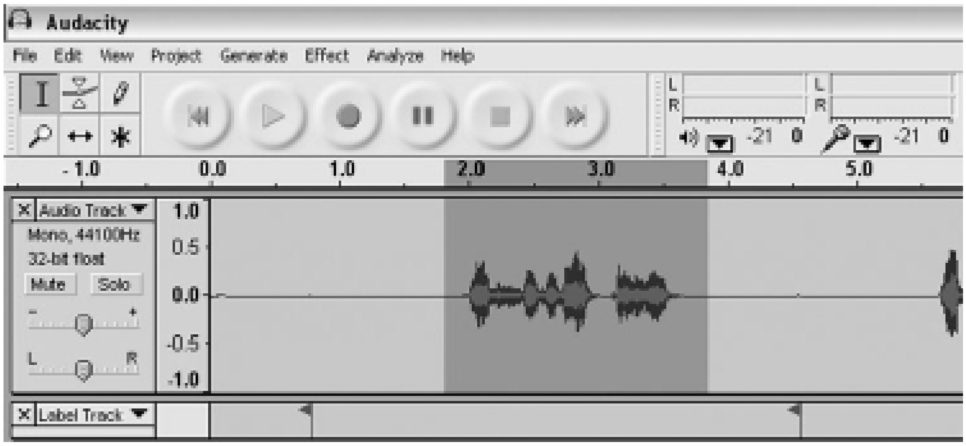


FIGURE 5.4
Using the selection tool

Envelope Tool

The envelope tool (figure 5.5) is used to adjust the level of a region. This tool is a bit different from most DAW applications in that you click with the mouse and drag up or down to adjust the level. Note that this is a primitive form of automation so, if you decrease the level in one place, the new level will remain in effect until you insert another control point.

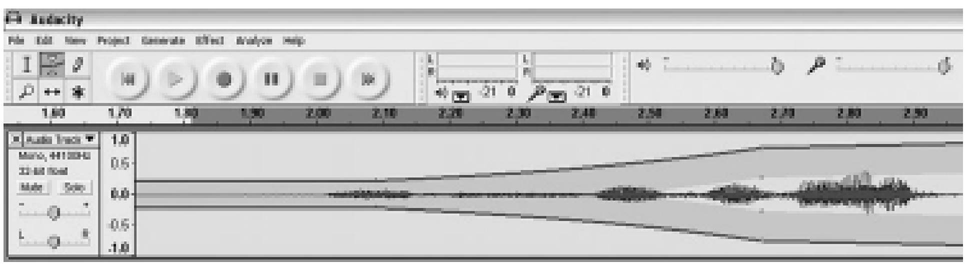


FIGURE 5.5
Adding a fade-in with the envelope tool

Draw Tool

The draw tool is used to fix anomalies such as a click or pop. Zoom in to the sample level and "draw" over any problem samples, as shown in figure 5.6.

Zoom Tool

Use the zoom tool to zoom in or out on the X-axis. The "zoom to normal" command will zoom in or out in order to fit the entire clip within the Audacity window.

Time Shift Tool

The time shift tool allows a clip to be moved to the left or right on the timeline. For example, you might want to drag a voice-over to the left or right to better fit with background music or sound effects (see figure 5.7).

Multi-Tool Mode

The multi-tool mode tool can be used to zoom, edit samples, adjust envelopes, or select a region. The function of the tool depends on the position of the cursor.

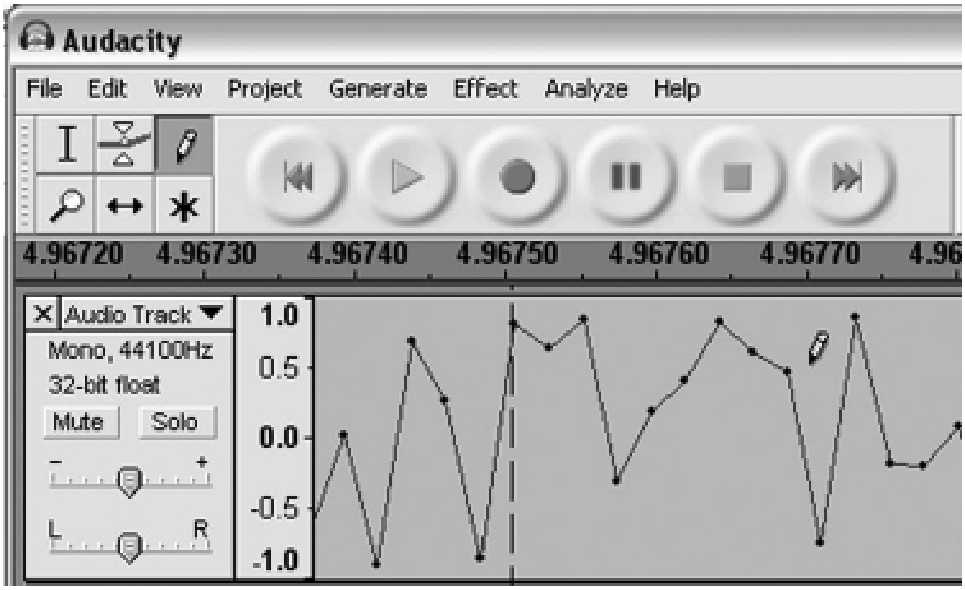


FIGURE 5.6

Sample editing (maximum zoom)

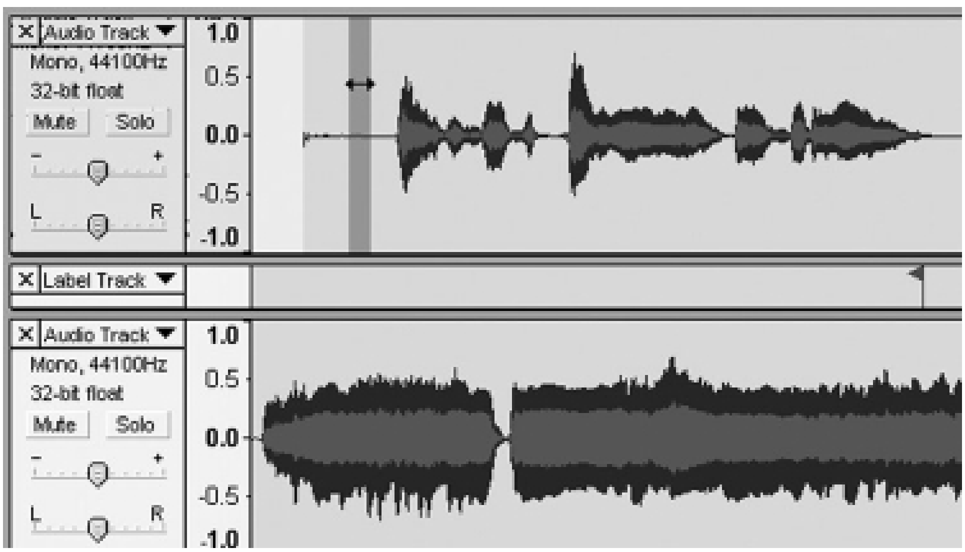


FIGURE 5.7

Adjusting the position of a voice-over

The key commands in table 5.1 can be used to easily perform most edits in Audacity.

Enhancing a Podcast

Once you have finished recording and editing your podcast, it's time to "sweeten" your work. We will start with several audio enhancements.

Noise Removal

Audacity provides a helpful way to remove noise from a recording. This function would be useful if your recording was created in an environment with background noise, such as a fan or refrigerator. First, use the selection tool to select a region of background noise with no talking. Then, select the Effect menu and choose Noise Removal. Click the Get Noise Profile button in the Noise Removal

TABLE 5.1 Audacity tool commands

Key	Function
F1	Selection Tool
F2	Envelope Tool
F3	Draw Tool
F4	Zoom Tool
F5	Time Shift Tool
Command* + 1	Zoom in
Command + 2	Normal zoom
Command + 3	Zoom out

*Use the Command key on a Mac and the Control key on a PC.

dialog to establish the noise signature. Return to the dialog and select Remove Noise, and Audacity will remove any noise that matches the noise signature.

Silence

The silence tool is useful if you want to remove a portion of audio without affecting the position of the surrounding material. For example, you might select a blank region of a voice-over track and use the Silence option to quiet any background noise during a musical intro. Unlike deleting a region, the surrounding material does not move.

EQ

Equalization can be useful for any number of reasons. You might want to enhance lower frequencies to create a full-bodied voice-over or emphasize frequencies in the speaking range in order to make the voice-over cut through background music. EQ can also be useful for toning down some kinds of noise such as low-frequency rumble. To add equalization, select the Equalization option from the Effect menu and click to insert control points in order to draw your intended EQ curve.

Compression

Compression (see figure 5.8) and equalization will be covered in detail in chapter 10. In short, compression can be useful in a podcast by minimizing the variance between soft and loud words. For example, a word that is quietly spoken can sound similar in level to a word that is spoken in a normal speaking voice when compression is used. In general, it is a good idea to apply a liberal amount of compression to podcasts since listeners are likely to hear your podcast in a noisy environment such as a car or at the gym. Compression can help to even loud and soft sections so that dynamics are more consistent. This will enable the listener to hear words that might otherwise be hard to hear while maintaining a comfortable listening level.

Music

Music can add a extra element of interest and professionalism to your podcast. You can import music via the Project . . . Import Audio menu. This will create an additional stereo or mono track.

A volume and pan slider are available on each track. Unlike the envelope tool, adjustments to the volume slider are global. Use this tool to adjust the overall level of your music or voice-over tracks, and use the envelope tool to pull the

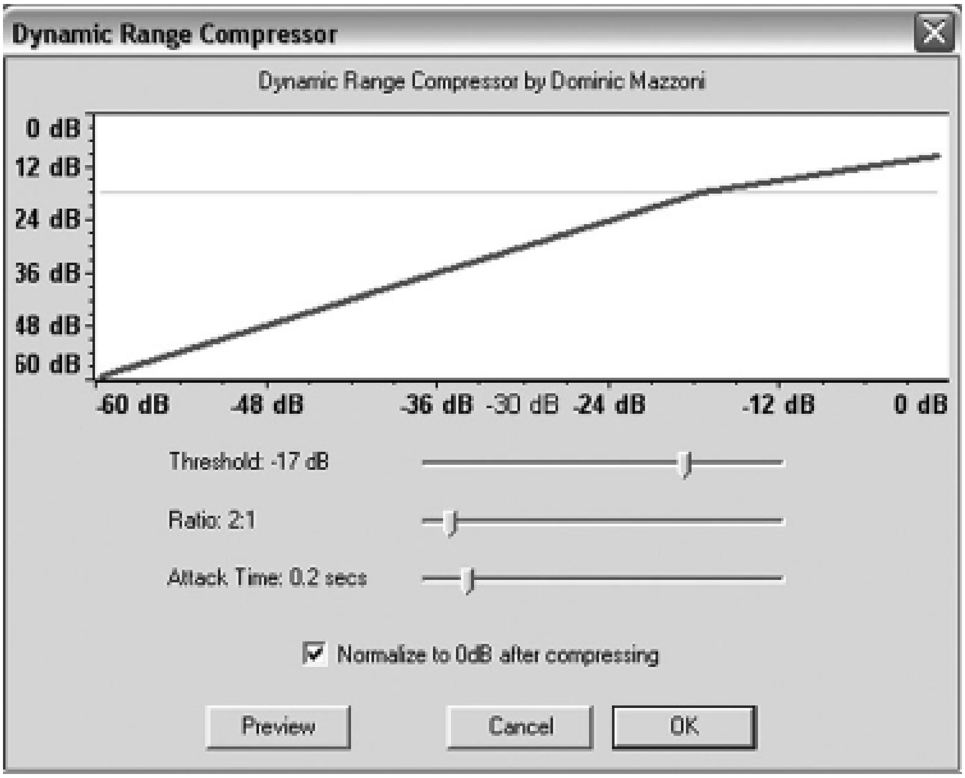


FIGURE 5.8

Applying compression to a podcast

level of the music down under a section of voice-over or bring it up at the end of the podcast. You will likely want to leave a voice-over in the center pan position, but for an interview, it might make sense to pan each voice to opposite sides.

Normalization

It is a good idea to normalize your file when your podcast is complete. The normalization process will analyze the file to look for the highest amplitude in the recording. If the highest amplitude isn't already at 0 dB, the normalization algorithm will calculate the difference and move the level of all samples up a proportional amount. Where compression is used to "squash" dynamic range, normalization is used to maximize the overall volume so that the podcast is at its maximum possible level.

Bouncing to Disk

Once your podcast is complete, it is time to bounce it to disk. This would be a good time to point out the difference between saving a project and bouncing to disk. You will want to frequently save your project file, as the project contains all of your audio tracks, settings, and edits. In contrast, the bouncing process renders all tracks and settings to a single stereo or mono audio file.

Audacity and MP3 Files

Raw Wave (.wav) files are large so you will likely want to export your completed podcast as an MP3 (.mp3) file if you intend to make the file available online.

Unfortunately, Audacity doesn't support MP3 files unless you take several steps when you install the program. Although Audacity doesn't natively support MP3, you can download and install a plug-in called the LAME MP3 encoder when you download Audacity. (A plug-in is a special type of software that provides additional functionality to a host application.) Installation is simple but make a note of the directory where you install the Lame encoder, as Audacity will ask you to find the library the first time you attempt to bounce an MP3 file.

You can also use the Lame encoder to change the default bit rate prior to bouncing to disk. Open the Audacity Preferences and choose the File Formats tab, and you will see an option to adjust the bit rate for MP3 export. You will want to experiment with this setting in order to find the best balance between file size and audio fidelity for your podcast.

FIGURE 5.9

ID3 settings

Before your MP3 is saved to disk you will be presented with the dialog box shown in figure 5.9. The ID 3 settings are tags that are embedded in an MP3 file that allow users to see the source of the podcast in Windows Media Player, iTunes, and most MP3 players. You will typically want to include a title and your name and/or Web site so that the source of the podcast is visible when someone plays your program. You can find additional information regarding data tagging at the following Web site: http://www.id3.org.

Publishing Your Podcast

Once your podcast is complete, it is time to make it available to potential listeners. The easiest approach is to provide a link to the file from a Web page, just as you would with any other Web media file. The following is an HTML code snippet for an audio link that was generated by Dreamweaver, Adobe's popular Web authoring tool:

<p>My link</p>

However, most podcasters will want to make their podcasts available via a service such as Apple's iTunes. As you will see in the paragraphs that follow, several steps are required in order to make your podcast available on iTunes.

Make the File Available on the Web

The first step is to make the podcast available on the Web with a publicly accessible *uniform resource locator* (URL) address. One way to do this is through a Web-hosting company. Most hosting companies offer the ability to upload podcasts, and you should receive a URL to the podcast after completing the upload process. The process will obviously vary depending on the hosting company, but typically, the hosting service will provide a Web-based tool for uploading podcast and other types of files. Another option is to use File Transfer Protocol (FTP) to move podcasts from your computer to a Web server. FTP is a component of some authoring tools including Dreamweaver, and many shareware FTP programs are available online.

Create an RSS Feed

A second step is to create an RSS feed. An RSS feed is simply a text file in *extensible markup language* (XML). An XML file consists of extensible tags in a "human readable" format that can be used for any number of things, such as Web formatting or data storage. In the case of podcasting, XML files define the tags that point to the URL of a podcast and provide information about the author, a description of the podcast, and so on. The details of RSS creation are beyond the scope of this book, but in most cases, an RSS feed should be created when you post your podcast to your Web-hosting company. Another option is to set up an account at www.feedburner.com, a service that facilitates the management and dissemination of podcasts. You can also create an RSS feed using any plain text editor.

Yet another option is to use a program like the Podcast RSS Buddy to create your RSS feed. RSS Buddy is a very intuitive program and will create an iTunes-compatible RSS feed in a matter of minutes. Figure 5.10 consists of an RSS feed that was based on a sample provided by Apple. The feed was created with a plain text editor and demonstrates the essential RSS tags, as well as several tags that are unique to iTunes. Note that I use bold typeface to make it easier to see the data that go with each of the most important tags. Although the lack of formating might be awkward, XML files are fairly easy to read. Most of the tags in this example are one-line tags. You will see a starting tag such as <title> followed by some text and then a matching closing tag: </title>. (Note how a slash is used to differentiate between starting and ending tags.) Some tags such as the <channel> tag are multiline—almost the entire feed is found between the <channel> tags. Similarly, the <item> tag is a multiline tag and refers to one episode of the given channel.

Table 5.2 lists the common podcast tags, including those that are unique to iTunes.[1] The table, as well as additional information regarding podcasting feeds, is available at http://www.apple.com/itunes/whatson/podcasts/specs.html.

Don't worry if the preceding text is confusing. You will rarely need to edit XML text by hand unless you want to add specific tags for iTunes that may not be supported by your RSS generator.

```
<?xml version="1.0" encoding="UTF-8"?>
<rss xmlns:itunes="http://www.itunes.com/dtds/podcast-1.0.dtd"
   version="2.0">

<channel>
<title>Podcast title</title>
<link>http://www.yourweb.com</link>
<language>en-us</language>
<copyright>&#x2117; &amp; &#xA9; 2010 Your name</copyright>
<itunes:subtitle>Subtitle</itunes:subtitle>
<itunes:author>Name of author</itunes:author>
<itunes:summary>Long podcast summary</itunes:summary>
<description>Description of podcast</description>
<itunes:owner>
<itunes:name>iTunes specific info: contact name </itunes:name>
<itunes:email>iTunes specific info: email</itunes:email>
</itunes:owner>
<itunes:image href="http://www.yourweb.com/image.jpg"/>
<itunes:category text="Music" />
<!--Comment: see Apple documentation for setting up multiple categories and
   sub categories  -->

<item>
<title>Name of an episode</title>
<itunes:author>Author of episode</itunes:author>
<itunes:subtitle>Subtitle of episode</itunes:subtitle>
<itunes:summary>Summary of episode</itunes:summary>
<enclosure url="http://www.yourweb.com/songname.mp3"
   length="7464204" type="audio/mpeg" />
<!-- Optional <guid> tag can go here. Episode URL will be used if no globally
   unique identifier is used.  -->
<pubDate>Tues, 22 Jun 2010 16:10:00 GMT</pubDate>
<itunes:duration>7:04</itunes:duration>
<itunes:keywords>keyword1, keyword2, keyword3 </itunes:keywords>
</item>

<item>
<title>Name of another episode</title>
<itunes:author>Author of another episode</itunes:author>
<itunes:subtitle>Subtitle of another episode</itunes:subtitle>
<itunes:summary>Summary of another episode</itunes:summary>
<enclosure url="http://www.yourweb.com/anothersongname.mp3"
   length="7464204" type="audio/mpeg" />
<!-- Optional <guid> tag can go here. Episode URL will be used if no globally
   unique identifier is used.  -->
<pubDate>Tues, 22 Jun 2010 16:10:00 GMT</pubDate>
<itunes:duration>7:04</itunes:duration>
<itunes:keywords>keyword1, keyword2, keyword3 </itunes:keywords>
</item>

</channel>
</rss>
```

FIGURE 5.10

Example of an RSS feed

TABLE 5.2 Common podcasting tags

xml tag	channel	item	where content appears in iTunes
<title>	Y	Y	Name column
<link>	Y		Web site link and arrow in Name column
<copyright>	Y		not visible
<pubDate>		Y	Release Date column
<itunes:author>	Y	Y	Artist column
<itunes:block>	Y	Y	prevent an episode or podcast from appearing
<itunes:category>	Y		Category column and in iTunes Music Store Browse
<itunes:image>	Y		same location as album art
<itunes:duration>		Y	Time column
<itunes:explicit>	Y	Y	parental advisory graphic in Name column
<itunes:keywords>	Y	Y	not visible but can be searched
<itunes:new-feed-url>	Y		not visible, used to inform iTunes of new feed URL location
<itunes:owner>	Y		not visible, used for contact only
<itunes:subtitle>	Y	Y	Description column
<itunes:summary>	Y	Y	when the "circled i" in Description column is clicked

Testing Your RSS Feed

It is important to test your feed to ensure that everything is functioning properly before trying to submit the podcast to iTunes. Apple offers the following suggestions for testing your RSS feed:

1. Launch iTunes.
2. In the Advanced menu, select Subscribe to Podcast.
3. Enter your feed URL in the text box and click OK.

Watch the small download circle to see if the podcast downloads to your computer and click the play button once the podcast has successfully downloaded in order to check the podcast. Common problems include incomplete or inaccurate links (e.g., use http://www.yoursite.com *not* yoursite.com), missing tags, and unmatched starting or ending tags.

Submitting Your Podcast to iTunes

The following steps are based on Apple's podcast submission documentation:

1. Launch iTunes.
2. Select the iTunes Store and click on the Podcasts submenu.

3. Click on the Submit a Podcast link from the Quick Links sub window.
4. Enter the URL address of your podcast. You will receive a confirmation e-mail in a day or two.

Video Podcasts

As an educator, I have found that video podcasts are particularly useful in teaching software applications like Sibelius and Logic, as it allows my students to view the operation of the software and work through tutorials at their own speed. Although video podcasts are beyond the scope of this book, the process of creating a video podcast is very similar to creating an audio podcast. Numerous screen-capture applications are available for Mac and PC to turn your onscreen actions into a standard video file that can then be imported into programs like Garage Band and combined with a voice-over and music.

Conclusion

Podcasts are easy to create and can be a great way to promote your music or share information or creative content with people around the world. Podcasts can be served on a Web site or intranet, and many educators are electing to post podcast content to iTunes University. With the popularity of services such as iTunes, your podcasts can reach a large audience so it makes sense to consider podcasts for their educational and promotional value. I would encourage you to spend some time listening to podcasts. There are some terrific podcasts available on a wide range of subjects, and you can learn a great deal about the organization and production of your podcast content by listening to these broadcasts.

6

Using a Mixer

Where cables are the veins and arteries of a recording studio, the mixing console, whether hardware or software, is the heart of a studio. A mixing console (also called a sound board, audio mixer, or just mixer) is used to mix a variety of signals and to adjust the relative level and pan position of those signals. Although high-end consoles may have dozens of knobs and faders, and look frighteningly complex, the signal flow is basically the same for a simple four-channel mixer as with its high-end cousin. This would be a good time to emphasize the fact that knowledge of mixers and signal flow will directly apply to the software-based consoles found in Digital Audio Workstations.

Mixers can be a source of confusion to people who are new to recording, but they are relatively easy to use once you learn some basic concepts. This chapter will provide a gentle introduction to many of the terms and concepts associated with modern-day recording consoles. You will learn the basics of signal flow, how to operate a console, and how to record a clean signal. We will explore the recording and mixing process in greater detail in the chapters on tracking, mixing, and mastering.

Top-to-Bottom Signal Flow

Mixers can be deceptively simple to operate if you remember that signals flow either from top to bottom or left to right. For example, when a microphone or other sound source is plugged into a channel, the signal flows through several stages from the top of the mixer to the channel fader at the bottom. As you will see in a moment, the term *bus* is used to describe signals that flow from left to right.

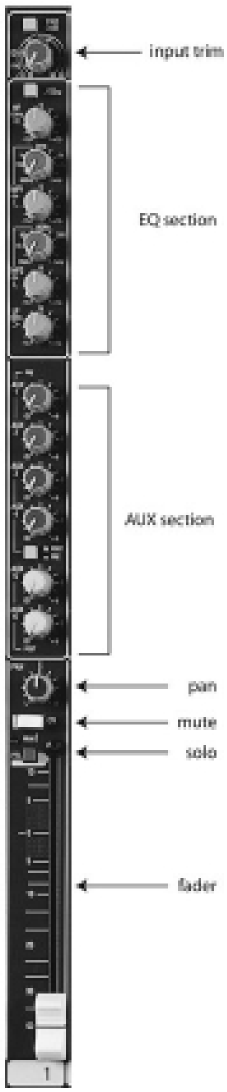


FIGURE 6.1
Typical channel strip

Channel Strip

The term *channel strip* refers to the components that make up a single channel on a given mixer. A basic channel strip will have an input section consisting of a preamplifier (used to boost or attenuate signals), one or more auxiliary sends (more on those in a moment), several EQ knobs, a knob to pan the signal from side to side, buttons to solo and mute the channel, and a fader that is used to mix the signal into the main bus (see figure 6.1).

Preamp

At the top of the signal chain, a microphone or other source is plugged into the channel preamp. A given channel may provide any number of input *jacks* (jacks receive a plug), such as an XLR , ¼-inch, or RCA. Some consoles provide both XLR and ¼-inch jacks on each channel, which offers better flexibility in terms of connectivity (see figure 6.2). The XLR jacks are used to connect microphones and other balanced sources. The ¼-inch jacks may be either balanced or unbalanced and may accept the output of a keyboard or other instrument. Some consoles even provide one or more high-impedance ("high z") inputs for connecting instruments such as an electric guitar or bass without the need for a direct box.

In evaluating a mixing console for purchase, it particularly important to assess the sonic quality of the unit's preamps. Preamps can impart a unique color to a signal that may be beneficial, neutral, or detrimental to the sound. As evidence of the importance of preamps, it is interesting to note that high-end studios purchase dedicated preamps that cost thousands of dollars in order to impart a special sonic character or color to key tracks such as voice or solo instrument.

A *gain* or *trim* knob will usually be located at the top of the channel strip. Its role, which we will explore in greater detail in the "Recording" section of this chapter, is to adjust the level of an incoming signal so that the signal matches the levels of other channels when the channel fader is set to the *unity gain* or "default" position.

Phantom Power

A *phantom power* switch is often provided in the input section of a mixing console (sometimes a single switch will provide phantom power for several channels). Phantom power is used to power condenser microphones and active

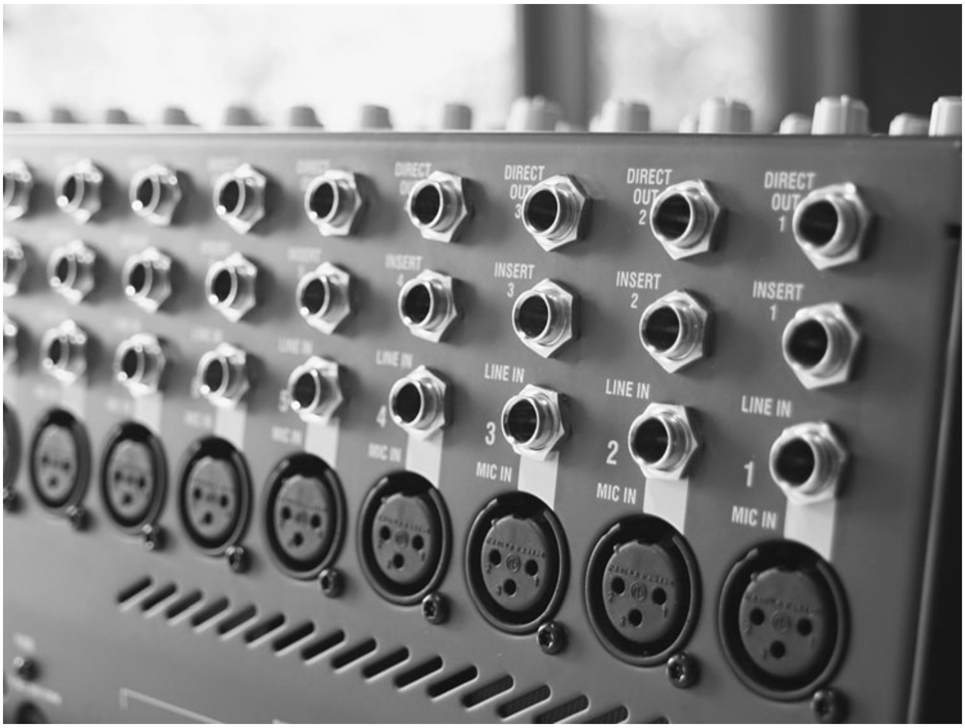


FIGURE 6.2

Input section of a mixing console (XLR and ¼-inch inputs)

direct boxes. When phantom power is engaged, a small amount of electricity flows down a connected microphone cable, providing the necessary power. Often, the symbol 48 v is used instead of "phantom power," since 48 volts is the amount of power provided.

One word of warning: take care when applying phantom power since it may cause a loud pop if the channel is not muted and the mains are up. It is also important to avoid using phantom power with a ribbon microphone since the microphone may be damaged by the application of phantom power.

Channel Insert

A *channel insert* (sometimes called an insertion point) provides a way to tap a channel for external processing by a compressor, noise gate, or similar device. Although channel inserts consist of a single jack, it is interesting to note that the jack is both an input and an output. This is done through a special stereo cable called a *Y cable*. One end of the cable consists of a single ¼-inch tip-ring-sleeve (TRS) plug that connects to the channel insert. The other end consists of two ¼-inch (tip-sleeve) TS plugs that are connected to the input and output of the given processing device. In this way, a single channel insert jack can send and receive a signal. Channel inserts may also be used as a form of *direct out* (see below).

Direct Out

As the name implies, a direct output provides access to an incoming channel signal. Direct outs are frequently connected to the inputs of a hardware recorder or digital audio interface and are typically used to tap the sound just after the signal passes through the channel preamp. On some soundboards, the signal might be sent post-EQ or even post-fader. Some consoles provide a toggle switch to select pre- or post-fader.

Pre-fader

The term *pre-fader* is often associated with auxiliary sends and direct outs. Simply put, a pre-fader signal is sent before the signal reaches the fader at the bottom of the channel strip. The pre-fader setting would be useful when providing a monitor or headphone mix since the musicians would want to hear a consistent level that is unaffected by changes made to the channel fader.

Post-fader

The post-fader setting is useful when you want changes to the channel fader to be reflected in an attached device, such as when mixing multiple tracks down to a stereo master.

Equalization

The equalization section of a channel strip is used to alter the timbre or color of a signal by boosting or attenuating certain frequencies. Typically, at least three knobs are provided to adjust low, middle, and high frequencies. Some consoles also provide additional knobs to set a center frequency and equalization width, or "Q," for each equalization knob. We will explore equalization in more detail in chapter 10. For now, suffice it to say that some tracks may need equalization, but it is rarely a good idea to commit to those changes during the recording process.

Pan

A pan knob is usually found just prior to the channel fader. The pan knob provides a means for adjusting the position of the signal to left or right in the stereo field. Panning is often used to provide some breathing space between signals that share a similar range of frequencies. For example, it often makes sense to pan electric guitar to one side and acoustic guitar to another since these instruments share the same range of frequencies.

Mute/Solo

Mute and solo buttons are often provided on each channel of an audio mixer. These buttons are particularly helpful during mixing and recording. For example, when checking the sound of an instrument, use the solo button to mute the other channels and focus on the channel in question. Mute and solo buttons can also be a source of frustration: it's easy to overlook the fact that a mute or solo button is engaged and wonder why certain channels aren't working.

Channel Fader

The channel fader is usually the last item on a channel strip. It is used to adjust the amount of signal sent to the main bus or a sub bus. Note that small faders may be awkward to use since small hand gestures may have a big impact on the level of the channel.

Left-to-Right Signal Flow: Busing

Up to this point, we have focused on the concept of a channel strip—the flow of an incoming signal from the top of the mixer to the bottom. In this section, we will consider the concept of *busing*, which refers to the flow of an audio signal from left to right.

Bus

There are several terms associated with busing, such as *main bus*, *sub bus*, and *auxiliary send/return*. Each of these terms describes a very simple concept: signals are combined from left to right and sent to some destination. For example, the faders at the bottom of a mixer are used to adjust the relative level of each channel. The channels may, in turn, be added to the main bus, which is used to drive the main out, control-room speakers, two-track recorder, or similar. Buses are wonderfully flexible. You can use them to create a monitor mix, combine groups of signals for external processing, control the overall volume of a group of signals with a single knob or fader, and many other tasks.

Sub Bus/Subgroup

A *sub bus* or *subgroup* is a special type of bus that is very useful during the mixing process. Consider a recording of a trio consisting of acoustic bass, piano, and drums. Many engineers would record the piano in stereo, use a microphone and/or direct insert on the bass, and allocate four or five microphones for the drums. Instead of riding eight or nine faders, the engineer may elect to group the signals in a convenient way and send the signals to several sub buses. For example, the engineer could use channel faders to balance the drums and assign each drum channel to sub bus 1. Now, a single fader (sub bus 1) can be used to adjust the level of the entire drum kit. A similar process could be used to assign the bass channels to sub bus 2 and the stereo piano to sub bus 3. At this point, the engineer could use three faders (sub bus 1–3) to control the relative level of each instrument. Clearly, the concept of using a sub bus or subgroup is attractive when working with dozens of tracks. Notice how, in figure 6.3, subgroups are used to combine the various "choirs" of a jazz orchestra in Logic Pro.

Auxiliary Sends/Returns

Auxiliary sends are another type of bus commonly used in recording and mixing. Most consoles will provide several "aux" knobs on each channel strip, which

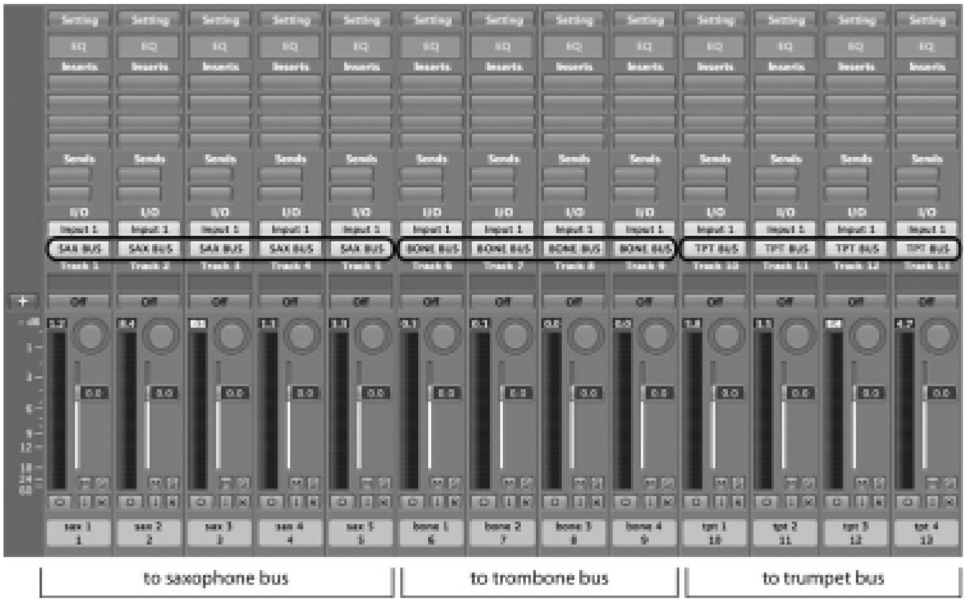


FIGURE 6.3

Subgroups of a jazz band in Logic Pro

can be used to send a signal to a processing unit such as a reverb or delay or for some other purpose such as to provide a headphone or monitor mix.

By way of example, consider a stereo recording of a piano where the left channel is assigned to channel 1 and the right to channel 2. Both the left and right channels can be sent to an external processing unit by turning up aux send 1 on channels 1 and 2. The output of the auxiliary send is attached to the input of an external device such as a reverb unit and the processed signal returns (often in stereo) to the mixer via the *auxiliary return*. The auxiliary return is similar to a channel strip in that its signal can be blended with the other channel and auxiliary return signals. In fact, some engineers may elect to plug the output of a reverb or delay into the input of another channel strip instead of using an auxiliary return. This approach provides an opportunity to use EQ and pan on the processed signal. Figure 6.4 provides a conceptualization of the flow of a signal from an auxiliary send to an external device and back via the auxiliary return.

Operating a Mixing Console

To review, a mixing console consists of incoming signals (channel strips and auxiliary returns) that flow from the top of the board to the bottom. Signals are mixed via a busing architecture, the most common of which is the main bus. Signals flow from left to right on a bus and varying amounts of signal can be added from each channel to a bus such as the main bus or sub bus. Inserts may be utilized to process a signal with an external compressor or similar device. Direct outs (sometimes a channel insert) may be used to send an incoming channel signal to an external recorder. Auxiliary sends are typically used to process a signal with an external reverb or delay but may be used for any number of applications. Figure 6.5 conceptualizes the signal flow of each of these components.

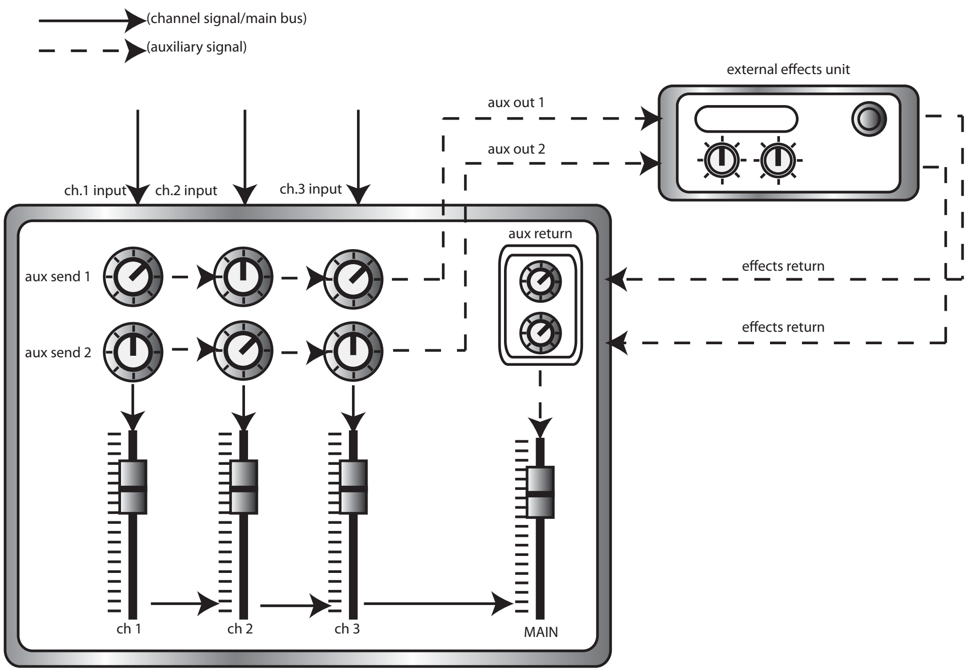


FIGURE 6.4

Signal flow of an auxiliary send and return

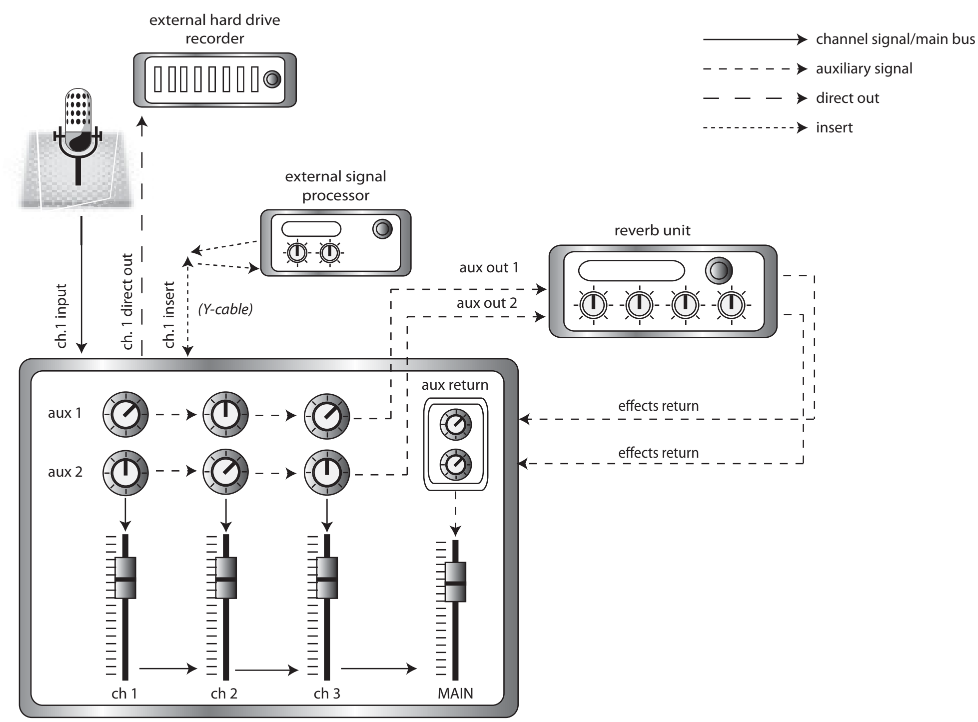


FIGURE 6.5

Conceptualization of the primary signal flow of a mixing console

When using or troubleshooting a console, I find it helpful to remember the concept of top-to-bottom and left-to-right. Start at the top: Is a signal actually reaching the board? If so, is it reaching the fader or is the given channel muted (or is a solo button engaged on another track)? If a signal is reaching the channel but not an attached recording device, check that the device is connected to the appropriate output, such as the direct out. Also check if the output is set to pre or post fader since no signal will reach the recording device if the fader is down and the direct out is set to post fader.

If you see that a signal is reaching a given channel but you can't hear the channel, check to make sure the channel isn't muted and then focus on the left-to-right signal flow of the buses: Is the given channel assigned to the main bus or subgroup? Is the main bus or subgroup actually assigned to the main out or control-room speakers? Are the main outs or control room outputs actually connected to an amplifier or powered speakers? Is the amplifier or powered speakers turned on? Following the top-to-bottom and left-to-right signal flow will enable you to troubleshoot and feel comfortable with a mixing console.

Recording

Chapters 9, 10, and 11 provide a comprehensive overview of the recording and mixing process. In this section, we will focus our attention on terms and concepts associated with a mixing console in a recording environment.

Making Connections: Balanced Signals

Unbalanced cables can be a source of hum and other problems in a recording environment so it is important to use balanced cables and equipment when possible. A balanced cable, such as an XLR microphone cable or ¼-inch TRS, has two wires surrounded by a grounded shield. One of the wires carries the signal and the other carries an inverted copy of the signal. At its destination, the inverted signal is flipped and summed with the other signal, which has the effect of canceling noise that might have entered the cable. Note that a conceptualization of signal balancing and photos of XLR and TRS plugs can be found in chapter 1.

When I first started exploring MIDI and audio recording, I tended to purchase less expensive consumer gear that did not provide balanced connections, and I was always frustrated by audio gremlins, such as unwanted noise and 60 cycle hum. As I learned more about recording, I made a conscious decision to purchase and use "pro" balanced connections whenever possible, and it has made a huge difference. I don't experience problems with noise or hum now that I have been more conscious of using balanced equipment and cables. Note that the final chapter of the book provides a tutorial on how to make quality balanced cables for a reasonable cost.

Using a Direct Box

As mentioned in chapter 1, a direct injection (sometimes called a direct insert, direct box, or simply DI) is used to convert high-impedance signals such as the output of an electric bass guitar to a low-impedance balanced signal that can then be plugged into a mixing console. A direct box allows for long cable runs and minimizes problems with sound coloration and distortion that can occur owing to a difference in impedance.[1] Using a direct box is functionally similar to using a microphone: plug the output of a guitar, bass, or similar instrument into the input of the direct box and connect the output of the direct box to a channel input on the console. Active direct boxes are similar to condenser microphones in that they require the application of phantom power.

Connecting to a Hardware Recorder

There are several common scenarios used to connect a mixer to a recording device. One option is to devote half the channel strips to the inputs you intend to record and use the remaining channels to monitor outputs from the recorder. In this scenario, incoming signals are routed to the recorder via direct out jacks on each channel. This approach is attractive because it is easy to switch between monitoring incoming signals and the output of the recorder. Some consoles provide an input toggle switch, which provides the ability to switch between two input signals (usually XLR and ¼ inch) on the same channel strip. In this scenario, incoming signals are again routed to the recorder via the direct out jack, but the engineer must select the toggle to switch between sources. One disadvantage of this system is that it's easy to create a feedback loop should you forget to disarm a given track for recording when you switch input sources.

Unity Gain

It is helpful to consider the concept of unity gain when adjusting incoming signals. *Unity gain* means that a signal is not being amplified or attenuated so, conceptually, you want to adjust the preamp such that the signal is at an optimal level when the fader is set to its "zeroed out" or unity position. Most mixers have a unity marking about a quarter of the way down from the top of the fader. Set the fader to its unity position prior to adjusting the trim knob on the preamp and check that the attached device such as a keyboard or preamplifier is set to its optimal output level (e.g., the output of a keyboard should be set at or near the maximum). This process will help to ensure that signals are at an optimal level for recording.

Establishing a Good Level

Once the cables have been plugged into the board, it is time to check each sound source and adjust the trim on each channel. Although the process can be time-consuming, it is essential to determine that the incoming signals sound good and are at appropriate levels. I would suggest soloing each track (or pair of stereo tracks) in order to focus on the instrument or voice at hand. Microphone placement (which is covered in chapter 4) can have a huge impact on the sound, so carefully consider if a change of microphone or its position is in order. For a source such as a keyboard, make sure the volume is turned up on the instrument but not so much that it overdrives the input to the board, which can happen with some keyboard instruments.

One you are pleased with the sound on a given channel, focus on optimizing the gain (or trim). Ask the instrumentalist or vocalist to play a wide dynamic range from very soft to very loud. The idea is that you want the *hottest possible* *signal without overdriving or causing distortion*. Depending on your setup, you may also need to adjust the input level on the corresponding channel of a digital audio interface or external recorder if one is attached to the output of the current channel.

Signal-to-Noise Ratio

The concept of *signal-to-noise ratio* is easy to visualize. Noise will be a component of any recording, even when appropriate precautions such as balanced cables are used. The electronics of each device in the signal chain will impart a small amount of noise, and noise might also come from an external source such as a heating or cooling system in a live venue. We obviously want to minimize the amount of noise in a recording, and this is where the concept of signal-to-noise ratio comes into play. If you record a weak signal, the ratio between the noise "floor" and the signal is relatively small. If you turn up the level of such a track, the noise increases in a proportional amount to the signal. On the other hand, when recording a hot signal, the ratio between the signal and the noise floor is increased and noise is effectively masked by the signal. Simply put, inherent noise should be inaudible when listening to a track that has been recorded with a high signal-to-noise ratio. Figure 6.6 illustrates the concept of signal-to-noise ratio.

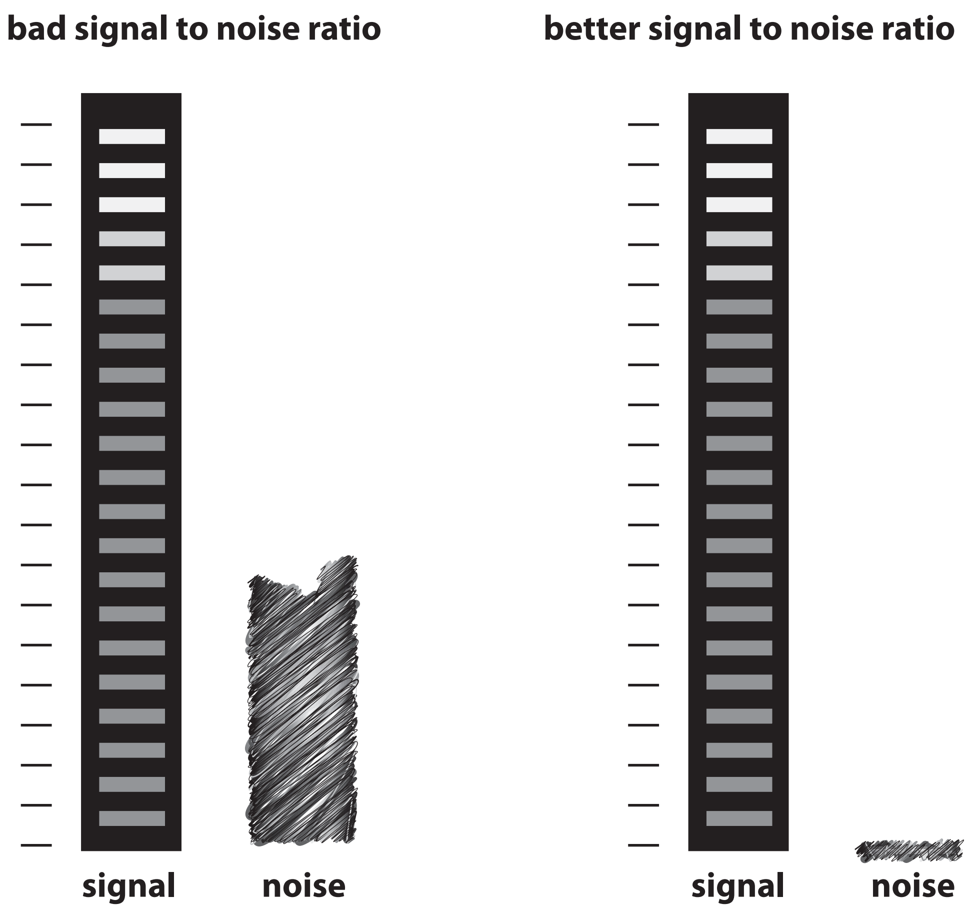


FIGURE 6.6

Signal-to-noise ratio

Practical Applications

In this section we will look at several common mixing and recording scenarios: live to two-track recording, multitrack recording, recording with an audio interface, and setting up a headphone or monitor mix. Note that detailed tips and techniques associated with tracking mixing and mastering will be presented in chapters 9 and 11.

Live to Two-track

Two-track recording can be as simple as running two microphones (or one stereo microphone) into a two-track recorder, or it might involve mixing any number of inputs to a stereo output. For example, five or more channels might be utilized to record a piano trio:

- Piano: two condenser microphones (small or large diaphragm)
- Bass: one condenser or dynamic microphone and/or optional DI
- Drums: two condenser overhead microphones with optional microphones for snare and bass drum.

Engineers sometimes use a microphone and a direct line (if one is available) when recording acoustic bass so an extra channel could be devoted to a direct box that connects to the bass pickup. Optionally, an external compressor could be attached to the channel insert on the bass since a touch of compression is often helpful when recording bass.

An auxiliary send and return could be utilized to route the piano and overhead drum tracks to an external reverb unit in order to add some spaciousness to the recording. The processed signal returns to the board via an auxiliary return and is mixed with the main bus via the auxiliary return knob.

Finally, the output of the mixer is attached to a two-track recorder. In many instances, the tape out (an unbalanced RCA connection) is attached to an external CD burner or other portable recording device.

Note that multiple microphones are not always the best choice and a good recording can be made with just two microphones in a stereo configuration. The biggest disadvantage of using multiple microphones when recording directly to stereo is that there are no second chances in terms of the mix since any level imbalances or other problems are recorded directly to the two-track master.

Multitrack Recording

In this example, a twenty-four channel mixing console is used to record eight simultaneous tracks (see figure 6.7). As with the two-track scenario, a variety of sources are plugged into the first sixteen channels of the mixing board. Direct outputs can be used to connect the console to a multitrack recorder or audio interface, but in this instance, a combination of direct outputs and sub bus outputs would be appropriate. For example, it would make sense to utilize a sub bus since multiple microphones are used to record a drum set but only eight inputs are available on the recording device.

For playback, the outputs of the multitrack recorder are plugged into the inputs of channels seventeen through twenty four. This setup provides a separate set of channels that can be used to mix, EQ, pan, and otherwise process the output from the recorder for mixing purposes.

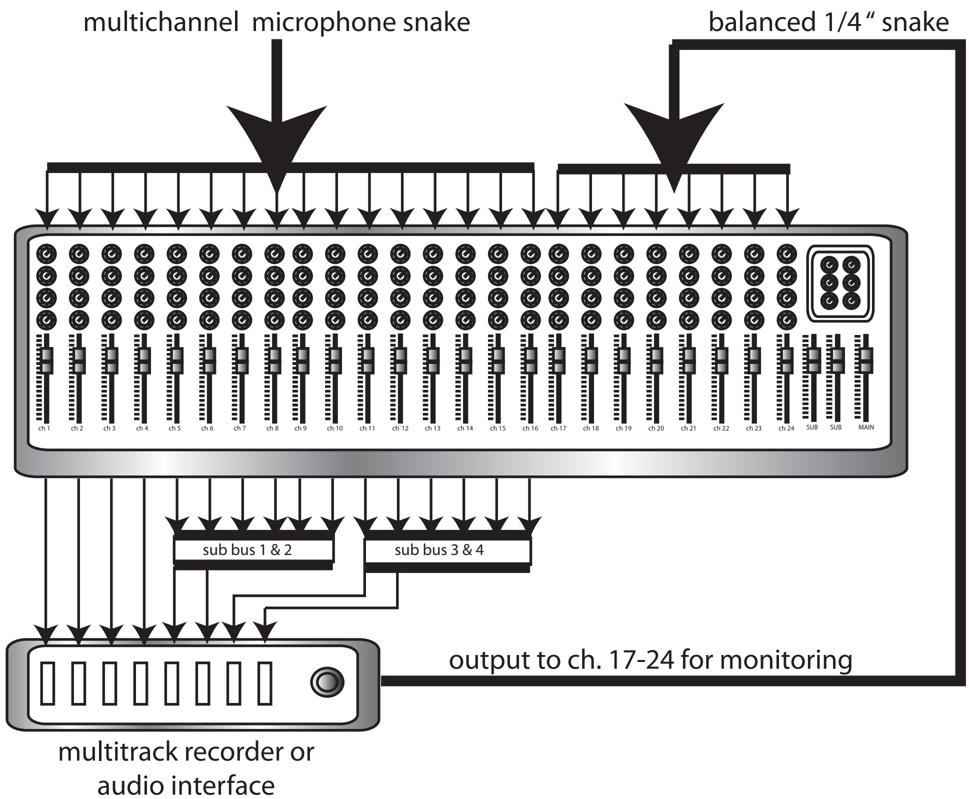


FIGURE 6.7

Connecting a twenty-four channel console to an eight-track recorder or audio interface

Some mixing boards provide a toggle to switch between microphone and ¼-inch inputs. If more channels are needed, a single channel strip can handle input from the microphones and an external recorder via such a toggle switch. To preview the output of the recorder in this setup, disarm all tracks in the recorder and switch each channel to the ¼-inch input. Be sure to switch back to microphone inputs prior to arming tracks for recording on the external device or you will hear a nasty feedback loop.

For monitoring, the control room out or main out could be attached to an external power amp that drives two passive monitors or the control room or main out might be attached to a pair of active (powered) speakers. Take care if you happen to be using a powered mixing console: the powered speaker outputs should be plugged into an appropriate speaker system—never into a jack that is expecting an unpowered source.

Multitrack Recording via an Audio Interface

In this example, an eight-channel audio interface is used instead of an analog or digital mixer. Microphones and other sources are plugged into the audio interface and the interface is connected to a computer via a FireWire or USB 2 cable (see figure 6.8).

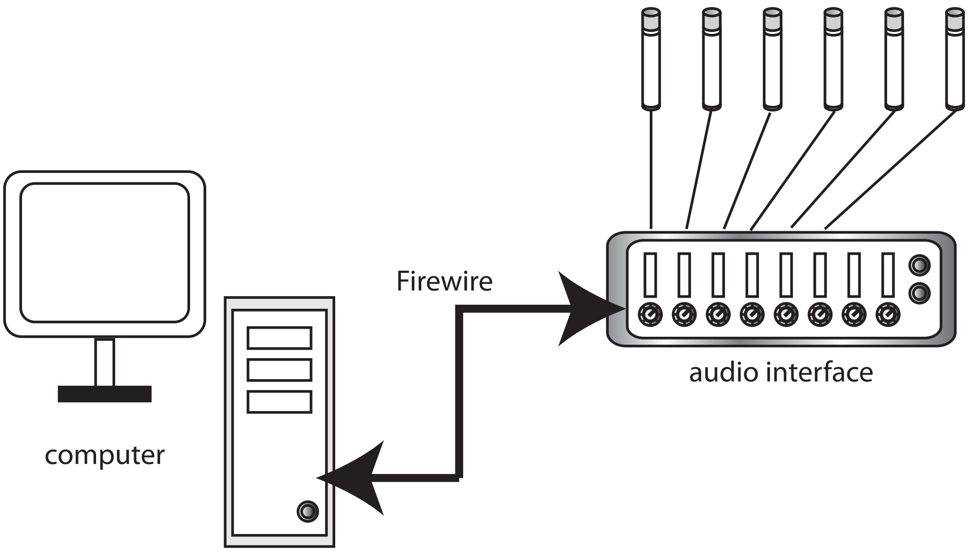


FIGURE 6.8

Multitrack recording with an audio interface

A software-based virtual mixer (see figure 6.9) or knobs on the interface are used to adjust the trim of the preamps. Recording is initiated on the computer with Digital Audio Workstation software and output levels are adjusted via channel faders in the DAW. One of the benefits of this system is that audio channels can be monitored virtually—additional output channels are not needed for monitoring. Headphones could be used to monitor the audio interface while

FIGURE 6.9

MOTU virtual mixer

recording live or the output of the interface could be connected to a passive or active reference system in a studio setting.

Creating a Budget Headphone Mix

It is often necessary to provide a headphone mix when multitrack recording or running a sound board for a live performance. One tactic is to assign channels to one or more sub buses (sub mix, or subgroup) and output those signals to a headphone distribution amplifier or monitor system. However, not all mixing boards provide sub buses or the buses may already be in use for recording or mixing purposes.

Auxiliary sends provide a convenient solution. You can visualize an auxiliary send like the main bus: varying amounts of signal can be added from each track and the sum of these signals can be sent via an auxiliary output to a headphone distribution amplifier. Most mixing boards provide at least a few auxiliary sends so this can provide a practical way to provide one or two headphone mixes. In a live sound application, separate monitor mixes could be provided using this same approach. The only difference is that each auxiliary send mix is sent to a powered monitor or amplifier instead of a headphone distribution amplifier.

Purchasing a Mixing Console

Although an understanding of mixers is essential for both hardware and software-based recording, hardware mixers are not always necessary. For example, an audio interface provides the functionality of a mixer but most of the controls are available via a software interface. For bands, a mixer or hybrid mixer/audio interface may be a better choice since the mixer (and channel settings such as EQ) can be easily configured during a live performance.

It can be a daunting task to select a mixing console for home recording. At the time of this writing, over two dozen sixteen-channel mixers in a wide variety of configurations are available from a popular vendor. Consider the following suggestions as you research possibilities for your project studio.

Narrowing the Choices

Although all mixing consoles require power to operate, some mixers that are designed for live sound reinforcement have built-in power amplifiers that are used to power speakers. Though there are always exceptions, it has been my experience that these types of mixers are generally not a good choice for home recording.

Digital or Analog

The lines have blurred in recent years between digital and analog consoles. For example, some analog mixers provide the ability to connect to a computer via a FireWire or USB 2 cable. Digital consoles may offer fader automation and built-in effects and can also be connected digitally to a computer or external recording device. Both of these options, depending on the specifics, may eliminate the need for an external audio interface. Although I would not hesitate to purchase a digital mixer, it is worth noting that there can be an extra layer of complexity involved in synchronizing streams of audio data from multiple devices.

Another option is to purchase a purely analog console and connect it to an external multitrack recorder or digital audio interface. In my estimation, the combination of a good analog mixer and a dedicated multitrack recorder such as an Alesis ADAT-HD24 is an ideal solution for live recordings.

Sound Quality

A primary purchase consideration involves the quality of sound versus the price point. While the assessment of sound quality is subjective, you can learn a great deal by reading professional reviews in trade magazines such as *Mix* and *Electronic Musician*. There are numerous online forums that can give you a sense of how a piece of gear is working in the trenches. At the end of the day, though, you should use your own ears to evaluate the audio quality of a given hardware component, so I would encourage you to support local vendors or purchase from online vendors who have a good return policy and can provide appropriate pre-sales advice.

Channels and Connectivity

Consider the total number of tracks you intend to record, as well as the type of inputs and outputs. Ideally, the mixer should provide a balanced XLR and ¼-inch input, an insert, phantom power, and a direct out on each channel. Practically speaking, most budget consoles will provide some XLR and some ¼-inch inputs but not necessarily both types of jacks on each channel.

Digital connectivity is also a consideration. Some mixing boards provide a USB connection capable of carrying only two tracks (used for recording the main output of the mixer to a computer) while some mixers provide multiple channels of data via a USB 2 or FireWire port. A given console may also provide an optical connection for transmitting multiple channels of digital audio to an external device such as a multitrack recorder.

Phantom Power

Phantom power is essential if you intend to power a condenser microphone. Phantom power may also be used to power active direct boxes, so consider both the number of condenser microphones *and* active DI boxes you intend to simultaneously use.

Sub Busing

You will typically see numbers like 32 × 8 or 16 × 4 used to describe a mixing console. The first number refers to the number of tracks and the second refers to the number of sub buses. Sub buses are very useful when mixing or running live sound. They can also be used to set up a monitor or headphone mix.

You will want to consider if sub busing architecture is important and determine the number of sub buses you will need. I typically use a computer when doing a final mix so, while sub busing is extremely helpful in a software-based mixer, sub buses aren't something I require in an analog sound board unless I intend to use the mixer in a live situation.

Conclusion

We have touched on many terms and concepts associated with mixing and mixing consoles. Although this chapter focused primarily on hardware consoles, in other chapters you will see that these concepts are readily applicable to other contexts, such as the virtual mixing consoles provided in Digital Audio Workstation software.

7

Using Digital Audio Workstation Software

The current crop of Digital Audio Workstation (DAW) software offers an incredible amount of power to musicians. These programs can handle multitrack recording at high sample rates, and they are used to mix and master everything from humble demos to platinum-selling albums and music for movies and television. There is a downside to the vast processing power: DAW software is typically very complex and can be difficult to learn. For example, the primary manuals for Logic Pro 8 total nearly 1,700 pages! The numerous training books geared to specific DAW applications attest to the complexity of these programs.

In this chapter we will explore the use of DAW software from a slightly different vantage point from the many "how to" books. Over the years I have used most of the major DAW applications, and it is evident is that there is a commonality to the feature set of most of these applications. Learning the central techniques and concepts associated with the primary DAWs will allow you to develop proficiency with a given program in a short amount of time. This doesn't mean you will never need to look at a manual again, but it does mean that you will be able to get up to speed by quickly finding the information that will allow you to make music.

In the following pages you will learn how to configure and operate a workstation. You will learn about common editing tasks and how to apply plug-ins. Along the way you will also learn how to bounce a project to disk and develop proficiency with your workstation of choice. You will notice that screenshots were taken from a variety of applications in order to emphasize the commonality among products from a variety of vendors.

Although Digital Audio Workstation software can be expensive, many of the entry-level programs offer plenty of power for most recording and editing tasks and a number of these applications are listed in chapter 2.

Setup/Input

In order to use a workstation, it is necessary to configure your digital audio interface to work with your workstation software. If you recently purchased an audio interface, you will first need to install drivers so that the interface is made available to your operating system and, in turn, to your DAW program. The specific steps will, of course, be different for each manufacturer, but I always recommend against installing drivers from any disks you may have received from the manufacturer. In most cases, it is a better idea to download drivers directly from the manufacturer since it is likely that the drivers have been improved since the product shipped.

Once you have installed the drivers, check your system to see if the audio interface appears to be working and is visible to the operating system. In Windows, open the sound control panel to see if the interface is available. OS X provides a similar configuration panel, called Audio MIDI Setup, which is located in the Utilities directory.

Note that some audio interfaces will be visible only to a particular application. For example, Pro Tools interface drivers can be installed for both Pro Tools and the operating system, or the drivers may be installed just for Pro Tools, in which case the interface will not be available to other applications.

A configuration application will usually be installed when you install a driver for an audio interface. This is where you can set the bit depth and sample rate, and configure additional digital inputs and outputs. Note that in some situations these settings will be made in a configuration window directly within your DAW.

Buffer

Your DAW software will provide an option to set the size of the audio buffer (see figure 7.1). Larger sample buffers will typically be less taxing on the computer so it may be possible to record more simultaneous tracks with a larger sample buffer. Try raising the size of the sample buffer if you experience clicks or pops in a recording. The downside is that larger sample buffers will contribute to latency—the amount of time between the input and output of a signal. For this reason, lower buffer sizes are typically used for real-time applications such as virtual synthesis.

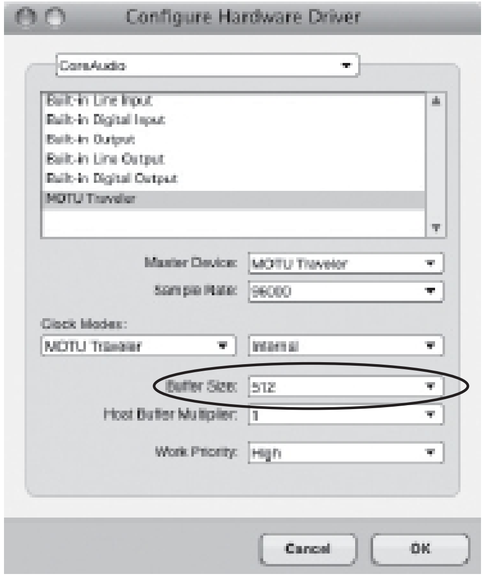


FIGURE 7.1

Setting the size of an audio buffer

Digital I/O

It is not uncommon for musicians to digitally connect a microphone preamp or additional interface to expand the number of inputs of a given interface. Digital inputs and outputs, as well as clock source, are typically configured in a hardware control panel. One of the most common formats is ADAT Optical (also called Lightpipe), which is capable of transferring eight channels of audio over a single cable.

An example of the way multichannel digital audio can be useful involves using two digital audio interfaces in conjunction with one another. The early model Digidesign Digi 003 audio interface has just four microphone inputs. One of the ways the number of inputs of such an interface can be expanded is to connect a second interface such as a MOTU 8Pre via a Lightpipe connection. In this instance, the audio control panel for the Digi 003 is used to configure the unit so that it knows that extra inputs are available via the Lightpipe input port.

It is important to synchronize devices whenever two or more digital audio devices are used in conjunction with one another. In the previous scenario, the Digi 003 could be set to synchronize its clock via the ADAT Optical input port. In this case, the 8Pre would function as the master clock.

Software Monitoring

Your DAW software will typically have an option to enable software monitoring (sometimes called patch-through or real-time monitoring). In most cases, the software can be configured so that musicians can hear reverberation or other effects while recording, but the effects aren't recorded to disk. A downside of software monitoring is that there may be a slight amount of latency, which can be disconcerting to players while recording. In some instances, a smaller audio buffer size can be effective in reducing latency. Also keep in mind that some signal-processing tools can add a significant amount of latency so it's a good idea to keep signal processing to a minimum during the tracking process.

Setting Input Levels

It is important to take the time to establish a good input level prior to recording. The level should be hot but not so hot as to cause distortion when a player plays a loud passage. Your digital audio interface should provide trim knobs for the purpose. Some audio interfaces will provide a software application that can be used to set input levels. Strangely, it is not usually possible to adjust input levels from within most workstation software. Although a few products integrate hardware and software in this way, in many cases it will be necessary to adjust trim levels directly on the audio interface. I mention this because the concept was a source of confusion when I first learned about audio recording. I remember being baffled as to how to set input levels until I realized it was not possible to set levels directly in my DAW software—it was necessary to adjust input gain directly on the audio interface.

Recording/Track View

All workstations provide a track view that is typically used while recording tracks and for editing of digital audio clips (see figure 7.2). In this view, audio tracks (and often many other types of tracks such as MIDI or virtual synthesizer tracks) are represented in a vertical list with a timeline that stretches from left to right. The word *clip* is typically used to describe the waveforms or MIDI data that are visible in a track view.

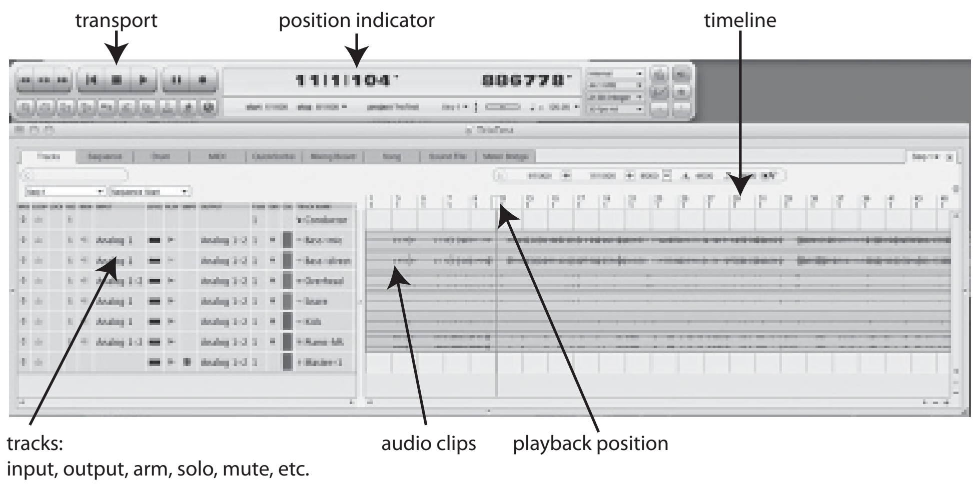


FIGURE 7.2

Essential elements of a track view

Selecting the Track Input and Output

You will want to set the track input and output prior to recording. With the input and output (I/O) settings, you are telling the software about the physical inputs and outputs you wish to use for the given track. The process of assigning inputs and outputs is very similar among DAW programs (see figure 7.3).

Some workstation applications provide the ability to rename the audio inputs and outputs in your system. This is a useful feature as it provides the opportunity to use a descriptive label for your equipment. For example, it's more convenient to select "ADAT 1" as an input instead of some arbitrary number like Audio 15 (which happens to be the name of the first channel of ADAT input on my system).

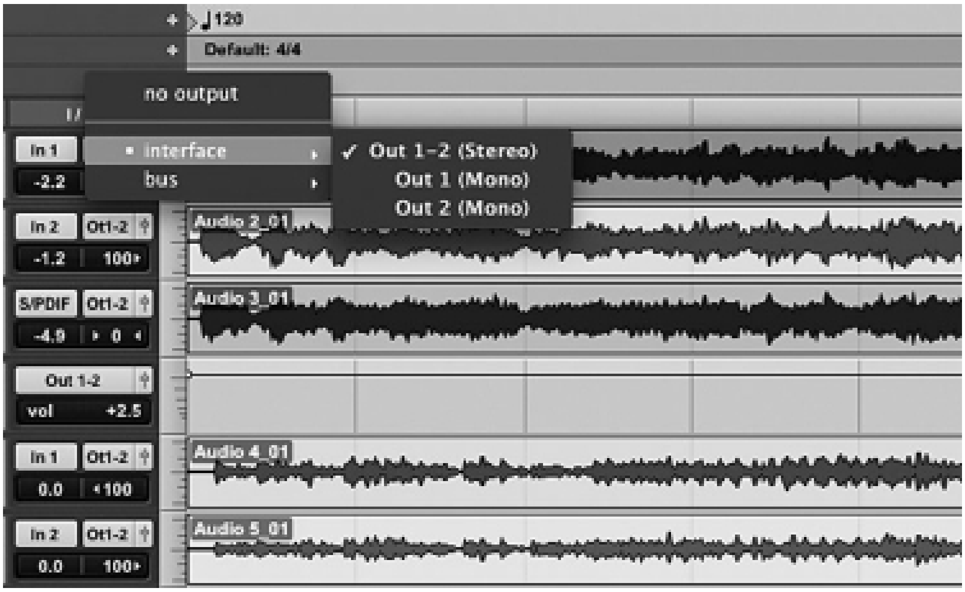


FIGURE 7.3

Assigning track input and output

Creating Tracks

Creating new tracks is usually as simple as selecting a Track menu and clicking Create. In most cases you can select the type of track (audio, virtual instrument, auxiliary, bus, etc.), as well as determine if the track will be mono or stereo. It often makes sense to select stereo when you use two microphones to record a single source, although it is necessary to create two mono tracks when recording using the mid-side technique as each channel will need to be handled independently. (You can read about mid-side recording in chapter 4.)

Naming Tracks

Although it might seem insignificant, it is helpful to name each new track prior to recording. Of course, it's much easier to mix when the tracks are labeled "Piano" and "Guitar" instead of "Audio 1" and "Audio 2," and I find that the best time to label a track is when a new track is created.

Arming Tracks

You will need to arm tracks prior to recording (see figure 7.4). Most workstations will provide a small red arming icon on each track for the purpose. Depending on the configuration of your software, you may not hear audio input on a given track until the track is armed for recording. Mute and solo buttons are typically located next to the track arming button and are useful during the mixing process in order to focus on specific tracks.

Transport Controls

After configuring your interface, setting levels, and creating and arming tracks, it is time to record. All workstations feature some variation of the ubiquitous tape transport controls shown in figure 7.5.

FIGURE 7.4

Track arm, mute, and solo buttons

FIGURE 7.5

Typical transport controls

Depending on your software, you will be able simply to click the record button or it may be necessary to click record and then play. In any case, you will want to learn the key commands for record, stop, pause, and play since they are used so frequently.

Metronome

Most transports provide a metronome icon that you can use to enable or configure the metronome. In the past, metronomes typically consisted of routing MIDI notes to an external sound module or synthesizer. Today, most workstations provide a primitive software synthesizer whose function is to supply metronome clicks. Additionally, you will want to consider if the metronome should provide a count-off, and if so, how many bars.

In some situations it may work better to use a prerecorded drum loop instead of a simple click track. Musicians often respond better to a discernable groove so it may make sense to provide a scratch drum loop if you intend to record a bass, vocal, or other instrument prior to recording drum tracks.

Position Indicators

The transport will provide an indication of the current playback position (see figure 7.6) and additional fields may be available to set the start and end locations for loop mode and automatic punch-ins. In most cases you can quickly set the start and end location by clicking and dragging in a timeline just above the tracks.

Loop Mode

Most workstations provide a loop playback mode that is typically found on the transport. This can be useful for tweaking the mix of a section or practicing a

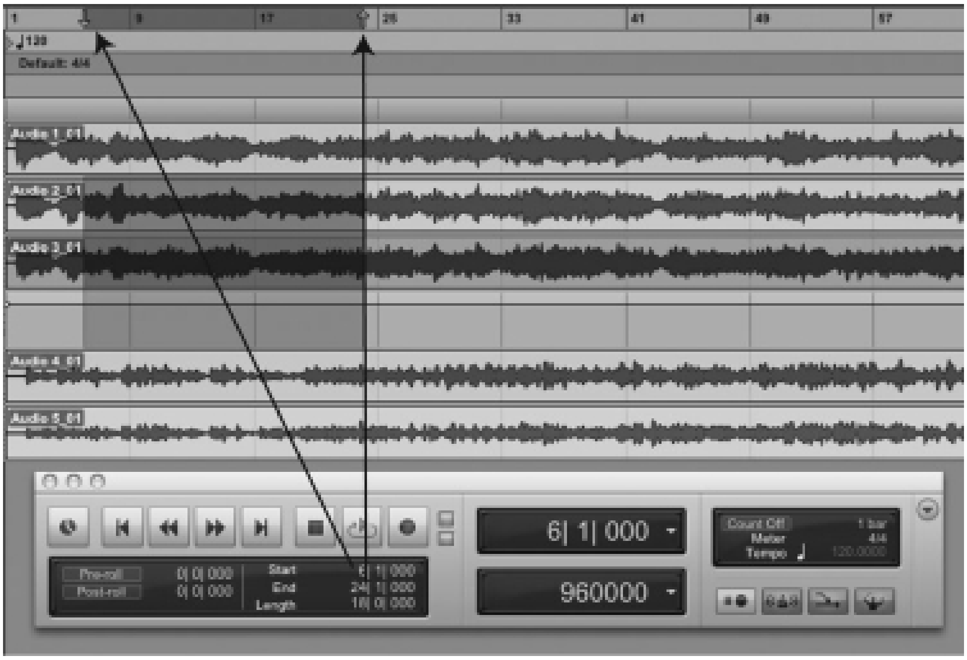


FIGURE 7.6

Location indicators in a typical transport

take. Some programs also support the option to record multiple takes while in loop mode.

Punch-ins

DAW programs typically provide an auto punch-in feature to facilitate overdubs (see figure 7.7). In the days of analog recording, engineers would manually "punch in" and "punch out" of record mode in order to capture a new take for a given musical phrase. The procedure took finesse since a portion of a preexisting take might be inadvertently recorded over if the engineer didn't engage the recording mechanism at just the right time. With a modern DAW, punch-ins are usually as simple as specifying the start and end location of the punch-in, enabling punch-in mode, and pressing record.

FIGURE 7.7

Setting up a punch-in

Alternate Takes

Alternate takes may be handled in a number of ways. Often, it's convenient to create several duplicate tracks and record different takes on adjacent tracks. Some programs also offer the ability to record multiple takes on the same track. In this instance, the program will keep a list of takes and provide a method to preview and select the best take. We will discuss a related concept, track comping, later in the chapter.

Common Editing Tasks

There are a number of editing tasks that you will frequently perform when working with digital audio data. As with all of the concepts in this chapter, there is a commonality to the basic editing procedures across all of the major DAWs. In fact, if you read chapter 5 on podcasting, it's likely that you will already be able to handle the following edits in most DAWs.

Destructive and Nondestructive Editing

Depending on the software application, edits may be either destructive or nondestructive. Destructive edits directly change the given audio file and are useful when you want to permanently alter a file, such as adding a fade-out to a master bounce prior to burning the song to CD. You will want to be judicious in your use of destructive edits as there may be a limit to the number of undo operations you can perform on the file.

Most workstation editing involves nondestructive editing. In many cases, when you cut or trim an audio clip, the program stores a list of edits but doesn't actually alter the underlying file. The edits are rendered when the program plays back the given clip.

Selection Tool

A selection tool is usually represented by a hand or pointer. Use this tool to select an entire clip to copy it, delete it, or apply processing such as a normalization function. Typically, you can also use the selection tool to lasso several clips.

Cut and Paste

Cut and paste operations are as simple in a DAW as they are in a word processor. Some programs provide a scissors tool or "split" function. Another common way to cut or select digital audio material is with a marquee or "I" tool. Click and drag with the marquee tool to select a portion of a clip and use the cut or copy command as needed.

In order to paste a clip that has been copied to the clipboard, position the playback marker and select a destination track. The playback marker can usually be adjusted by dragging the marker or clicking in a timeline above the track view. These edit, cut, and paste operations are largely the same with all DAWs.

Trim

A trim tool will make quick work of clipping the start or end of a phrase of audio. In some cases, you can simply hover the mouse over the left or right side of a clip to engage the trim tool or the tool may be found in a tool palette.

Common Editing Operations

Although modern DAWs provide numerous ways to manipulate data, the most common tasks are listed in the paragraphs that follow.

Cross-fade

Unlike the days of analog tape, splicing digital audio is very simple to do. In some situations an abrupt splice will be necessary, but if your goal is to make a seamless transition you will want to use a *cross-fade* (see figure 7.8). Cross-fades are useful if you want to splice a sustained note from two different takes. The idea behind a cross-fade is that the first clip fades out while a second clip fades in. Depending on the configuration of your software, the fades may be linear or exponential.

FIGURE 7.8

Using a cross-fade to splice audio

Editing Samples

A pencil tool is typically provided to edit individual samples and is commonly used to fix clips or pops. For example, an audio anomaly is clearly visible in the first screen shot of figure 7.9 but was removed by application of the pencil tool.

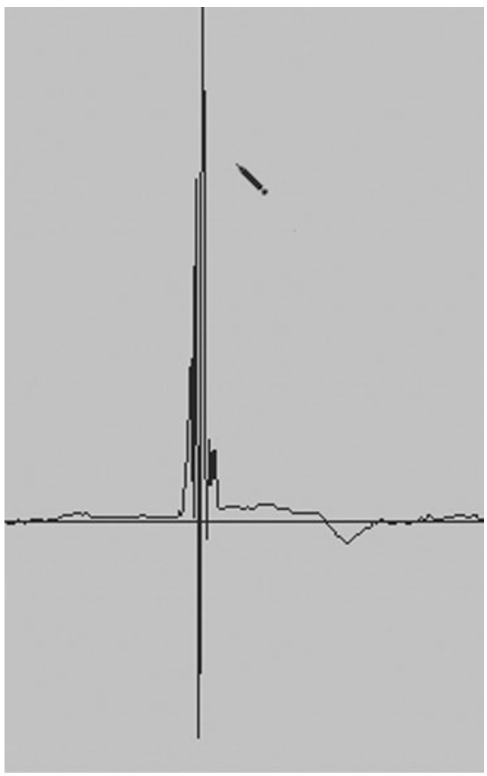


FIGURE 7.9

Using a pencil tool to fix an audio anomaly

Track Comping

The term *comping* is used to describe the process of creating a composite track from numerous takes (see figure 7.10). Track comping can be done using the editing methods listed above, but many modern workstations provide additional support for this common process. In the following screenshot, multiple takes are visible in an opened take folder in Logic. The selection tool is then used to select the best parts of each take to create a perfect composite of all the best parts. Most modern DAWs offer a similar comping function.

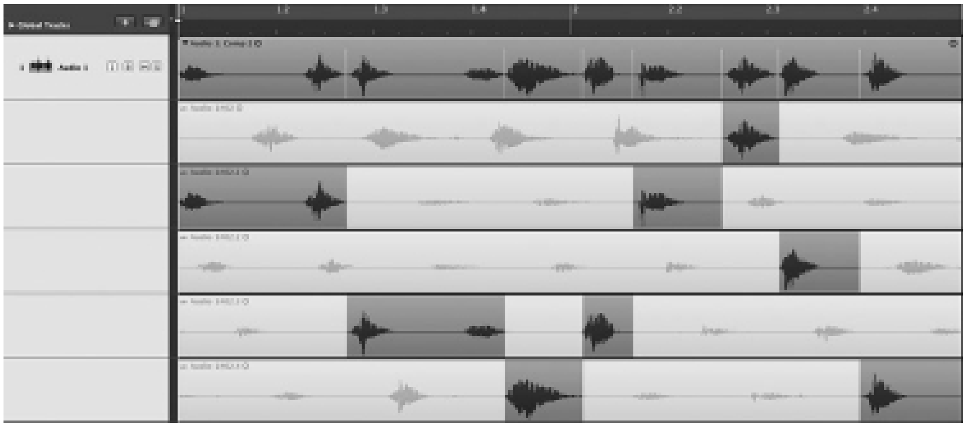


FIGURE 7.10
Creating a comp track in Logic

Conductor Track

A global track, sometimes called a conductor track, will be available for inserting markers, meter, and tempo changes in a project (see figure 7.11). If you use a click track, it will be important to establish an accurate tempo at the outset since it can be problematic to change the speed of playback of an audio track. (Pitch and time stretching tools are available in most workstations, but the results tend to be less satisfactory for large tempo changes.) In most cases markers, meter changes and other events can be inserted or removed with a pencil and erasure tool directly within the conductor track.

FIGURE 7.11
Conductor track

Mixing View

The next section will focus on the concepts that will enable you to use a virtual mixer within a Digital Audio Workstation. The aesthetics of mixing, compression, EQ, and the like are covered in later chapters of the book.

Virtual Signal Flow

Although some mixing tasks like automation may be handled in track view, most mixing operations will be handled in a mixing view. If you haven't done so already, I encourage you to read chapter 6 because concepts such as input, output, and busing are the same for hardware and software mixers. However, a brief review is appropriate at this point.

Just like an analog mixer, signals flow from the top of the channel strip to the bottom. There is typically a space at the top of the channel strip for inserts, which can be used to configure EQ, reverb, compression, and the like for an individual track. One or more auxiliary or bus knobs may be available on the channel strip. These knobs are used to tap an incoming signal for processing. For example, a single auxiliary channel could be configured with reverb and one or more channels could utilize an auxiliary send to route audio to the auxiliary channel for processing. This is more efficient in terms of computing power than inserting individual reverb on each channel. Channel strips will provide a level control slider and pan knob, as well as buttons for mute, solo, and record. In most cases, a virtual channel strip will provide access to hardware input and output assignments. Figure 7.12 shows the many elements found in a typical virtual channel strip.

Most DAWs provide a place for comments at the bottom of a channel strip. Use the comments section to document the type of microphone, polar pattern, pad, roll-off, preamp, or other settings. This type of documentation can be very helpful when revisiting a project months or years after a recording session.

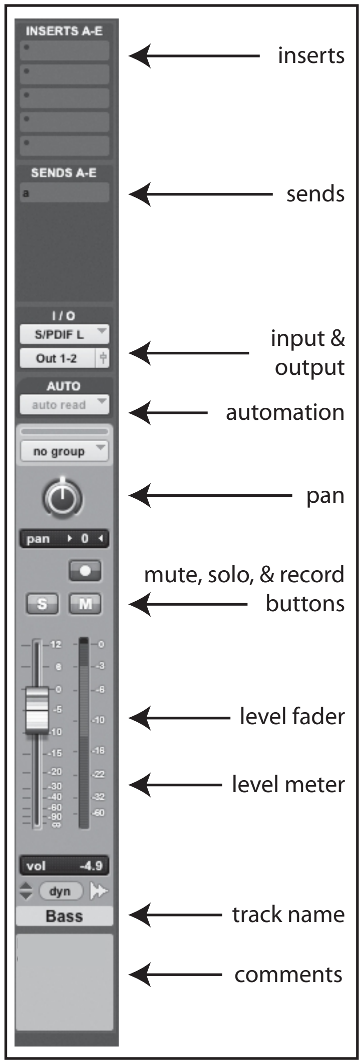


FIGURE 7.12

Typical elements of a virtual channel strip

Automation

Track automation is a convenient feature that is used to record mixing moves such as a change in the level or pan position of a track. You might, for example, want to boost the level of a guitar during a guitar solo but pull the level down at the end of the solo. You can automate the process by enabling track automation so that your fader movements are recorded by the workstation. There are four common automation modes available in most DAWs.

Read

When a track is in read mode, the program will respond to previously automated events but will ignore new fader movements. Read mode is a good choice when you are happy with a track and don't want to inadvertently change the level of a track.

Off

Of course, the off setting means that a track will ignore any automation events. You will typically want to disable automation when starting a mix because it can be distracting when a program plays back automation during the initial leveling process.

Touch

Touch mode is similar to read mode, but current automation events are overridden when a fader is moved. The fader will return to the previous setting when a fader is released. This is likely the most useful mixing mode since the program will use exiting automation but update the data as you adjust your mix.

Latch

Like touch mode, fader moves will be automated if you adjust a fader in latch mode. The difference is that, upon releasing a fader, the fader will stay at the new position instead of returning to the previous value, as in touch mode. For example, this mode is useful when you realize that your automated guitar track is too loud or soft throughout a song. In this case, engage latch mode, initiate playback, and drag the fader to a more appropriate level.

Editing Automation Data

Although track automation works well in mix view, there are times when it is useful to visually edit automation data. For example, visual editing provides a convenient way to hide a wrong note (see figure 7.13). You can zoom in on the waveform and use control points to draw automation values in order to mute a wrong note. This type of automation can be difficult to achieve by adjusting volume sliders in real time, but it's easy to do using control points. A track view will typically provide a button or toggle to reveal automation for a given track. In most cases, you can simply click to add control points. Most programs will remove a given control point with a second click or the application of an erasure tool.

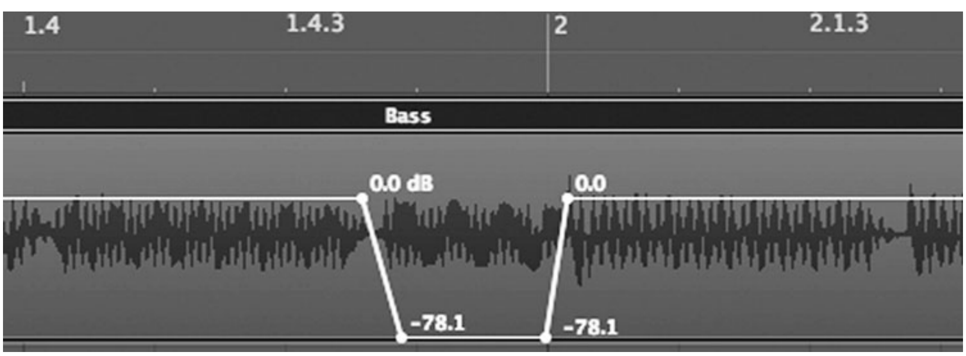


FIGURE 7.13

Using automation control points to fix a wrong note

Plug-ins

The use of plug-ins is an important part of most mixing projects. A plug-in is a type of software that can be used to expand the functionality of a software host. You are probably already aware of plug-ins for your Web browser, which may be used to facilitate audio or video playback. In a similar way, audio plug-ins offer a way to expand the functionality of your DAW.

Your workstation will undoubtedly come with a variety of built-in effects and signal processors, but many users will add additional plug-ins for special tasks such as music synthesis. There is a wide variety of third-party plug-ins available from numerous vendors. Some plug-ins are free while top-tier plug-ins may cost over $1000. Of course you will want to carefully check to see if your workstation supports a given plug-in format prior to investing in any new plug-ins.

In most cases, a plug-in is inserted into a track or the main bus. Most programs feature several rows of inserts near the top of each track and, in most cases, a track plug-in can be selected by double-clicking an available insert or clicking and holding the mouse. Some plug-ins such as virtual instruments may be selected as a track output destination instead of being inserted into a track.

The types of plug-ins that are available to you are determined by your operating system and choice of DAW. For example, Audio Units (AU) plug-ins are unique to the Mac OS while Virtual Studio Technology (VST) plug-ins are found on Macs and PCs but depend on a host program that supports Steinberg's VST format. Table 7.1 details the common plug-in formats.

Plug-in Wrappers

In some instances, you might want to use another type of plug-in that is not supported by your DAW. In these cases, a special type of application called a *wrapper* is used to provide an interface between the host and the plug-in. The VST to RTAS Adapter by FXpansion is one such program and provides the ability to use VST plug-ins within Pro Tools.

Overview of Plug-in Categories

There are many categories of plug-ins available within most DAW applications. Some categories such as dynamic processors are almost always applied to a channel strip or the main bus while others types of effects may be inserted into the channel or added via the auxiliary send and return mechanism. The most common processors are listed in the next section. Note that we will explore each of these categories in more detail in chapter 10, with a focus on both the creative and the technical aspects of these tools.

TABLE 7.1 Common types of plug-ins

Acronym	Title	Description
AU	Audio Unit	Developed by Apple computer, Audio Units are an integral part of the OS X operating system.
AudioSuite		Digidesign describes AudioSuite plug-ins as providing "file-based processing, meaning they process or alter the sound file and create a new file with the processed sound. The resulting effect is applied to the entire file." AudioSuite plug-ins are typically used for normalization or tasks that are not time based.
DirectX		DirectX is an application programming interface (API) developed by Microsoft to support multimedia programming. The term DX indicates an audio processing plug-in. DXi plug-ins are virtual instrument (synthesis) plug-ins.
HTDM	Host Time Division Multiplexing	HTDM is a hybrid of TDM and RTAS. These plug-ins are only available on Pro Tools HD systems.
MAS	Motu Audio System	MAS is a plug-in format developed by Mark of the Unicorn for their workstation application, Digital Performer.
RTAS	Real-Time Audio Suite	Developed by Digidesign, RTAS plug-ins are used in Pro Tools LE and Pro Tools M systems. Where high-end Pro Tools HD systems utilize special digital signal processing (DSP) cards, the RTAS plug-ins utilize the power of the "host" system.
TDM	Time-Division Multiplexing	TDM plug-ins are used with Pro Tools HD systems. Most processing is handled by external DSP cards in a Pro Tools HD system.
VST	Virtual Studio Technology	A popular plug-in format developed by Steinberg. There are many free VST plug-ins available on the Web, as well as commercial products from a wide variety of vendors.

Dynamic Processors

Dynamic processors are used to limit, compress, or expand the dynamic range of a recording. Common dynamic processors include compressors, expanders, limiters, and noise gates.

Space and Time

Effects processors such as reverb and delay are used to give a sense of expansiveness or dimension to a recording. These tools are used to convince a listener that a recording was created in a large performance hall or cathedral instead of a small project studio.

Tone and Modulation

Equalization (EQ) is used to boost or attenuate a range of frequencies and affects the tone of a recording. EQ can be used to make a piano sound brighter or an acoustic guitar more mellow. EQ can also affect our perception of loudness. Modulating processors such as chorus and phase shift modulate a signal to create interesting effects that occur as the result of delaying, modifying, and combining signals.

Modeling
Modeling plug-ins are used to simulate hardware devices such as a guitar or bass amplifier. Some modeling plug-ins are designed to emulate classic dynamic processors and preamps.

Pitch Correction
Pitch correction plug-ins analyze the pitch of a monophonic signal in real time and modify the output as needed to fix any pitch imperfections. Pitch correction is typically used on vocal tracks or other mono sources to fix out-of-tune notes but the technology can also be used for interesting special effects.

Synthesis
Synthesis plug-ins don't ordinarily affect an audio signal (although some are used in this way). I mention synthesis plug-ins at this point because they represent an important category of music-making tools to many musicians. Virtual synthesizers are the software equivalent of a hardware synthesizer and include many subcategories such as sample playback, analog emulation, additive synthesis, and beat boxes.

Working with Files

When you record audio in a DAW, incoming audio is stored to your hard drive as a series of independent files. This is an essential point to understand if you have never used a DAW before: your project file is separate and distinct from the individual audio track files. In most cases, the DAW will create a project folder and audio files will be stored in a subfolder.

Location of Track Files
It's important to keep track of both the project file and track files so that you can backup your work or move a project to a different computer. Although modern DAWs will usually take care of setting up a subfolder within the project hierarchy, in some situations it is possible that track files may be recorded to a different location so it is a good idea to verify that data are being stored where you expect it.

Backing up a Project
You will want to create frequent backups of important audio data. As long as data are recorded in a subfolder as described above, you should be able simply

to copy the project folder (with all subfolders) to an external hard drive or burn the data to a DVD. Although newer Blu-ray burners are still expensive, I predict that these disks will provide an attractive storage option in the future. A two-layer Blu-ray disk can store about six times as much data as a two-layer DVD.

Exporting a Project to Another DAW

It may be necessary to export the raw data files created in one DAW so that they can be imported into another DAW for editing and mixing. The process usually consists of selecting audio clips in the track view and selecting an Export function from the File menu. Be sure to disable signal processing and track automation if you plan to export the files so that they can be mixed on another system, as you will likely not want to commit to these settings when exporting tracks. In many cases, you can simply import audio files directly into another DAW although the previous steps are usually necessary if your project contains punch-ins and overdubs.

Bouncing to Disk

Up to this point, we have considered the primary recording and editing functions that are available in a typical DAW, as well as the type and variety of plug-ins that can be used to process signals in the digital domain. Once a project has been recorded and mixed, you will want to bounce the project to disk so that the file can be burned to an audio CD or imported into an MP3 player. The term *bouncing* refers to the process of rendering the tracks, complete with all of the track settings and plug-ins, to disk as a stereo file. In most cases, bouncing is initiated via a dedicated bounce button or menu item.

Depending on your software, the bouncing process may be in real time or the program may be able to render the master file without actually playing through the project. In some instances, real-time rendering is a necessity, such as when you combine "real" MIDI tracks from an external synthesizer or sound module. In this case, it's usually most convenient to record the external MIDI tracks as audio tracks prior to bouncing the project to disk. The process of recording MIDI tracks as audio is similar to the process of recording any other type of audio track. Start by connecting the audio outputs of the keyboard or synthesizer to the line level inputs on an audio interface. Next, create one or more audio tracks that correspond to the keyboard inputs on the interface and arm the tracks for recording. Initiate the record function in your DAW and the existing MIDI tracks will be recorded to disk.

Selecting a File Format

Your workstation software will likely offer the option to bounce different file formats. You will want to select an uncompressed file format such as WAV or

AIFF if you intend to burn the file to a CD. Compressed formats such as MP3 and M4A are useful if you want to trade quality for file size. For example, MP3 format is a good choice if you intend to embed the file in a Web page or e-mail a low-resolution preview to a client.

Developing Efficiency with Audio Software

Given the complexity of modern DAW programs, it can take some time to become fluent with a given software application. The following tips and strategies will help you apply the concepts presented in this chapter to your DAW of choice and quickly get up to speed.

Common Key Commands

Though some types of edits are better handled with the mouse, you will want to avoid using the mouse for most functions since key commands are typically much faster. One of the first things I do when learning a new DAW is to visit the program Preferences panel in order to learn the essential key commands. This can also be a great way to learn about other functions that may not be emphasized in the manual.

Your work style is likely different from my work style, but my "must know" key commands include the following. Note that it is often much faster to learn these commands by looking at the application's Preferences menu instead of flipping through many pages of a manual.

- Play
- Pause
- Stop
- Record
- Arm/disarm a track
- Solo/mute a track
- Create a new track
- Delete a track
- Go to measure
- Zoom in/out (horizontally and vertically)
- Toggle loop
- Selection tool
- Marquee tool
- Toggle between track and mix views
- Insert marker

I think you will be surprised at how much work you can accomplish once you learn to navigate, select tools, and operate the transport with key commands.

Use Markers

I tend to make liberal use of markers when recording and mixing music. Markers can be added to specify the location of an interlude, chorus, or solo, and this is a great convenience when you revisit a project. Most programs allow you to insert markers on the fly so you might want to consider mapping the form of a song as it is being recorded to disk. It is not uncommon for a player to want to punch-in a solo during a tracking session so markers are one of the tools you can use to anticipate the needs of musicians during a recording session.

Using a Control Surface

For some users, a control surface will enhance fluency with a given workstation. Some control surfaces feature motorized knobs and sliders, which will provide a more tactile experience during the recording and mixing process. Gestures made on the control surface will be reflected in the automation of tracks in the DAW, and in the case of motorized control surfaces, automation data from the workstation are updated on the control surface.

You will want to give some forethought if you plan to purchase a control surface and digital audio interface. Some control surfaces require a MIDI in and out port so you will want to consider this when purchasing a digital audio interface, as not all audio interfaces feature MIDI ports. Some control surfaces such as the Digi 003 feature an integrated audio interface, and these systems are a good choice for some users.

Although control surfaces can be wonderful tools, they are not for everyone. A control surface is likely not the best choice if you tend to use the mouse to insert automation control points in a track view. On the other hand, a control surface might be very useful if you find yourself recording track automation such as fader moves in a mix view.

Figure 7.14 shows a control surface interacting with a software application in a recording session. Note how track names, pan position, and other data are reflected on the control unit.

Using a Wireless Remote Control

A wireless remote control can be very convenient if you frequently record yourself. Devices such as Frontier Design's Tranzport can allow you to wirelessly control your DAW while you are in a vocal booth or drum room.

Virtual Network Computing (VNC) technology provides another way to control your Digital Audio Workstation. With VNC, a computer can be configured so that mouse input and keystrokes can be controlled from a remote location. Screen updates are sent over a network to provide visual feedback. One interesting application of this technology is Jugaari's Jaadu VNC, which provides the ability to control a PC or Mac over a wireless network from an iPod Touch or iPhone. Several virtual DAW controllers such as ProTransport and AC-7 are available for the iPhone and iPad at a nominal cost.

FIGURE 7.14

Close-up of mix settings on a control surface

Conclusion

Although DAW software can seem complex to a new user, the concepts and techniques listed in this chapter are applicable to most Digital Audio Workstation applications and a familiarity with these concepts will allow you to become reasonably proficient in a short amount of time.

If you are new to audio recording, I would encourage you to record a short take and practice moving between track and mixer views and doing basic edits such as cut, copy, paste, and trim. As these tasks become comfortable, try adding a few inserts or enabling and editing track automation. As you become more fluent with the primary tasks, consider assigning custom key commands and screen sets to enhance your preferred workflow. When you feel comfortable with the software, try some of the mixing and editing "etudes" listed below.

Practical Application: Mixing and Editing Etudes

- Record and edit a simple podcast.
- Create a "fast talk" commercial by recording several sentences of text and removing the space between the words. Apply time compression to speed up the words.
- Record a few sentences of text and reorder the words so that they sound seamless.
- Use EQ and effects to mimic the audio quality of a telephone, bathroom, cavern, cathedral, etc.

- Download audio files consisting of sound effects and mix them to make an audio story.
- Import an instrumental track from a CD and record, edit, and mix a voice-over such as a faux commercial, poem, children's story, etc.
- Import several tracks from a CD and use cross-fades to create a seamless transition between the tracks.
- Load several .wav files and create a *musique concrète*, a composition based on the combination and manipulation of recorded sounds.
- Record and overdub several instrumental or vocal tracks and use track automation to mix the tracks.
- Import audio from several CD recordings and cut and edit phrases to make a remix.
- Record several long vocal and instrumental notes and use cross-fades to morph between the notes.
- Record several takes of a vocal or instrumental line and explore the concept of track comping to make a perfect final take that is a composite of several parts.
- Create a backing track and add a vocal or instrumental overdub. Try to punch in a phrase or single bar.

8

Active Listening

A premise for this book is that most musicians already posses the necessary aural skill to record and mix a successful recording. Unfortunately, the type of ear training that we receive in a lesson or the classroom tends to focus solely on the performance aspect of music making: playing in tune, producing a beautiful tone, identifying chords and other structures, using musical phrasing, blending with other musicians, and playing in time. Though these are all essential considerations for a musician, most of us have never been trained to evaluate recordings with the same level of detail and that is the purpose of this chapter.

One way to visualize an engineer's approach to listening is to think of a spotlight that is focused on a symphony orchestra in a darkened room. The spotlight might be wide enough to allow you to see the entire orchestra, but it can also be adjusted to focus on a single section or a specific performer. In a similar way, your ears can be taught to hear an entire mix or focus on the individual elements that make up the mix. By the end of this chapter you will possess a number of concepts that will help you pick apart a recording, and you will learn many techniques that you can implement to improve your ears in a proactive way.

Developing Aural Skills

Before we begin I would like to dispel what might be called the "Myth of the Golden Ears." The term *golden ear* refers to audio specialists who have a particular affinity to evaluate audio recordings. Although I recognize that these individuals exist, it is a fallacy to think that only a select few have the capacity to hear music on this level. I feel strongly that the ear *can* be trained to hear many of these details and that our capacity to hear can improve over time.

My first glimpse at the malleability of the human ear comes from my experience as an aural skills teacher. Most students arrive at the university with at least some intellectual knowledge of scales, chords, and intervals, yet many students are unable to identify these sounds by ear. This is largely due to the way musicians are taught in the United States—the development of aural skills is not usually part of a musician's education. With training, most students will develop the ability to recognize complex chord structures and progressions, identify modes, and pick apart four-part chorales. Clearly, then, the human ear has a capacity to be trained to hear these types of details. As an ear-training teacher, the process I focus on involves personalizing sounds (e.g., recognizing that a given chord might sound pretty, strident, unstable, and so on) and creating a taxonomy for sounds. For example, the interval of a second and a seventh both sound dissonant while a third and a sixth might be described as pretty. I will use a similar process in this chapter to facilitate the identification of sonic elements of a recording.

A second insight into the process of ear training comes from my work as a professional music transcriber. I started transcribing music in high school and distinctly remember trying to transcribe a recording by Emerson Lake and Palmer. That particular recording was beyond my aural abilities at the time so I moved to something else. It was interesting to revisit the recording later in life and discover that the sounds of the recording were very accessible to me. Had my ears magically changed? No, clearly my capacity to better hear the sounds of the recording came from many years of experience as a musician and transcriber. Now that I am more intentional in listening to the sonic properties of music, I can attest that my ears are improving in this regard as well. While I don't yet possess the golden ears of a Grammy-winning producer, my ability to discern problems in a recording are improving, and I know that my skills will improve as long as I am intentional about focusing on the sonic properties of recordings instead of focusing primarily on the performance aspects. This brings to mind the concept of brain plasticity. In his book *Musicophilia*, Oliver Sacks describes a conversation with a physician, Dr. Jorgensen, who lost hearing in one ear: "Dr. Jorgensen says that he believes his ear is 'better than should be expected from a seventy-year-old.' One's ear, one's cochlea, cannot improve as one gets older, but as Jacob L. clearly demonstrated, the brain itself can improve its ability to make use of whatever auditory information it has. This is the power of cerebral plasticity."[1]

Finally, it is helpful to recognize that the brain is a complex entity and that hearing is a multifaceted process that involves subconscious processing, as well as well as the application of the intellect. In his book *This Is Your Brain on Music*, Daniel J. Levitin describes the challenge of identifying auditory objects:

> The brain faces three difficulties in trying to identify the auditory objects we hear. First, the information arriving at the sensory receptors is undifferentiated. Second, the information is ambiguous—different objects can give rise to similar or identical patterns of activation on the eardrum. Third, the information is seldom complete. Parts of the sound may be covered up by other sounds, or lost. The brain has to make a calculated guess about what is really out there. It does so very quickly and generally subconsciously.[2]

The last sentence of the previous quote really spoke to me. When it comes to evaluating a recording, most of us let our subconscious do the majority of the work. The thing that I would like to stress is that, in my estimation, an *active* or *intentional* approach to listening is the key to developing your ear for this type of work.

Training the Ear for Active Listening

The most important step in training your ear for active listening is simply to develop an awareness of sound. Like most things musical, this awareness will likely come from a combination of intellectual and intuitive processes. For example, my composition students rarely use panning when they first compose and record a piece using sequencing software—they tend to focus on notes, rhythm, timbre, and perhaps the formal construction of the piece. A simple suggestion on my part to consider how panning might improve the mix is usually all that is required to open their ears to this important aspect of music production. Most musicians focus primarily on notes and rhythms and, thus, need to be proactive in focusing on sound in order to develop skill in the recording arts.

Another relevant quote comes from *Your Brain on Music,* where the author describes a conversation with Paul Simon: "Many years later, Paul Simon told me that the sound is always what he was after too. 'The way that I listen to my own records is for the sound of them; not the chords or the lyrics—my first impression is of the overall sound.'"[3]

Just as knowledge of fugal process can enhance our perception of a composition by J. S. Bach, so too can an appreciation of the details of a finely crafted mix enhance our appreciation of a musical recording.

Sound from an Engineer's Perspective

In this section we will consider the sonic properties of music from the perspective of an engineer with the goal that deliberate consideration of these properties will open the door to intentional listening on the part of the reader.

Although there are many elements that work together to form music, the primary elements consist of pitch, intensity, timbre, and rhythm, which is sometimes described as tone duration.[4] In addition to these primary elements, engineers are concerned with the placement of instruments within a virtual soundfield, the acoustical properties of the recording space, and the clarity of a recording. Let's consider how a recording engineer might listen to these elements.

Timbre

An engineer can use any number of tools to change or enhance the tone of an instrument or voice. Sometimes the change can be obvious, such as using EQ to accentuate the airy sound of a ride cymbal, but timbre can also be a subtle and highly subjective consideration—studios may spend thousands of dollars for boutique microphone preamps that add a subtle timbre change—distortion really—to a signal.

Engineers use adjectives such as *warm, brittle, metallic, silky, raw*, or *sterile* when describing the timbre or sound quality of a microphone, preamp, track, or an entire recording. Just as a winemaker strives to perfect the bouquet of a fine wine, engineers strive to record tracks in such a way that the tracks will work together to form a perfect audio bouquet. An important part of this balance comes from the way instruments of varying frequency ranges and timbres are combined, and engineers may subtly (or not so subtly) adjust the tone of an instrument or voice so that it fits better into the mix as a whole.

Intensity

Where musicians focus on relative dynamics, an engineer will be concerned with the relative balance or blend of each instrument in a mix, as well as the dynamic level of a recording in its entirety. Fader adjustments are an obvious part of the process, but dynamic processors such as compressors and limiters may also be implemented.

Location (Pan)

Where intensity refers to the relative level of instruments in a mix, I use the term *location* to refer to the position of instruments in a virtual soundfield. For example, drums might be panned to the left, guitar to the right, and bass in the middle.

Pitch/Frequency

Just as a musician must be concerned with intonation, engineers sometimes use pitch correction plug-ins to fix intonation problems in a vocal performance. Engineers are also very sensitive to the frequency spectrum and try to mix so that energy is relatively evenly distributed throughout the frequency spectrum. EQ is the primary tool used to balance frequencies, and although engineers are very careful about the application of EQ, judicious use of EQ can add body, fullness, sparkle, warmth, clarity, or any number of other attributes to a track or the entire mix.

Timing

In some instances, an engineer might be required to tighten the timing of one or more tracks. This often involves moving a note slightly earlier or later in the timeline, or it might involve quantization of attacks or releases of audio or MIDI data to a grid. Time also comes into play in establishing pre-delay—the amount of time before reverberations are heard—or adjusting the length of reverberation or signal delay. Engineers must also be aware of timing issues that result from synchronization problems.

Dimensionality

I use the term *dimensionality* to refer to the sound of the physical space (real or virtual) in which the recording was made. In most cases, virtual reverb and other processing are added to a signal that will change our perception of the dimensions of the recording space. Important considerations are the size of the room, the reflectivity of the surfaces, and the proximity of the microphone to the sound source and to other instruments.

Technique

A final category, which I call technique, deals with less tangible aspects of a recording such as the overall clarity, as well as potential anomalies such as unwanted distortion, poor signal-to-noise ratio, improper microphone placement, noise, and other problems. In a well-made recording, all of the elements of the project are recorded cleanly and fit together to make a cohesive whole. The flip side is that inattention to detail can compound to create a sonic mess. A good example comes from an experience I had with a musician last year. The individual wanted to stop by my studio to record piano tracks for a band demo. He had already tracked the bass and drums at another location, and he brought his laptop in order to record my piano. I noticed that he didn't take time to carefully check levels or place the microphones to advantage. I also noticed that the bass and drum tracks were rhythmically sloppy. I know that he was pressed for time and intended to fix things in the mix. Needless to say, he called to set up another session because the bass and drum tracks ended up being unusable. Although that is an extreme example, it does highlight the importance of using good recording techniques and striving for excellence at every stage of the recording process, as problems may compound to create insurmountable issues.

Evaluating a Recording: Global Observations

In this section we will utilize active listening to pick a recording apart. My goal is to provide a methodology that will enable you to practice the process because, as I mentioned in the introduction to this chapter, your ability to focus on the individual components of a recording and to assess the recording as a whole can improve with practice. I find that a top-down approach works well. With this method, focus on the "big picture" items first and drill down to the individual details. Using the analogy I presented in the introduction, in this section you will learn to *focus* your audio spotlight.

Listen to the Room

A good starting point is to consider the recording environment or, to use the term I coined earlier, the recording's dimensionality. This may be difficult to discern or simply not applicable for a heavily produced pop or rock recording, but room sound (natural or artificial) will be evident on most acoustic recordings.

EXAMPLE 8.1

By way of example, listen at the companion Web site to Frank Sinatra's recording of "I'll Be Seeing You" from *Frank Sinatra, The Capitol Years*. There is an immediacy to his voice and the orchestra, but also a sense of expansiveness owing to the reverberant quality of the recording. Capitol is well known for the use of special reverb chambers in which engineers would pipe a signal to loudspeakers in an underground chamber and rerecord the processed signal to create a form of natural reverb. It is interesting to attempt to discern the direct sound of the recording room with the reverberant sheen in this recording. Also, compare the spacious sound of full strings and brass with the more intimate sound of double basses and woodwinds. The orchestral recording sounds so natural that it's easy to imagine sitting on the soundstage in the midst of the recording session.

EXAMPLE 8.2

Another example on the companion Web site comes from an Emerson String Quartet recording of the Beethoven's String Quartet No. 12 in E Flat. The reverberance is natural without being obtrusive.

EXAMPLE 8.3

Yet another Web site example comes from the group Tower of Power. Notice the reverberant sound of the drums in the introduction to "Spank-a-Dang." At first you might think that the drums were recorded in a large room, but listen carefully to the snare drum and you can hear that the drum kit must have been recorded in a fairly small space—that is not a bad thing since the sound is pleasing to the ear—but it does illustrate how active listening can yield many insights into the recording process. In this instance, it is evident that the engineer placed microphones close to the drums in a fairly confined space and added artificial reverb to the signal. Also, notice how claustrophobic the baritone saxophone sounds in the entrance after the drum introduction.

Tone Color

Just as it is possible to describe the sound of a clarinet as wooden or a piccolo as shrill, it is usually possible to apply adjectives to a recording as a whole. The color or timbre of the recording is, of course, derived from the individual parts that make up the whole (e.g., a symphonic recording might consist of strings, woodwinds, brass, and percussion), but an engineer also has an effect on the overall color of a recording in the choice and application of microphones, preamps, EQ, reverb, compression, and myriad other tools. Forethought on the part of the engineer is essential if the individual parts of the recording are to work together in a convincing way. Of course, there are as many colors as there are recordings, but I find it a useful exercise to attempt to describe this rather intangible aspect.

EXAMPLE 8.4

On the companion Web site, listen to the excerpt from the quintessential jazz album *Kind of Blue*. There is warmth and presence but also a sense of spaciousness that perfectly complements Miles Davis's melancholic playing.

EXAMPLE 8.5

In the next example on the Web site, notice how the recording has a sparkling quality—the cymbals are almost too crisp—but the effect is very pleasing because of the depth and balance of the recording. The recording is obviously heavily produced, but to my ear, the composite color is very effective.

EXAMPLE 8.6

Where the previous Web site example is heavily produced, notice how Stevie Ray Vaughan's recording of "Pride and Joy" might be described as raw. That is not to say that the engineer didn't pay attention to details; the earthy sound of this recording matches the sound of Vaughan's voice and guitar and makes for a convincing whole.

Dynamics

Dynamics represent another "big picture" consideration. Modern popular recordings are heavily compressed, which serves to give the music a sense of immediacy or being bigger than life, but the downside is that compression can also serve to suck the life out of a recording by reducing or eliminating dynamic contrast. This is largely a question of style—compression is part of the popular music genre while jazz and classical recordings typically feature more dynamic contrast.

EXAMPLE 8.7

A great example of stylistic overcompression can be heard on the companion Web site in Filter's "What's Next." Notice how distortion is held in check—but just barely. It's easy to visualize how individual tracks were heavily compressed, as was the final mix during the mastering phase.

EXAMPLE 8.8

Heavy compression is not just used in rock recordings. In the next Web site example, notice the in-your-face quality of Mariah Carey's ballad, "Bye Bye."

EXAMPLE 8.9

In contrast, listen to the dynamic phrasing that is evident in the vocal line from the Web site excerpt from "Mais Que" Nada by Sergio Mendes and Brazil '66.

As these examples illustrate, an engineer is concerned with dynamics in two ways. First, the engineer will consider the dynamics of individual tracks and apply dynamic processing as necessary. A common example would be the application of compression to a lead vocal so that none of the nuances of the performance are lost in the mix. Second, the engineer will consider the contrast (or lack of contrast) of a recording as a whole and apply dynamic processing as needed to the entire mix. For example, compression and limiting might be applied to a pop recording to make it subjectively sound louder or an expander might be used to give a dynamically static recording more life.

Clarity

I use the term *clarity* to mean the coherence and intelligibility of a recording. Coherence is achieved when the components of a recording work together to make a convincing whole. Intelligibility is achieved when the individual parts are recorded cleanly and mixed well. In some cases, a recording might be coherent but distortion or other problems can detract from the clarity of the individual instruments. On the other hand, individual tracks can be well recorded but might not work together to form a coherent mix. The Duke Ellington excerpt on the companion Web site is coherent—the individual instruments work together in a convincing way—but the recording lacks clarity or intelligibility because of the distortion that is evident in the piano.

EXAMPLE 8.10

EXAMPLE 8.11

Conversely, the individual parts sound good in the Herbie Mann example, but in my estimation, they are less convincing as a whole because the brushes on the snare drum seem slightly loud and serve to distract from the rhythm guitar track. This is a subjective comment, of course, but it does serve to highlight the important relationship between the clarity of the individual parts and the relationship of these parts in the mix as a whole.

EXAMPLE 8.12

One of the things that occurred to me as I became more intentional about focusing on the sound of music and not just notes, rhythms, and phrasing is how many wonderful yet disparate approaches there are to creating a good mix. The Joni Mitchell excerpt on the companion Web site is exceedingly busy—you will hear foreground vocals, background vocals, three guitars, bass, and drums, yet the mix is clear and cohesive albeit busy.

Foreground and Background

In most popular recordings, a single instrument or voice is in the foreground and supported by backing tracks. Although it might seem obvious that a lead vocal or instrumental solo should be in the foreground, it is easy to lose sight of that fact as you work with the details of a recording when trying to perfect the sound of a snare drum or the blend of a horn section. In evaluating a recording, take a moment to let your ear pick out the primary foreground elements and then focus on the backing tracks.

Component Observations

To this point we have focused on listening to the overall sound of a recording. The next step is to focus on the individual instruments or voices that make up the recording.

Instrumentation

It is helpful to list each of the instruments and voices on a sheet of paper. As your ears develop, listen for the individual instruments such as snare, kick, toms, and cymbals that make up a drum kit. With practice you will even be able to hear each hand of a pianist and other details, such as the unique sound of the open strings on a guitar or violin.

In some instances it may be difficult to discern the instrumentation. A well-known example, on the companion Web site, comes from Duke Ellington's "Mood Indigo." For the melody, Ellington utilizes a combination of clarinet, bass clarinet, and muted trombone. Strikingly, the trombone is voiced as the top note and the unique combination of instruments and registers creates a new timbre that is an interesting blend of the individual instruments.

EXAMPLE 8.13

Make a note of the tone of each of the components. For example, an electric bass might sound warm, edgy, wooden, or metallic. By focusing your attention on the individual elements you will be able to start listening as an engineer listens—you will not only hear the timbre of a recording as a whole but you will also hear how the tone of the individual parts contributes to the overall sound.

An excellent example, included on the companion Web site, comes from Pink Floyd's "Us and Them," from the *Dark Side of the Moon* album. As a whole, the mix has a pleasing wash of sound that could be described as warm. However, active listening reveals a unique blend of tone color that might be describes as follows:

EXAMPLE 8.14

- Organ: moderately bright, slightly distorted
- Electric guitar: bright, clean
- Bass guitar: full, warm
- Snare drum: crisp
- Ride cymbal: metallic, vibrant
- Bass drum: round, somewhat indistinct
- Acoustic piano: wooden, somewhat out of tune
- Tenor saxophone: warm low register, somewhat brittle higher register
- Lead vocal: airy, processing evident

Signal Processing

In some instances, an engineer might apply an effect such as reverb, delay, chorus, or phase shift to an individual track. As with the application of EQ, these effects can contribute (or detract) from the mix as a whole. Adding an excessive amount of processing to individual tracks is generally avoided if the goal is to make an "organic" recording, although a judicious amount of processing on individual tracks can be useful in many instances. The David Sanborn excerpt on the companion Web site provides an interesting variety of effects processing on individual tracks.

EXAMPLE 8.15

Relative Position of Instruments

A concert stage provides a helpful model for visualizing the position of instruments in a mix. The drummer of a rock band will likely sit in the middle of the stage behind the lead singer while the bass and guitar players will arrange themselves across the stage. Engineers position elements of a mix in much the same way, although one of the differences is that an engineer might create a mix that is impossible in the real world, such as making a piano sound as if it's as wide as the stage or placing a kick drum and snare in an impossible physical position.

Engineers strive to find a place for each element in the mix. This is usually achieved by considering the frequency ranges of the elements and using pan to place elements that share a similar frequency range in different positions. Although it sometimes works well to mimic the physical position of instruments in a mix, it can be just as effective to experiment with other options. Pan, more than any other element of a mix, can be approached in a variety of ways. On the companion Web site, note the old-school approach to pan in the Marvin Gaye example. In this instance, the drums, rhythm guitar, and percussion are panned to the left side and strings, electric piano, and background vocals are heard on the right side.

EXAMPLE 8.16

EXAMPLE 8.17

A contrasting example can be heard on the Web site's Steely Dan cut, "Do It Again." Here, the drums are almost impossibly wide. Listen (with headphones) to the ride cymbal on the right, snare slightly left, and crash far left. Clearly, there are many ways to approach the placement of instruments and vocals in a mix.

Relative Level

The balance of each instrument is obviously an important consideration. Engineers carefully adjust the relative position and level of individual tracks so that background and middleground elements stay behind any foreground elements. To complicate matters, there is an intrinsic relationship between the position of instruments in the stereo field and perceived loudness. Often, a slight change in pan position will be enough to bring an element forward (or backward) in the mix and the same, too, could be said of judicious application of EQ. So, in evaluating a mix, I would suggest noting both the position and the relative level of each track. An excellent example of this relationship comes from "Street Dreams 4" from the Lyle Mays album, *Street Dreams,* as included on the companion Web site. Although the composition is very repetitive, there are many sonic elements that combine to form an interesting soundscape. It is evident that a great deal of attention was paid to the details of this mix. Though the music is complex and consists of many different elements, each element has its own space and has been balanced to contribute to the overall sonic texture without being overly cluttered.

EXAMPLE 8.18

Sonic Idiosyncrasies

I also find it helpful to listen for instrumental and vocal idiosyncrasies when focusing on the tone quality of the individual components of a mix. By idiosyncrasy, I mean the little details that are an important part of an instrument's sonic signature. Examples would include breaths and sibilance of a vocalist, fret noise on a guitar, or the sound of keys or the damper mechanism in a piano. Developing sensitivity to these details can be very useful during the mixing and mastering process. It is also wise to pay close attention to these details when tracking, as some problems (such as an undue amount of fret noise) can create insurmountable problems during the mixing and mastering process. Listen to the subtle sound of a piano dampening mechanism in the Web site example 8.19.

EXAMPLE 8.19

Dynamics

In a previous section we considered the sonic effect of dynamic processing when applied to a mix as a whole. It is also helpful to consider the dynamic range of individual tracks. Mature performers tend to use dynamics to shape phrases so that a performance is not static or lacking in emotion, but tools such as compressors can remove or limit this type of dynamic nuance. That is not to say that dynamic processors should not be used—that choice is determined by the genre and artistic sensibilities of the engineer and producer. Dynamic processors *are* useful in many contexts, but I have noticed in my own work that, as I use compression to exert a measure of control over a track, it is very easy to squash the musicality from a performance. Consider the dynamic range of the guitar solo in each of the Web site guitar solo examples. In the first example, compression was used and each of the notes has a similar dynamic level. Note how, in the second example, more dynamic contrast is evident in the recording:

EXAMPLE 8.20

Sonic Anomalies

There are any number of problems that can serve to blur the clarity of a mix. These anomalies might be evident on the global level (e.g., distortion that is evident in a final mix) or on the track level (e.g., amplifier hum or noisy fret board). Although some of the items listed below, such as distortion, can be used in a positive way, it is generally best to guard against these types of problems.

Noise

Noise can take on many forms, such as the hum of a guitar amp or a buzzing string. It may also be the result of irregularities in an electrical signal or clicks and pops that result from digital editing. In some instances, extraneous noise can make an instrument sound more authentic, such as the mechanical sound of a whirling Leslie speaker. In any case, it is helpful to practice identifying such sounds in order to determine if a noise gate or other remedy is necessary. Notice the extraneous noise emanating from Jimi Hendrix's amplifier and the distortion evident in the bass cabinet in the Web site example. These sounds might be considered an authentic part of the Hendrix "experience" and certainly provide a sense of rawness to the recording.

EXAMPLE 8.21

Compression Pumping

When a large amount of compression is added to a track, a phenomenon called *compression pumping* can occur. The effect can be subtle or extreme, but it is usually undesirable. Listen for compression pumping in the Web site example.

EXAMPLE 8.22

Distortion

Distortion is obviously an important part of popular music—rock wouldn't be rock without the characteristic sound of distorted guitars. What might be less evident is that distortion is sometimes purposely used to add "grit" to other types of tracks or even to a mix as a whole. Notice, on the companion Web site, how distortion is an integral part of the final mix of Carey's performance of "Migrate."

EXAMPLE 8.23

Distortion can take on many undesirable forms. It can occur as the result of an overdriven channel input, the inability of a microphone to deal with high sound-pressure levels, problems with synchronization between digital devices, or artifacts in a long reverb tail. The Web clips demonstrate two common problems.

EXAMPLE 8.24

EXAMPLE 8.25

Conclusion

We have covered many concepts in this chapter, and it is my goal that this discussion will help you become more proactive in developing critical listening skills. It is also my intent that the concepts presented in this chapter will provide specific steps that will help you unlock your golden ears. To that end, you will find a recording evaluation sheet in the appendix that can be used as a starting point. The process can also be useful in evaluating your own work—something that can be difficult for most of us to do without a bit of objectivity.

I have mentioned this several times in the course of this book, but it bears repeating: your ability to hear and the musical sensibilities you bring to a project are more important than the equipment or software that you use—assuming your audio tools provide a reasonably professional level of performance. For this reason, this chapter on active listening might be the most important one in the book. I would reiterate the importance of developing critical listening skills and stress that these skills *can* be developed over time through intentional practice. Although this type of detailed approach to listening might seem to take some of the fun out of listening, I have found the opposite to be true. As my ears continue to develop I enjoy hearing new things in recordings, and my emerging interest in the sonic qualities of recordings has given me a new perspective on music.

9

Recording Tracks

Up to this point, we have primarily focused on equipment such as a mixer, computer, and microphones. Now comes the fun part—the creative application of these tools in the context of making music. This chapter will provide a number of tips that will help your sessions run smoothly and it will conclude with a step-by-step description of a typical recording session.

Preplanning

One of the keys to a successful recording session is preplanning. Unexpected situations can and will arise in the studio, but many of these problems can be anticipated in the planning stage. Ideally, the session will run smoothly and you will be able to anticipate the needs of the musicians. A good first step is to devise realistic goals for the session. Musicians sometimes have unrealistic expectations with regard to the recording process, and they may not understand the amount of time it will take to position microphones and set good levels. Drums, in particular, can take a great deal of time for load-in, setup, and microphone placement, so I always request that the drummer show up an at least an hour before the other musicians to allow sufficient time to position the microphones and set levels. Also consider the abilities of the musicians you intend to record when establishing session goals. For example, an amateur vocalist or musician may want to do numerous retakes while a seasoned professional may record his or her part in a single take.

Also consider the type of music you intend to record. For example, most pop recordings involve the layering of many independent tracks while jazz or classical musicians will want to record their parts simultaneously. Even though pop recordings may involve recording instruments individually, it is sometimes advantageous for several musicians to play at the same time for a more live feel. For example, on one recent session I had the vocalist sing a scratch track in the control room while I tracked the bass and drums. This enabled the musicians to play more musically while listening and responding to the vocalist, but the vocalist was able to retake her tracks once the backing tracks were complete.

I find templates to be an essential part of any recording session. Nothing wrecks a studio vibe more than having to wait for an engineer to configure the software for recording a new song or worse—having to stop a good take because not all of the tracks were armed for recording. To this end I always take time to create a template for a given session. The template should include appropriate track names, reasonable default values for level and pan, and possibly a few essential inserts.

If you are to the point of recording a band, then you are likely fluent with your DAW software. However, it is helpful to anticipate the needs of other musicians and learn the key commands that will help you function fluently. A case in point: I rarely use the punch-in function of my recording software. I don't say that to brag; it's usually just faster for me to rerecord a long phrase or even an entire song than to try to match a performance and make a convincing punch-in. However, many musicians prefer to fix a phrase by punching-in on an otherwise perfect take. It made sense for me to learn how to efficiently configure my software for punch-ins in case a session player asks to rerecord a phrase. Other common requests might include changing the headphone mix, adjusting the level or sound of a click track, and fixing an intonation problem or a wrong note.

Planning the Session

One of the joys of owning your own project studio is that you do not need to be as concerned with the clock. I have certainly spent many blissful hours exploring microphone placement and stereo microphone techniques. However, I have noticed that musicians tend to have a few golden hours within which they will do their best work. After a time, nerves, fatigue, or any number of other factors will get in the way if a session runs too long. For this reason, it is a good idea to establish a rough timeline for the session. The outline can help to keep the players on track when they invariably want to break the flow of the session by listening to takes.

Equipment Check

I prefer to set up microphones and check my gear well in advance of a session. I consider the probable placement and orientation of each of the instruments, and I place microphones and stands in rough proximity to each player. I then run microphone cables (trying to keep the cables out of the path of the musicians) and make sure that each microphone and cable is working. This is also a good time to set up a headphone distribution amplifier and make sure that headphones are working and within reach of each musician.

File Management

Although file management is not a glamorous topic, it's an important aspect of digital audio recording. Multitrack recording requires many gigabytes of hard-drive space, and it's easy to lose track of important files without a file-management plan. First, make sure that plenty of drive space is available on your recording hard drive. Before starting a new recording session I find it helpful to make a folder with the name of the band and the date—for example, C:\BAND_NAME_06-01-09.

As the session progresses, I make sure that each new project file is created and stored in the appropriate client directory. I am also in the habit of appending a version number to the project name, such as SONG_NAME_v1b. This allows me to quickly find the most recent version of a project and provides a method for documenting revisions during the editing and mixing process.

As mentioned previously in the book, it is important to back up important files. If time permits, I like to burn all of the session files to an archival DVD and hand the DVD to a band member before they leave. This is a convenient way to ensure that a copy of the files is stored off-site.

Session Documentation

I want to stress the importance of session documentation. When I first started recording, I was more concerned with microphone placement and input levels—important aspects of recording to be sure, but I didn't take time to document sessions. A downside is that I missed out on an excellent learning strategy. Many years ago I learned to brew beer from a book, and the author suggested keeping detailed notes of the brewing process, as well as observations about the final results. Nothing could be more appropriate for a "home brew" recordist. Now I keep detailed notes not only about good takes and bad takes but also about microphone placement, sample rate, and many other details. I am even in the habit of snapping a few digital photos to document the exact position of microphones and instruments used in a given session.

A database application can simplify the process of session documentation, and many applications will allow you to store digital photos with your session notes. Database applications such as Bento are available for a modest cost. Open Office is a free suite of programs that includes a database application. The suite is available at http://www.openoffice.org.

The Studio Vibe

Musicians will play and sing well when they are happy and comfortable. Conversely, a session can quickly go awry if technical problems, poor attitude, or some other problem gets in the way. A typical situation is that an amateur musician is nervous about recording and becomes frustrated during the recording process. Many years ago I had the pleasure of doing a number of recording sessions with Grammy winning engineer and producer Steven Heller. One of the many things I learned from Steven was how to gently provide direction to an artist without making him or her feel self-conscious. It is sometimes necessary to provide some gentle guidance to a band or performer if you sense that a session is not going well.

Recording Tracks

Before recording, consider an appropriate sample rate and bit depth. I always enable twenty-four-bit recording on my system because there is an audible difference between twenty-four- and sixteen-bit audio. Higher sample rates may or may not be advantageous since it may tax your recording system and unduly limit track count. Although I sometimes use the 192 kHz setting when recording solo piano, I typically use a lower setting when recording multiple tracks.

Establishing a Good Level

Establishing an appropriate input level is one of the single most important aspects of the recording process. An input level that is too hot will cause distortion and a signal that is weak will make for an inferior recording since more noise will be evident when the signal is boosted. A good starting point is to set channel faders on your mixer to *unity gain*. The unity setting means that the faders are not boosting or reducing a signal. Then, use the input trim knob to adjust the gain so that the signal is at an appropriate level.

I find it helpful to solo each channel in turn while setting levels. For an instrumentalist, ask the musician to perform something that utilizes the full range of the instrument at varying dynamic levels. Adjust the input trim on your mixer or audio interface so that it is just shy of clipping. During this stage it is essential to listen critically for any audio anomalies such as problems with the signal that might indicate a bad cable or dirty trim knob or channel strip. Also, listen for problems with the sound source such as a rattling bass pickup, noisy bass drum pedal, or other problems. These types of problems are often magnified when recorded so it's essential to find and fix these before they are recorded to disk. This makes sense when you consider that a closely placed microphone is essentially the same as placing your head near an instrument.

Along these lines, I once recorded a solo piano album with a pair of AKG 414s. The piano sounded great from the player's position, but the faintest anomaly could be heard in the recording. I took the time to track the problem and eventually found that a paperclip was hanging from one of the highest strings of the piano. A subtle problem but a significant one, to be sure.

It is also helpful to listen for problems in the room itself since an instrument or voice may create sympathetic vibrations that cause a light fixture or other furnishing to vibrate in an audible way. For example, metal music stands can resonate and produce an audible pitch if a singer sings at a particular frequency and amplitude. Other innocuous items such as the handle on a guitar case or a wristwatch can create sounds so it is helpful to be aware of potential problems before they show up in a recording.

Not only are levels important, it is also important to listen critically to the *sound* of each instrument and voice. In most instances, it will be necessary to adjust the position of a microphone or even swap a microphone if you are not pleased with the sound. Often, a change of a few inches in the placement of a microphone will provide a dramatic improvement. Through experimentation and experience, you will learn the strengths and weaknesses of your microphones so don't be afraid to experiment if time permits. For me, one of the real joys of home recording comes from being able to take the time to learn and improve by experimenting with microphones and their placement.

The most helpful advice I can share is *to take the time to ensure that each instrument or voice sounds great prior to recording*. Digital audio technology offers a virtually unlimited potential to edit and otherwise tweak a recording, so it is easy to take a cavalier attitude toward microphone placement. I speak from experience when I say that it is a bad idea to rely on your DAW in this way. Problems such as low input level, poor microphone placement, and distortion can't really be fixed. The mixing process will be an exercise in futility if tracks do not sound good to start with, so any time devoted to improving the placement of microphones is time well spent. Don't be afraid to make a drastic change if you are not pleased with what you are hearing in the control room—even if it means a delay for the musicians.

Dealing with Leakage

Once levels have been set for each channel, you might want to ask the band to play a section of music at full volume. This is the time to listen for problems with leakage that can occur when an instrument bleeds into another microphone. The best way to check is to simply solo each channel in turn and listen for bleed. The goal is not to prevent all leakage but to minimize leakage so that it is not excessive. At this point it may be necessary to re-position a player and/or microphone or utilize baffles. In some situations, a padded moving blanket can be used to isolate a piano or guitar amplifier.

Dealing with Phase Cancellation

Phase cancellation can be another problem when multiple microphones are used in the same recording space. Engineers sometimes use the word *phasey* to describe this problem, in which a signal may sound unnaturally hollow or weak. Problems can arise from the interaction of sound waves or with wiring. For example, electrical phase cancellation will occur if two microphones are used to record a source but the polarity of one of the cables is reversed. Phase cancellation can also occur when two microphones are placed at differing distances from a sound source. Some phasing will occur when two or more microphones are used but the goal is to ensure that phase cancellation is not extreme. Most DAWs provide a plug-in to adjust the phase of a signal on a given channel. Such a signal inversion function is typically a component of a gain or input utility plug-in.

When checking for phase issues, it is helpful to make frequent use of the mono switch on a monitor mixer or the main bus of your DAW. For example, solo the drum tracks and listen carefully to see if the sound becomes thinner in mono. If so, try inverting the phase of one of the overhead microphones or change its position. If the overheads sound good, check the other microphones in the kit in relation to the overheads both in mono and in stereo.

Sending a Mix to the Musicians

Musicians will need to wear headphones if they are recording in isolation from one another or adding tracks to an existing recording. Headphones may also be necessary when miking an acoustic instrument such as an acoustic bass so that other musicians can hear the instrument at an appropriate level during the tracking process.

A headphone distribution amplifier is a helpful device that provides a way for each musician to adjust the level of his or her headphones. Products such as Mackie's Big Knob provide a convenient way to route a stereo signal to several destinations, such as the headphone amplifier, speakers, and a CD recorder (see figure 9.1). In this instance, the stereo output of an audio interface or mixer is connected to the Big Knob, which provides multiple outputs for connecting to reference monitors, CD burner, and the headphone distribution amplifier.

FIGURE 9.1

Mackie Big Knob

It's usually a good idea to add a bit of reverb to the headphone mix because it can be awkward for musicians to perform when they hear an extremely dry signal. I have noticed that singers are particularly apt to want a fairly wet signal. Along these lines, it may be necessary to send a different mix to the musicians than the mix you are listening to in the control room so it's a good idea to plan for this contingency. If your interface provides multiple pairs of outputs, you can configure your software so that one pair functions as a dedicated headphone mix. You can use individual sends on each virtual channel in your DAW software to feed the headphone outputs. Notice how individual sends are used to feed a second pair of outputs, shown in figure 9.2. Also notice how a reverb plug-in has been inserted into the dedicated headphone bus.

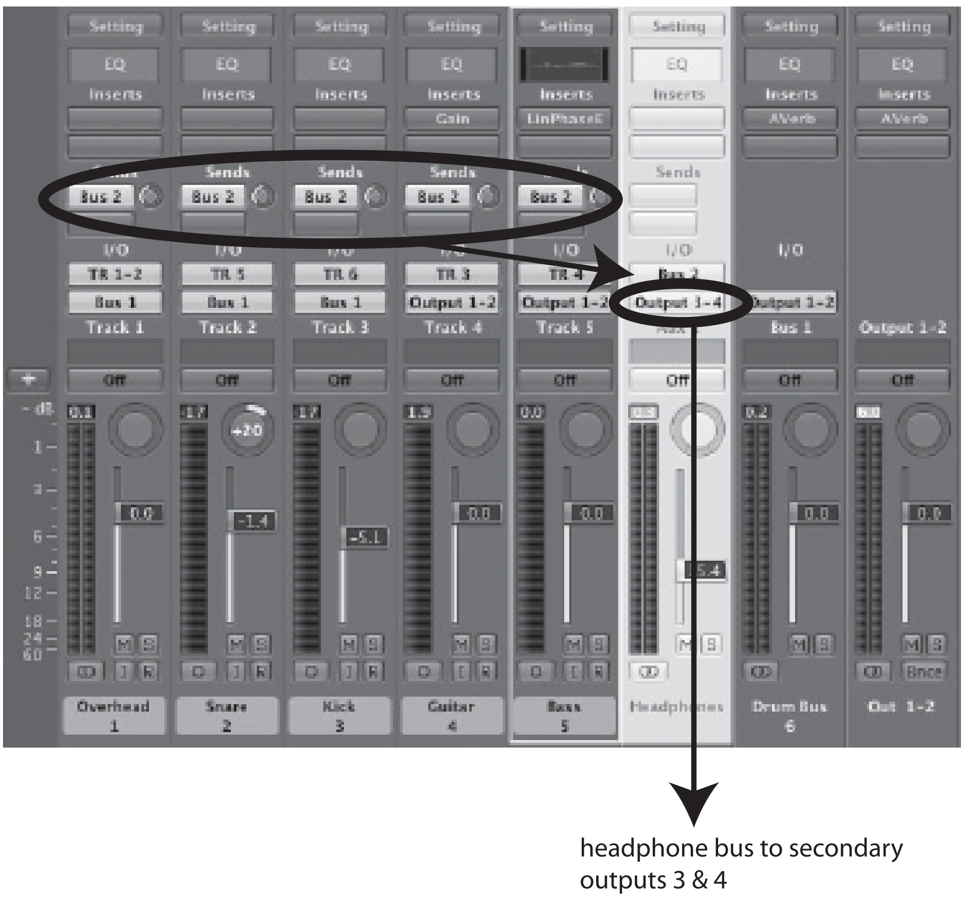


FIGURE 9.2

Using virtual sends to create a headphone mix

Talkback

You will likely want to set up a talkback microphone to facilitate communication with the musicians during the tracking process. Some studio command systems, such as the Big Knob mentioned previously, have a built-in talkback microphone that can be routed to the headphone mix. Many mixing consoles also have a talkback microphone that can be used for the purpose. In lieu of a dedicated talkback system, you can simply plug a talkback microphone into an empty channel on the mixing console or audio interface and route the signal to the appropriate headphone bus.

Laying Down Tracks

Once microphones have been placed, input levels are adjusted for each channel, and musicians are comfortable with their headphone mix, it's time to begin the recording process. Although it might seem obvious, make sure that each relevant track is armed for recording. This is so important that it's usually the *first* thing I do when I load a template in preparation for recording.

Although it can be tempting to sit back and enjoy the music during the tracking process, it is essential to be a particularly active listener because many problems can arise. A musician might move in relation to the position of the microphone, input gain might need to be adjusted as the players relax and get into the music, or unwanted external noise such as a siren or garbage truck can interfere.

It may be helpful to add markers to important structural points during the recording process. Consider assigning a convenient key command so that markers can be added on the fly. This is another example of anticipating the needs of the musicians. It's likely, for example, that a soloist will want to punch in a new solo or that the vocalist will want to do another take of the bridge. Markers make it easy to quickly return to these locations in the song for a punch-in or overdub.

Using a Click Track

It's important to consider the benefits and disadvantages of using a click track for a given recording. Although it is possible to add a click to a track after recording through a process called *beat mapping*, it's always best to start with a steady click if you intend to use prerecorded loops or if you want to be able to quantize MIDI tracks such as a MIDI keyboard or drum part. Although it might seem irrelevant, take the time to learn how to adjust the sound and level of the click because musicians can be very sensitive to the timbre and volume of the click—it's no fun to sing an emotive pop ballad while listening to an obnoxious cowbell.

You will generally want to avoid using a click track for more organic genres such as jazz or folk. However, it may be necessary to provide one or more measures of count-in if musicians will be asked to overdub at the start of a track that has been recorded without a click track or audible count-in. In this instance, a count-in can be quickly generated with an available MIDI track. Arm the MIDI track and start playback of the song. You will likely hear a clear tempo by the end of the first or second bar, at which point you can use a MIDI keyboard to record a full measure of beat pulses plus one extra beat (or use a microphone to record a count in on an available track). Cut the clip and paste it so that the extra beat (e.g., the fifth beat in 4/4 time) aligns with the first beat of the first measure.

Strategies for Trouble-Free Recording

Recording need not be a stressful activity. Ideally, the musicians will enjoy the process and make great music, and you will record a great-sounding performance. Consider the following strategies:

- Preplan the session
- Prepare musical charts, if necessary
- Allow ample time for setup
- Listen to each instrument in the recording room prior to placing a microphone
- Take time to place microphones to best advantage
- Set appropriate input levels
- Don't record until everything sounds good
- Record to a secondary hard drive
- Ensure that plenty of hard disk space is available
- Run a minimal number of plug-ins while recording
- Have extra equipment such as cables, microphones, and stands on hand
- Save often
- Make sure your computer is well ventilated
- Don't forget to arm tracks
- Turn off cell phones
- Keep a positive vibe

Tips for Music Educators

Band, orchestra, and choir directors will face a unique set of challenges when recording an ensemble. Although many of the concepts presented in the last section are applicable, educators will typically want to capture the live sound of an ensemble without overdubbing. For this reason, the selection and placement of microphones is even more critical.

Consider the Recording Space

A first step is to consider the recording space in order to determine how far you wish to place the microphones from an ensemble. If the acoustics are good, then it will be possible to move the microphones away from the ensemble to capture the sound of the recital hall or rehearsal space. Alternatively, it may be advantageous to move the microphones closer to the ensemble and add artificial reverb to the final mix. This is particularly true in situations where unwanted noise from lights or a heating system are problematic.

The proportions of a room may also determine the position of microphones. Microphones will likely need to be placed fairly close to an ensemble in a rehearsal room so it may not be possible to capture the entire group with microphones configured in a coincident or near-coincident pair. In this case, a spaced pair might be a better option. As described in chapter 4, the combination of a center microphone (or stereo pair) and a spaced pair can yield excellent results.

Microphone Selection

As I mentioned in chapters 1 and 4, a matched pair of quality condenser microphones will typically be the best choice for ensemble recording. Ribbon microphones can also work well, although they are fragile and, thus, are not recommended given the potential for mishandling in a school environment.

The selection of a polar pattern is also a consideration. Omnidirectional microphones are a good choice if the acoustics of the recording space are excellent, but cardioid or even bidirectional microphones may be a better choice in some instances. For example, a cardioid microphone can help to minimize audience noise or audio anomalies in a recital hall. A multipattern microphone is a good choice for many educators since it will provide an opportunity to experiment with polar settings in order to find the best solution for a variety of recording spaces.

Microphone Placement

Several microphone placement strategies are presented in chapter 4, but a brief review is appropriate at this point. Consider starting with a spaced pair of omni or cardioid condenser microphones. Listen closely to determine the ideal distance from the ensemble. This is obviously a subjective judgment, but the goal is to find a distance that provides a pleasing balance of direct and reverberant sound.

The distance between microphones should also be determined by critical listening. Larger distances will create a bigger stereo image but can result in a weak middle. A center microphone (or stereo pair) can be useful in this instance. Also take the time to experiment with the height of the microphones. In general, it will sound best to place microphones well above a choir or orchestra.

Listen

Ideally, a noncritical rehearsal will provide the best opportunity to experiment with microphone placement. Document each new position by keeping a log or describing the setup at the start of a take. Spot-check each take with a pair of good headphones and make adjustments to the position of the microphones as necessary.

Use an Assistant

Consider teaching one or two students to use your recording hardware or software. It can be very challenging to function as both a director and a recordist, and students can obviously benefit from an introduction to the recording arts. The students can help log the session and, with some instruction, they will be able to listen for problems such as signal distortion, a car horn, or slamming door.

Recording

A multitrack recorder will be a good choice for many educators. Although excellent recordings can be made with a quality stereo recorder, a multitrack recorder (either hardware or software) can provide more opportunities for experimentation and will be more forgiving with regard to microphone placement since the relative level of multiple microphones can be adjusted *after* a recording session.

As with a project studio session, ensure that microphones are positioned to best advantage and that the input gain of each track is set to an appropriate level. Be sure to arm each track prior to engaging the record button.

In the last section, I listed a number of strategies for trouble-free recording and those strategies are also relevant to ensemble recording. The following tips are geared specifically to music educators:

- Tape signs on each door to warn students and faculty that a recording session is in progress.
- Remind students to wait until the sound of the last note has faded away before they talk or move.
- Print a session sheet to keep track of recordings and retakes as the session unfolds.
- Don't rely on meters: spot-check takes with a pair of headphones.
- Be sure to arm tracks and engage record prior to the start of a concert.
- Consider conducting frequent recording sessions with students so they can learn to be relaxed and perform well during a recording session.

Case Study: Anatomy of a Recording Session

I teach at a small university and was asked by a colleague to record our student jazz ensemble. This provided a wonderful opportunity for a "Recording on a Budget" case study, as our selection of microphones and other equipment is fairly modest. We had access to a number of Shure SM57 and SM58 microphones, and I supplemented these microphones with a few relatively inexpensive small-diaphragm condenser microphones. The department was able to purchase an Allen and Heath mixing console and Alesis HD24 recorder, which enabled me to record the band with twenty-four simultaneous tracks. Although the mixing console and recorder are by no means budget items, the selection of microphones was modest and is similar to the selection of microphones that might be available to a home recordist. A more economical approach would have been to record the ensemble with fewer microphones and tracks or utilize two microphones in a stereo configuration with accent microphones as needed. Yet another option would have been to record and mix multiple tracks directly to a two-track CD burner or DAT machine.

A first step was to preplan the session and determine how microphones would be assigned. Our performance hall is not a pristine recording room, so I decided to utilize close microphone placement for each of the woodwinds and brass. I elected to use two overhead microphones on the drums, as well as dedicated microphones for the snare and bass drum. One channel was assigned to the guitar amp and two for the piano. I used a DI on the bass since I was worried about getting a good acoustic sound, given the close proximity of drums and brass. In hindsight, I wish I had placed a microphone on the bass and surrounded it with baffles because I was not pleased with the direct sound of the instrument. The remaining channel was designated as a solo microphone.

There are advantages and disadvantages to utilizing multiple microphones in this type of scenario. Individual tracks make it possible to create a more produced sound during the mixing phase, but multiple microphones can be a disadvantage if the goal is to create an organic sound. Given the acoustic problems in the recording space, the use of multiple closely placed microphones was a good choice for this project.

I set up the mixing console and recorder in an adjacent hallway and ran a microphone snake to the performance hall. I was careful to plug microphones to their prearranged channel assignment because I knew the session would be hectic once the band arrived. I lowered the piano lid to quarter-stick and positioned the microphones inside the lid of the piano. A padded moving blanket provided a measure of isolation for the instrument.

I placed a pair of small-diaphragm condenser microphones in an XY pattern over the drum set and positioned a dynamic microphone near the snare and another microphone in front of the kick drum. The drummer arrived early so that I would have an opportunity to listen to the drums, adjust the microphones, and set levels prior to the arrival of the full band.

Once the members of the ensemble arrived, it was time to check each channel. I asked each player to play a short phrase at full volume and listened closely while adjusting input gain. It was necessary to make a few adjustments to the placement of the microphones and to talk to the students about keeping a consistent position in front of their microphone. I then asked the band to play a short excerpt for a test recording. I listened closely to the recording and asked the director to check the recording as well. At this point it was time to record the CD.

About halfway through the session I noticed that the bass signal vanished —this was particularly insidious since I was trying to focus on twenty-three other tracks. I had to stop the recording and find the problem—a temperamental instrument cable that the bassist used to connect to the DI box. Other than that problem, the session ran smoothly and the ensemble was able to record a number of selections.

Immediately following the session I backed up all of the data to a secondary drive and then began the process of transferring files to a computer for mixing. Although it would be possible to mix the project with the Allen and Heath console and some outboard equipment, a DAW offers many features and conveniences so I elected to utilize software for that stage of the process. I imported the tracks from one of the recordings into our DAW software and created a rough mix. For convenience, I named each track and routed each section of the ensemble to a sub bus. Once I was happy with the mix, I used the settings of the project to create a project template. The template was used to create project files for the other recordings and provided the director with a convenient workspace to mix the remaining selections.

A different ensemble would obviously present a different set of challenges and considerations in terms of recording, but it is my hope that the previous discussion will be helpful in providing an overview of a typical session. I would invite you to listen to excerpts of the recording at the companion Web site. As you will hear, there are problems with the recording but also many positive aspects, and I hope that the recording will provide some encouragement for your own recording endeavors. As a case study, the jazz band recording project demonstrates that it is possible to use a modest selection of microphones to create a good recording. A home studio could be equipped with fewer channels of comparable equipment for a modest cost. For example, a sixteen-channel Allen and Heath MixWizard 3 utilizes the same preamps as the larger format console but costs approximately $1000.

Conclusion

I will conclude this chapter by stressing that preplanning and flexibility are two of the key aspects of a successful recording session. It is also important to take the time to properly position microphones and set good levels. Note that, in this context, the word *properly* means placing a microphone such that it will yield the best sound. There are many things you can do in terms of mixing with tracks that have been cleanly recorded, but there is very little you can do with tracks that sound bad to start with. For this reason, don't feel awkward about taking the time to swap microphones or reposition players if you feel it will improve the sound.

10

Processing Signals

This chapter deals with the manipulation of signals in the digital domain. In a typical project, multiple tracks are recorded to disk and the engineer will, as necessary, utilize a broad palette of signal-processing tools to alter the sound of a single track or the mix as a whole. In this chapter, you will learn about a wide variety of signal-processing tools. You will learn how the tools work and how to incorporate them into a mix.

Before we begin I would offer one bit of advice. Voltaire stated that "the better is the enemy of the good," and that advice is applicable to the use of signal processors and mixing in general. A personal anecdote might be useful to clarify this comment: Several years ago I was hired by a vocalist to record some keyboard tracks at a local studio. After I completed the tracks, I went into the control room to listen. The singer sounded good to my ear, and I listened as the engineer went to work on the track. I noticed that, after a great deal of tweaking on the part of the engineer, the signal processing he utilized on the singer's track was detrimental to her voice. I recognized the problem immediately because I have (and do) suffer from the same affliction—a tendency to want to be proactive in making something that sounds good sound even better. I'm sure that's not always a bad trait to have—audio engineering does require attention to detail—it's just that the process of tweaking a mix can sometimes be detrimental. My best advice is to try to retain your vision for a project while you focus on minutiae and be cognizant of the fact that you probably won't need to do much to a track if it consists of a strong performance that is well recorded.

One of the challenges in mixing is to know which tool to use to fix a perceived flaw in a mix. Complicating matters, there are often a number of ways to achieve a similar end. While it might seem obvious that an adjustment to a level fader is the way to fix a level imbalance, pan and EQ can also affect our perception of loudness for a given track. I mention this in order to highlight the interdependent nature of these tools.

Signal Processing

Most of the tools described in this chapter are in the form of virtual plug-ins. Depending on the type of tool, these plug-ins might function as a channel insert, or they might be used to process a sub bus or even the main bus (more on this later in the chapter). Note that hardware versions of these tools are also available, but the application of these concepts is largely the same for hardware processors and their virtual counterparts.

Signal processors can generally be divided into three broad categories: dynamics, equalization, and effects. We will also consider some specialty tools such as pitch correctors and amplifier simulators.

Dynamic Processors

Dynamic processors are used to control the level of individual tracks or an entire mix. They can be used to increase the perceived loudness of a recording or to make a track more dynamically manageable.

Compressor

A common use for a compressor is to tame or tighten a track in which the dynamic range is too great. For example, a lead vocalist might take a boisterous approach to one phrase and a subdued approach to another. It might be impractical to ride the fader level in order to bring the subdued phrase up in the mix without letting the boisterous section overpower. A compressor can, to an extent, automate this process. Compression has earned a bad reputation since some engineers feel that contemporary popular music is overcompressed and lacking in dynamic nuance. However, some amount of compression is necessary so that a recording will retain its clarity on a variety of playback systems. Interestingly, our ears even compress sound. Neuroscientist Daniel J. Levetin writes : "The inner hair cells have a dynamic range of 50 decibels (dB) and yet we can hear over a 120 dB dynamic range. For every 4 dB increase in sound level, a 1 dB increase is transmitted to the inner hair cells."[1]

Compression can be counterintuitive at first glance. A subjective description of the application of compression to a recording is that it "makes things sound louder." However, a compressor actually works by lowering or attenuating levels. This seeming contradiction can be understood by considering the example of a singer whose performance is too dynamic to work with a given mix. Where an adjustment to a fader level would increase the volume of the soft phrases *and* louder phrases, a compressor puts a limit on louder signals while providing a mechanism to increase the level of softer signals. Simply put, there will be less dynamic contrast between loud and soft passages when compression is applied. Figure 10.1 will help to clarify this concept.

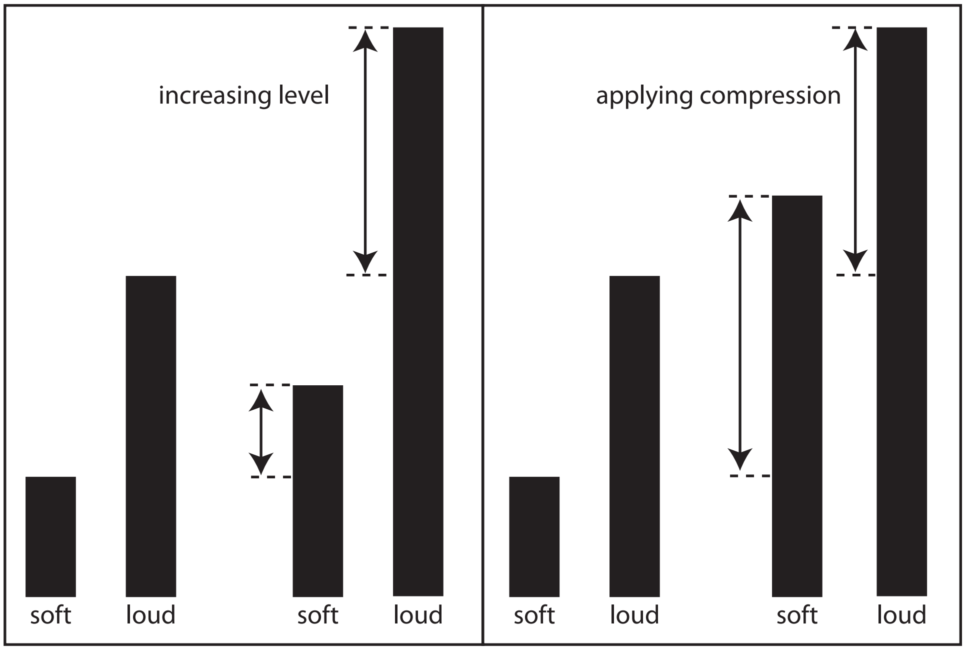


FIGURE 10.1

Compressing a signal vs. increasing overall level

Components of a Compressor

The four primary components of a compressor are compression threshold, compression ratio, attack time, and release time (see figure 10.2). Each of these controls is interdependent so it can take time to develop a feel for the adjustments that are necessary to compress a track to best advantage.

FIGURE 10.2

Simple compressor

Compression Threshold

Compression threshold refers to the input level at which compression will occur. When the threshold is reached, the signal is reduced by the compressor. The compressor will not respond to levels below the threshold so a high-threshold setting may have the effect of turning the compressor off.

Compression Ratio

When a signal reaches the compression threshold, compression is applied and the amount of compression is determined by the compression ratio. Higher ratios provide more compression and less dynamic contrast. By way of example, a compression ratio of 2:1 means that the output will increase 1 dB for every 2 dB increase in the input signal. A 4:1 ratio means that it will take a 4 dB increase in the input to increase the output by the same 1 dB.

Attack Time

You can think of *attack time* as a compressor's reaction rate. Higher values mean that it will take more time for the compressor to respond once the threshold level has been reached. Longer attack times will retain the initial attack of an instrument or voice but may cause the compressor to be ineffectual. Shorter attack times are useful in maximizing signal level but may have a detrimental effect on tone. As with other compression parameters, attack time must be carefully adjusted to fit a given input signal.

Release Time

A compressor will stop compressing a signal when the input falls below the compression threshold. *Release time* refers to the number of milliseconds it takes for the compressor to return to its normal state.

Knee

Some compressors provide an additional parameter called the compression *knee*. The knee relates to the way the compressor functions as an input signal approaches the compression threshold. In a hard-knee compressor, compression is immediately (and fully) applied when the threshold is reached. In contrast, a soft-knee compressor gradually applies compression as a signal approaches the threshold. As Bob Katz states in his *Mastering Audio* book, "soft knee can sweeten the sound of a compressor near the threshold. For those models of compressors that only have hard knees, some of the effect of a soft knee can be simulated by lessening the ratio or raising the threshold, which will result in less action."[2]

EXAMPLE 10.1

The samples on the companion Web site will help to illustrate some of the common applications of a compressor. In the first example, a subtle amount of compression is applied to an electric fretless bass. Notice how the compressed track has a tighter sound because the dynamics are more consistent.

EXAMPLE 10.2

In the next example, compression is used to make a snare drum "pop." In this instance, the compression ratio is fairly high and a touch of reverb and EQ is added.

EXAMPLE 10.3

Compression is particularly useful when applied to a vocal track. In the third example, the singer uses many subtle nuances that might be lost in the mix without the application of compression.

Multiband Compression

A multiband compressor is a useful tool that provides the capability to apply compression independently to specific bands of frequencies. For example, compression might be applied to the band of frequencies from 5 kHz to 20 kHz in order to impart a sparkling quality to a recording without affecting other bands of frequencies where compression might have a detrimental effect. Similarly, a multiband compressor could be used to tighten lower frequencies without

EXAMPLE 10.4

making the recording overly bright or sibilant. Notice how much more vibrant the next musical excerpt sounds. This is due to the way multiband compression is utilized on the main bus.

Limiter

A limiter essentially functions as a compressor at an extremely high ratio. Whereas a compressor compresses signals proportionally, the limiter reduces any signal over the threshold to the value of the threshold. As such, a limiter is typically used as the last processor in a main mix to keep signals from exceeding 0 dB.

Limiters can also be useful during the tracking phase. Although it is not possible to use a limiter to remove distortion that is recorded to a track, a limiter *can* prevent a loud attack from overdriving a channel. In the next example, a limiter is applied to a bass track to prevent the small amount of distortion that is heard when the signal overdrives the channel.

EXAMPLE 10.5

Expander

An expander is similar to a compressor, but instead of reducing dynamic range, the expander increases dynamic contrast. When a signal reaches the expansion threshold, the output is increased by an amount that is determined by the expansion ratio. For example, an expansion ratio of 1 to 4 means that, for each 1 dB of input, the output expands by 4 dB. An expander is sometimes used after a compressor in the signal chain to make the music sound more vibrant. It is interesting to compare the sound of the next excerpt with and without expansion.

EXAMPLE 10.6

Noise Gate

A noise gate is another type of expander, but it is used to remove unwanted sounds. A noise gate can also be used to tighten a drum track or for special effects. The primary controls on a noise gate are the threshold and reduction knobs. The threshold control determines the level below which a signal will be reduced, whereas the reduction control determines the amount of reduction that will occur. In the next screenshot, note the inclusion of a three-stage envelope consisting of attack, hold, and release parameters. The attack and release parameters function in a similar way to their counterparts in a compressor or limiter. The hold parameter determines the amount of time the gate will be held open after a signal falls below the threshold.

EXAMPLE 10.7

In the next sample, a gate is used to remove extraneous noise from a guitar amp. Gates can also be used to tighten the sound of a percussion instrument by limiting the sustain of high-level transients or instrument ringing between attacks.[3]

Look Ahead

Some dynamic processing plug-ins provide the capability to analyze prerecorded information prior to the point at which the signal actually enters the processor. The benefit of engaging look-ahead is that the processor can respond in a more musical way by anticipating events. The disadvantage is that the process introduces latency—the difference in time between when a signal arrives at an input and when it reaches the output stage. Latency is not a problem during the mixing process, but look-ahead processors should be disabled when recording because any signal delay will cause problems when musicians attempt to play to a click track or overdub a part.

Equalization

An equalizer is a circuit that "allows the frequency-selective manipulation of a signal's amplitude."[4] EQ is utilized for a number of reasons, including to change the timbre of an instrument so it will better fit within a mix or to fix a problem such as sibilance or low-frequency rumble. EQ might be used to make an entire mix sound brighter or darker (similar to the way treble or bass controls are used in a home stereo). Yet another application is to tone down certain frequencies that leak into a microphone. For example, an engineer might attenuate higher frequencies on a kick-drum track in order to tone down bleed from the cymbals.

It is important to remember that EQ can be used to boost a range of frequencies, but it can also be used to attenuate frequencies. I mention this because, for most of us, it is natural to want to turn up a range of frequencies when, in many instances, this can result in a mix that sounds muddy or can be fatiguing to listen to. You will often have better results by toning down a range of frequencies. The next Web site example is a case in point. The first excerpt demonstrates a muddy and fatiguing mix that resulted from the overapplication of equalization. In the second example, more clarity and warmth is achieved by attenuating frequencies.

EXAMPLE 10.8

Shelving EQ

There are a number of types of filters that allow an engineer to adjust the level of specific frequencies. You are already familiar with a shelving EQ, since this is the filtering method used in the treble and bass controls on most home sound systems. The curve of a shelving EQ slopes to (or away) from a preset value and then flattens out to form a shelf. Notice in figure 10.3 how a shelving filter is used to attenuate low frequencies.

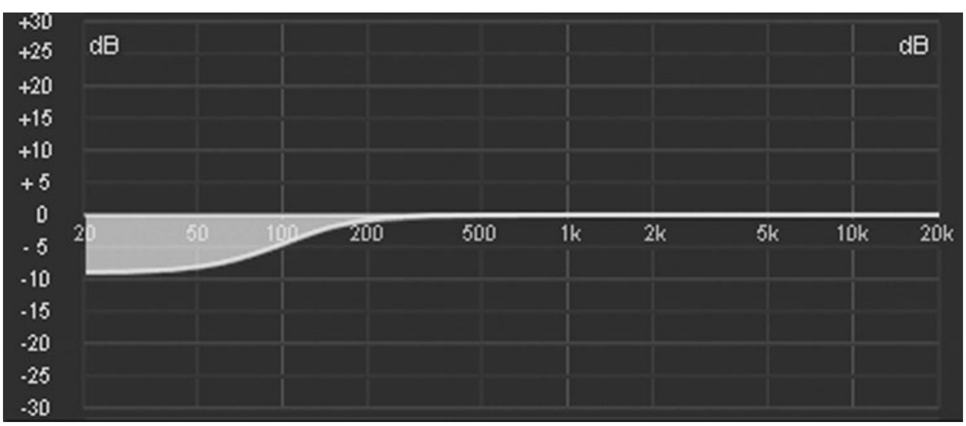


FIGURE 10.3
Shelving EQ

High- and Low-Pass Filter

High-pass and low-pass filters function in a similar way to a shelving EQ, with the exception that the filter curve does not flatten out. In this instance, filtering continues at a rate that is determined by the filter slope. As the names imply, a high-pass filter lets high frequencies pass and a cutoff is established to filter low frequencies. Similarly, a low-pass filter lets low frequencies pass and a cutoff is set for the removal of higher frequencies. Compare the curve of a high-pass filter in figure 10.4 with the low-shelving EQ from the previous figure.

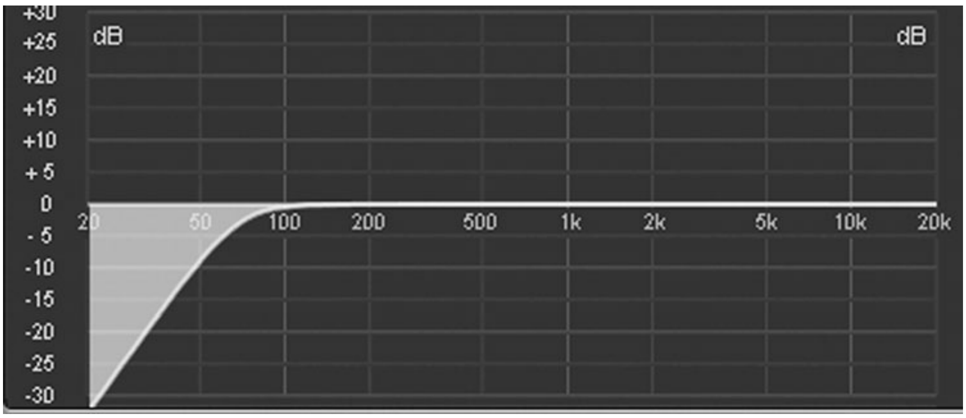


FIGURE 10.4
High-pass filter

Peak Filter

A peak filter is used to amplify or attenuate a range of frequencies around a particular center frequency. The term Q, sometimes referred to as *bandwidth*, denotes the range of frequencies that are affected around the center frequency. A high Q value means that a narrow band of frequencies will be affected while a low Q value will affect a broad range of frequencies (see figure 10.5).

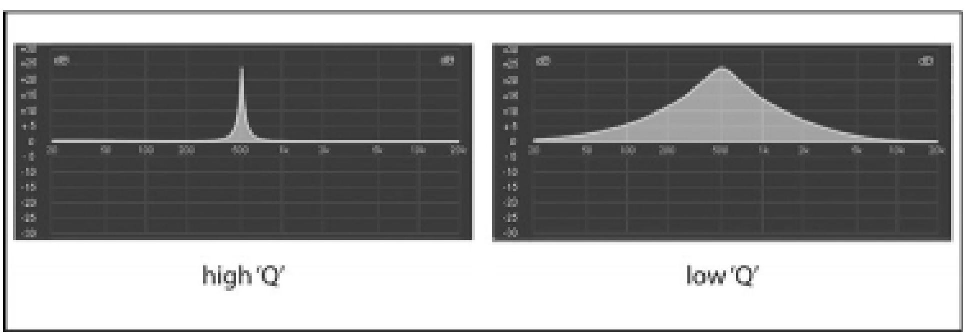


FIGURE 10.5
Peak (bell) filter with high and low Q values

Multiband EQ

Multiband EQs are found in most DAW programs and will typically consist of a shelving EQ and/or high- and low-pass filters to control the high and low bands of the frequency spectrum. Several parametric EQs are used to control the middle bands of frequencies. Notice how the bands of a multiband EQ work together to alter a broad range of frequencies, as shown in figure 10.6.

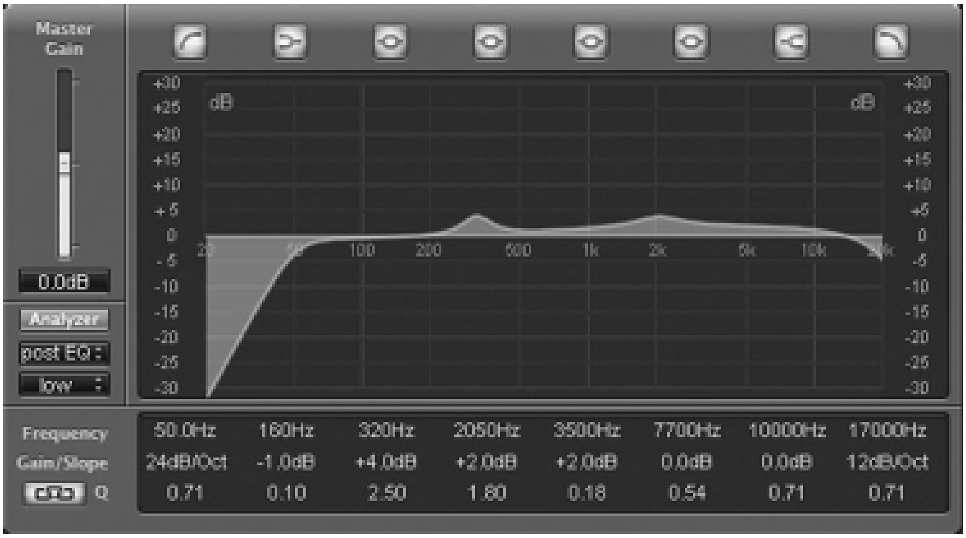


FIGURE 10.6
Multiband EQ

Effects

There are a wide variety of effects available in modern audio workstations. Effects like reverb and delay can help to give a recording a sense of ambience or spaciousness while processors such as a phase shifter or chorus are typically used for special effects.

Reverb

As you learned in the chapter on acoustics, reverberation results from multiple reflections of a sound source as sound waves interact with the surfaces of walls and objects in a room. While natural ambience or reverberation is part of most classical recordings, popular recordings typically utilize close microphone placement to minimize the natural acoustics of a room and artificial reverb is added during the mixing and mastering process. As detailed below, digital reverb plug-ins provide a number of parameters that can help you to tailor reverberation for any application.

Room Size/Reverb Time

Early reflections are the primary aural cue that help us determine the dimensions of a room.[5] One of the reasons we can aurally differentiate between a small room and an auditorium is the delay in time between the start of a sound (e.g., clapping) and the early reflections. As such, room size is a primary consideration in establishing the dimensionality of a recording. Some reverb plug-ins provide the ability to adjust the dimensions, reflectivity, and even the shape of a virtual room.

Density/Diffusion

Density refers to the number of reverberations. A diffuse sound is created by multiple reflections while echo effects can be created when individual reflections are discernable.

Pre-delay

The term *pre-delay* is used to define the amount of time between the start of a signal and the return of early reflections. Although pre-delay is conceptually similar to room size (e.g., it will take longer for early reflections to return in a large room than in a small room), the two parameters work together. One way to visualize this concept is to think of a band that is performing in a large room. If you stand in the middle of the room, early reflections will return to your ear at a rate that is determined by the dimensions of the room. Now, imagine that you are standing at the end of a long hallway that is connected to the room via a large door. There will be a delay before the reverberant sound of the band reaches the end of the hallway. This delay of reverberant sound is conceptually similar to the pre-delay setting found on most reverb units.

Mix

The mix parameter refers to the mixture of original (dry) signal and the processed (wet) signal. As with all of the other reverb parameters, finding the right balance of wet and dry signal is a highly subjective process, but it is interesting to note that recordings of popular music seem to be dryer today than in previous eras.

Convolution Reverb

Many digital audio workstations now feature what is known as a convolution reverb. In this type of plug-in, a variety of performance spaces are sampled and the acoustic properties of the room are stored in an audio file known as an *impulse response file*. A convolution reverb combines or convolutes a source signal with the impulse response file in order to apply the characteristics of the given performance space to the sound source. Convolution reverbs are typically extensible so that the user can create and import his or her own impulse response files.

Delay

Where reverberation occurs as the result of the combination of multiple reflections, delay involves fewer discrete reflections. Both reverb and delay can be used to simulate a sense of three-dimensional space, but delay can also be beneficial in establishing the position of an instrument or voice in the soundfield. Delay can also be used for doubling or echo effects.

One of the ways that a human can identify the position of the sound is by differentiating a slight delay in the arrival of the signal in one ear. You can test this by playing a mono signal through the left and right channels and adding a small amount of delay to either the left or right side. Delay on the left side will help to move the instrument to the right and vice versa. This phenomenon can be heard in example 10.9 at the companion Web site.

EXAMPLE 10.9

Modulation Effects

Modulating plug-ins such as chorus and phase shift are used to create a variety of undulating effects. These effects are typically used on guitar or keyboard tracks but can also be heard on some processed vocal tracks.

Chorus

The chorus effect is useful for special effects or to thicken a track in order to make it sound as if the track consists of multiple instruments or voices. The chorus effect utilizes multiple delays—the duration of which are modulated by a low-frequency oscillator. A rate control is used to adjust the speed of the chorusing effect. Examples of chorus and other modulation effects are available at the companion Web site.

EXAMPLE 10.10

Phase Shifter

A phase shifter is another type of modulation processor that is used for special effects. As the name implies, a phase shifter works by continuously altering the phase between a primary signal and a processed version of the signal. As the phase relationships change, certain frequencies are cancelled or reinforced to create the phasing effect.

EXAMPLE 10.11

Flanger

A flanger is similar to the chorus effect but shorter delay times are used. Flangers are sometimes used to create interesting vocal doubling effects or as a special effect on an electric guitar or other sound source.

EXAMPLE 10.12

Specialty Tools

Two other types of plug-ins deserve mention because they are commonly used in music production: pitch correction and amplifier simulation.

Pitch Correction

Pitch correction plug-ins work by adjusting the playback speed of a track so that pitches are quantized or "fixed" to a grid of notes. A pitch correction plug-in may provide the ability to specify a palette of "correct" notes in the form of a scale, key, or other set of notes. Some pitch correction plug-ins provide the ability to ignore certain notes which can be useful if the singer uses a scoop or other intentional nuance. Listen to example 10.13 at the companion Web site to hear the application of pitch correction to a phrase that is badly out of tune.

EXAMPLE 10.13

One of the things to keep in mind when utilizing pitch correction, or any other form of processing or editing, for that matter, is whether the quest to make something perfect will mar a track that sounds organic or natural. There is no question that pitch correction can be useful, and even necessary in many situations, but as with compression, you run the risk of sucking the life out of a track.

Amplifier Simulation

Bass and guitar amplifier simulators are found in many modern DAWs. These plug-ins offer an engineer the option of significantly altering the color of a bass or guitar track. Amplifier simulators model the sonic characteristics of a variety of amplifier and speaker combinations and may even provide the option of specifying the type and position of one or more virtual microphones. Web site example 10.14 demonstrates the application of a guitar simulator to a guitar that was recorded directly to disk.

EXAMPLE 10.14

Practical Application: Using Multiple Signal Processors

In the previous sections, we explored a number of tools such as dynamic processors, equalizers, and effects that are an important part of the mixing engineer's toolkit. Having a conceptual understanding of these tools is a good first step, but learning why a tool should be used and how to incorporate multiple plug-ins takes some practice. In the next section, I present a number of audio scenarios. My intent is not to demonstrate the correct use of these tools but, rather, to provide a perspective on how these plug-ins might be utilized in different contexts. Before we begin, it will be helpful to talk about the selection and use of these tools in your workstation software.

FIGURE 10.7
Adding a track insert

Using Processors in a DAW

In chapter 6 you learned about concepts such as a channel strips, auxiliary sends and returns, and the main bus. This is a good time to review those concepts as they relate to signal processing. An effect such as reverb can be added to your mix in several ways: one technique is to insert the plug-in directly into a track. In most cases your software will provide a space for setting up a track insert near the top of the track (see figure 10.7).

Reverb could also be added to the entire mix as an insert in the main bus. I have found that this technique works well for classical and jazz recordings when I want to add some natural ambiance to the overall mix.

Another way to add reverb is to use an auxiliary send and return. In this instance, reverb is inserted into an auxiliary bus and the auxiliary send is used to tap an appropriate amount of signal from each track. The processed signal is added back into the mix via the auxiliary return. A benefit of this method is that it is more efficient to process multiple tracks with a single effect than to add the same effect to individual tracks. Another advantage is that reverberation may sound more natural since the same reverb is used to process multiple tracks. With that said, there are times where it makes sense to insert reverb into one or more individual channels.

Unlike effects such as reverb, delay, or chorus, dynamic processors are always inserted into a channel or the main bus. This makes sense when you consider how these processors are used. Since a compressor is used to compress dynamics, it would make little sense to use an auxiliary send and return for the purpose. Similarly, EQ is used to fix a frequency imbalance of a track or the entire mix so makes sense that EQ be inserted on a specific track or the main bus. However, EQ is sometimes applied to a reverb return in order to warm, brighten, or otherwise change the timbre of the processed signal.

Electric Bass: Compression and EQ

Acoustic, electric, or synthetic bass provides a foundation for most popular music. These instruments are very dynamic so it can be difficult to dial in just the right amount of bass in a mix—the bass may be too soft in one passage and overpower in another, or certain notes may speak more than others. For these reasons, it is not uncommon to use compression on a bass track. In the example on the companion Web site, I also used EQ so that the bass would better fit the mix. You can listen to the processed and unprocessed examples at the Web site.

EXAMPLE 10.15

Guitar: Amplifier Simulator

EXAMPLE 10.16

The next example on the Web site demonstrates the use of a guitar simulator to create a robust distorted sound. It is interesting to note how differently the processed track sounds when compared to the unprocessed sound of the original recording.

Guitar: Compression, Chorus, Delay

EXAMPLE 10.17

In the next Web example, a combination of compression, chorus, and delay is used to create a sound that might be useful in the context of a jazz fusion recording.

Saxophone: Reverb and Compression

EXAMPLE 10.18

The performer of the next Web sample is a very expressive player and always plays with a great deal of dynamic nuance. In this instance it was necessary to compress his performance to an extent so that it would stay in the foreground of a rather thick backing track. I also added reverb (as a channel insert) on the track so that the saxophone would have a more distinctive character and dimension in relation to the backing tracks.

Piano: EQ, Delay and Reverb

EXAMPLE 10.19

Piano can be a challenging instrument to record, given its large dynamic and frequency range. The next example demonstrates how a subtle use of EQ can be used to advantage in two different scenarios. In the first Web example, I used a multiband equalizer to attenuate a range of frequencies in order to create a warm piano tone. In the second example, multiband EQ was used to brighten the sound of the instrument.

Voice: EQ and Compression

EXAMPLE 10.20

As with bass, it is often necessary to use some compression on a vocal track. This is particularly true for popular music. In the next example, a touch of EQ was also helpful to tone down sibilance.

Drums

The following audio excerpts represent a variety of signal-processing scenarios commonly used when recording drums.

Kick

EXAMPLE 10.21

In the first Web site example, a compressor is used to tighten up the bass drum and EQ was used to provide more body. You will notice a fair amount of bleed from the cymbals so a low-pass filter was used to tone down some of the higher frequencies.

Snare

EXAMPLE 10.22

There are innumerable ways to approach a snare track. Bear in mind that the best solution is often to leave the track alone if it sounds good and fits with the mix. In the next example, a compressor with a slow attack setting and EQ was used to create a fairly robust sound.

Composite

EXAMPLE 10.23

In the next example, multiple drum tracks (overheads, snare, kick, and toms) are sent to a submix and compression and reverb is added to the mix of drums.

Global Effects

EXAMPLE 10.24

Up to this point, we have focused on processing individual tracks with one or more effects. This would be a good time to demonstrate the application of global processing to an entire mix. In the first example on the Web site, reverb is added to the main mix of a jazz recording and EQ was used to tone down midrange frequencies that were overemphasized in the recording.

EXAMPLE 10.25

In the second example, reverb was added to the main bus of a recording of a string duo in order to mimic the sound of a performance hall.

Don't Fix It in the Mix

As I mentioned in the last chapter, some problems such as poor microphone placement, distortion, excessive bleed, and various types of noise may present insurmountable problems. It will be relatively easy to incorporate a track in a mix if the track sounds great as it is recorded to disk. Unfortunately, you will spend an inordinate amount of time trying to use signal processors to fix a poorly recorded track. For this reason, the extra upfront effort involved in properly placing microphones will pay big dividends in terms of your time and the quality of your final product.

Budget Signal Processing

Most digital audio workstations provide a reasonable selection of processing tools so you will likely not need to purchase third-party plug-ins unless you require special tools for mastering or a particular type of processor such as an amplifier simulator or convolution reverb. In many cases, it is possible to create new effects through the creative application of tools that you already own. For example, you can create a wide variety of effects by recording a vocal track in a bathroom, barn, or garage.

Similarly, the concept of *reamping* provides a wealth of opportunities for exploration. In this scenario, a prerecorded guitar or other signal is routed back to an amplifier or speaker system and the output is rerecorded with one or more microphones. This concept could also be used to create a custom echo chamber in a bathroom or other reverberant room.

It is also interesting to consider routing the output of an audio interface to an old tape deck or analog mixer in order to add color or warmth through a process called *summing*. Dedicated summing boxes can be expensive, but there is obviously no cost in experimenting with equipment you already own—you might be surprised with the results.

11

Mixing and Mastering

This chapter brings a number of concepts, both artistic and technical, to bear on the topics of mixing and mastering. In chapter 8, "Active Listening," I presented a number of concepts that will help you pick apart a mix. In chapter 10 you learned about the many tools that are available to process digital audio. This chapter will focus on the use of these tools to mix and master music. We will consider the operation of each tool, as well as strategies that can help you to make good artistic decisions.

Mixing

The mixing process involves many activities. An engineer will need to consider the relative level of each track, as well as the placement of instruments in the soundfield. It may be necessary to apply EQ or compression to a track or to the entire mix, add artificial reverb, or digitally edit a performance. Given the many hats that an engineer must wear, it is easy lose sight of the bigger picture when focusing on the details of a mix.

What Constitutes a Good Mix?

In a sense, this is a rhetorical question as there are so many variables and stylistic considerations involved that the question becomes moot. On the other hand, most of us recognize a good mix when we hear it, and I would offer that good mixes tend to share the following characteristics:

- Clarity: Tracks are well recorded and the listener need not work to pick out individual parts. The recording is not marred by extraneous noise, distortion, audible edits, or other problems.
- Balaced levels: The delineation between foreground and background elements is clear. Foreground elements don't overpower and background elements don't disappear.
- Balanced frequency spectrum: No part of the frequency spectrum is unduly loud or soft.
- Balanced image: Elements are placed in the soundfield such that each element has its own place in the mix.
- Dynamic contrast: This is somewhat dependent on genre, but mixes that exhibit some dynamic contrast tend to be more interesting to listen to.
- Interest: Mixes of popular music tend to include some elements that are attention grabbing. This might be in the form of active panning, a striking signal processing effect, the use of distortion, or even an abrupt silence.

Equipment

Mixing can be accomplished solely "in the box" with a computer, audio interface, reference speakers, and headphones. This approach is economical and can yield excellent results. Alternatively, you may wish to use a mixing console (either analog or digital). For example, the PreSonus StudioLive is a digital mixing and recording console with FireWire connectivity that provides the benefits of a digital audio interface with the tactile feel of an analog console. Mackie is another company that offers a range of popular mixers with FireWire or USB connectivity.

Another alternative is to use a mixing control surface such as the Euphonix MC Control or Mackie Control Universal. While these products do not provide audio input or output, they offer a hands-on approach to the control of an audio workstation.

The Listening Environment

In an ideal world, your mixing room will have perfect acoustics, extremely low noise floor, and a set of high-end reference monitors. For the rest of us, it is necessary to make the best of a modest budget. A good start is to consider investing in a pair of active near-field monitors. As I mentioned in chapter 1, quality reference monitors can be expensive, but some of the budget offerings are quite good and will be much more useful than home stereo speakers.

Consider the placement of your monitors: Your user manual will provide instructions for the configuration and calibration of your speakers, but you should also consider potential acoustical problems. It is usually best to place the monitors so that they are the same distance from the side walls, so that early reflections reach your ears at the same time. Also, be aware that tall equipment racks, computer monitor, and other acoustic obstructions can create problems in your listening space. As detailed in chapter 3, acoustical treatments such as bass traps, absorbers, and diffusers may also be necessary to fix problems in your mixing room. In some cases, alternate placement of bookshelves and other furniture can have a positive effect on acoustics.

Although I would not understate the importance of acoustics in your mixing environment, this book is about being practical in your approach to recording music. One economical approach is to purchase a pair of headphones. Quality headphones are a relatively inexpensive addition to a home studio, and while I don't suggest relying solely on headphones for mixing, a set of "cans" such as the Audio-Technica ATH-M50 or Sony MDR-7506 *can* be used for mixing if your budget doesn't allow for near-field monitors.

Developing Mixing Skills

I always tell my composition and theory students to think in terms of letting their intuition guide their intellect, and I think that advice is appropriate here as well. It's always a good thing to consider the technical aspect of audio recording, but technique should serve your artistic goals and not the other way around. As with all musical endeavors, your mixing skills will improve with practice and experimentation. In addition to practice, make a habit of getting feedback from other musicians and engineers—their perspective will provide many insights that will help you improve. I would also encourage you to listen critically to a wide variety of music. The suggestions made in chapter 8, "Active Listening," will help you to pick a mix apart and glean insights into the mixing process.

Make the Best of What You Have

In this age of reasonably priced digital hardware it is easy to think in terms of purchasing new equipment to improve our work in the realm of the recording arts. Sometimes a new microphone or preamp is needed, but "gear lust" can be counterproductive when it comes to developing skill in the recording arts. In art, limitations can be an advantage. Consider how a mature improviser might develop a solo using just a note or two for an entire chorus of the blues, or the way a graphic artist might use subtle shading when drawing with pencil and paper. The same could be said of the recording engineer who knows how to get a variety of colors from a single microphone. I mention this because audio trade magazines can be discouraging when you read something along the lines of "I always use a matched pair of Cost Too Much microphones going through Your Kids College Fund preamps, and I mix with More Than Your Mortgage reference monitors." While I am all for quality and attention to detail, I would also suggest that you can learn a great deal by working with modest tools and learning to use them to the best advantage.

Getting Started

I have been blessed to work with many excellent engineers over the years, and one of the things I have noticed is that there is no one-size-fits-all approach to mixing. Some engineers bring the faders down and start working on a single track at a time while other engineers might start leveling all of the tracks right away. One of my favorite experiences came from a trio session at a small studio in the 1990s. The engineer (I regret that I don't remember her name) took the old-school approach of mixing in real time and recording everything straight to two-track tape. That recording was so warm and natural, and it has always stuck in my mind what a wonderful sound she got out of her rather primitive equipment. Ultimately, our goal is to create great-sounding music so don't be afraid to explore and experiment. It's also helpful to remember that making music is *fun,* so don't put too much pressure on yourself by thinking that your first mix must be perfect.

Getting Organized

Sub Bus

A sub bus (sometimes called a sub mix) can be very useful in organizing and controlling multiple tracks during mixdown. It is not uncommon to use four or more microphones when recording a drum set, but individual tracks can be unwieldy to control. Instead of routing each track to the main mix, consider routing the output to a stereo sub bus. You will still be able to control the level and pan of each track and apply signal processing, but the sub bus will provide a single point of control should you want to mute all of the drums or adjust the overall level of the kit.

Mono Switch

It is useful to periodically check a mix in mono in order to listen for phase problems or level imbalances. Your mixer may provide a mono button, but consider setting up this functionality in your DAW if a mono switch is not available. Your software should provide a switch (often as part of a gain insert or directly on the channel strip) to toggle between stereo and mono.

Channel Recall

Most DAWs provide the ability to save and import the settings for a channel strip or even an entire mix. Consider focusing your efforts on a single song because you will likely be able to reuse at least some of the settings for multiple songs in a project.

Primary Mixing Domains

One definition for a domain is "A sphere of activity, concern, or function";[1] and that is a useful way to organize a thought process for mixing. The process of mixing tracks is, in many respects, akin to the process of music orchestration, and this analogy will be helpful to those readers who approach audio recording from the perspective of a classically trained musician. An orchestrator will consider any number of ways (and often a combination of techniques) to bring an element to the foreground.

Dynamics

Louder instruments will tend to move to the foreground. Note that another way to phrase this is that softer instruments will tend to move to the background. I mention this because, for most of us, our tendency is to raise the level of a track, but it is sometimes more effective to lower the level of backing tracks.

Register

Instruments in higher registers will tend to be heard in the foreground. By way of example, consider the rather homogeneous sound of a string quartet. The top voice (usually first violin) will tend to be in the foreground unless another part is differentiated by a contrast in dynamics, rhythm, or timbre (e.g., with the application of a mute).

Rhythm

Contrasting rhythmic elements tend to move to the foreground. Consider the string quartet from the last example. The viola (usually the third part from the top) will not ordinarily sound in the foreground unless the other instruments are softer. However, it is easy to see how the viola can come to the foreground if the composer writes moving eighth-notes for the viola set against static whole notes in the other parts.

Timbre

Timbre refers to the color or tone quality of a sound. Building on the last example, a viola can come to the foreground if its timbre contrasts with the surrounding texture. Thus, an orchestral composer might double the viola with an English horn or clarinet. The distinctive change in timbre will easily bring the line to the foreground in the context of an orchestra. For a recording engineer, the application of EQ is used in a similar fashion. Certain frequencies might be emphasized to affect a timbral variation that will bring an element to the front of the mix. Similarly, certain frequencies might be attenuated in a background element so that the part stays in the background.

Starting a Mix: Balance and Dynamics

A good starting point is to focus your attention on the relative level of each track. If you are mixing a pop or rock project, consider soloing the bass and drums and adjust the level of each track so that they blend in a pleasing way. A touch of compression may be helpful if certain bass notes sound too soft or loud. As you mix, be aware of the concept of foreground and background. The rhythm section will typically be in the background, although you might want to emphasize certain elements within the background texture. For example, the snare is typically prominent in most pop recordings. Keep in mind that pan and EQ can have an impact on your perception of loudness for a given track. For this reason, I often alternate between adjusting faders and pan at this stage in the process.

As the rhythm section starts to take shape, bring in other elements such as keyboard, guitar, and vocals. Vocals are usually the primary foreground element so it is often necessary to utilize some compression to keep the voice in the foreground. One strategy is to adjust faders so that the balance between background elements and lead vocal is generally good. You can then add an appropriate amount of compression so that the vocal track consistently stays in the foreground.

The mixing strategy presented thus far is an additive process: tracks are added to the mix and the engineer adjusts balance, pan, and dynamics as needed. Another approach is to consider a subtractive process. In this scenario, adjust the level of each track so that their signals are strong but not distorted. At this point, the mix will likely sound bad because the tracks are competing with one another. Next, pull down faders as necessary to reveal the primary foreground elements. Although this approach may seem less intuitive than the additive process, the technique can provide a unique perspective on a mix.

Spatial Positioning

Pan controls are one of the tools used to position an instrument or voice in the stereo soundfield, and as I mentioned in the last section, it is often helpful to make pan adjustments as you balance the level of tracks in the mix. One way to visualize the relationship of instruments in the soundfield is to consider a three-dimensional space. In figure 11.1, the X-axis represents the position of instruments from left to right across the soundfield. The Y-axis represents the prominent frequency range of the instrument or voice—for example, piccolo near the top and double bass near the bottom. The Z-axis represents the position of the instrument in terms of foreground or background. This is largely determined by the perceived loudness of the instrument, but effects such as reverb can also have an impact on the position of an instrument on the Z-plane.

Although it can be tempting to pan elements such as a stereo synthesizer pad to the extreme left and right pan positions, be aware that your mix can suffer from what some engineers call "big mono."[2] Big mono results from panning many stereo elements hard left and right, resulting in little differentiation between the channels. In some instances, it may be better to use a mono version of the track so that the element will retain its clarity when mixed with other instruments.

As I mentioned in chapter 10, delay can be a useful in localizing an element in the soundfield. You could, for example, process a monaural signal with a stereo delay and vary the amount of delay in order to move the instrument left or right (e.g., a delay on the left will tend to move the element to the right).

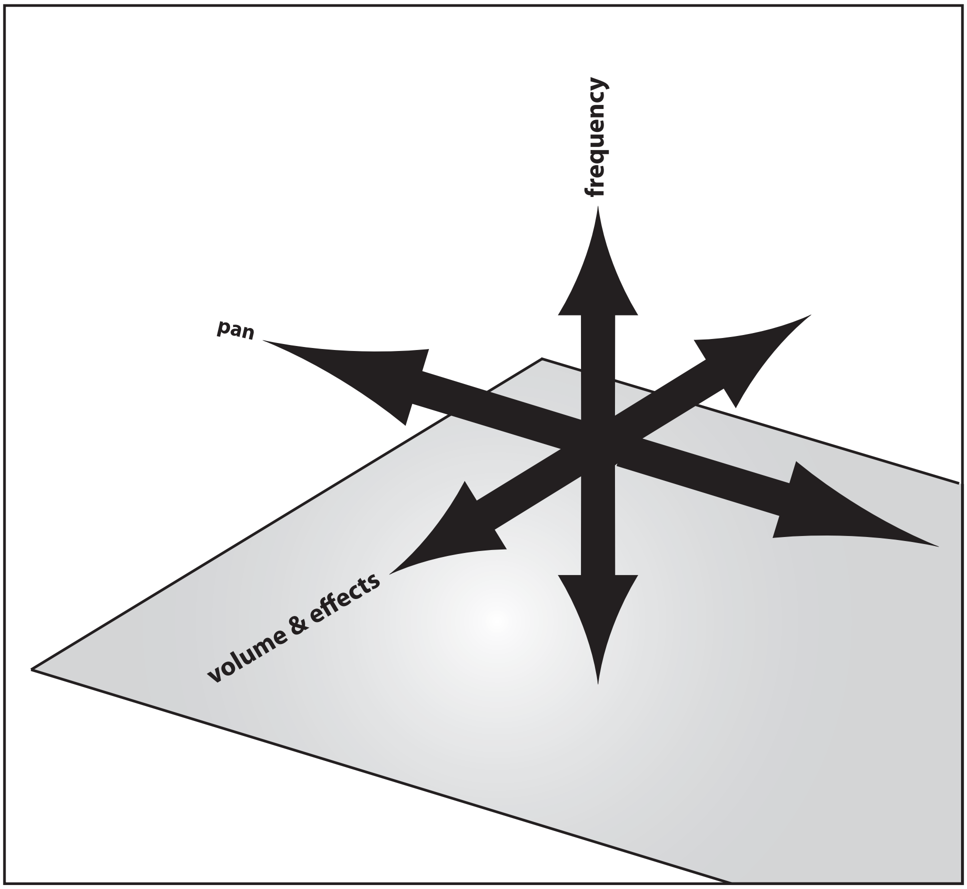


FIGURE 11.1
Three-dimensional representation of a mix

Frequency Spectrum

In an earlier section, I described how a change of timbre can help to bring an instrument to the foreground. Equalization is one of the tools that engineers utilize for this purpose. EQ can also be useful in providing clarity to a mix by selectively filtering or emphasizing certain frequencies.

When working in the domain of frequency spectrum, it is helpful to solo a track in order to focus on the sounds emanating from an instrument or voice. One of the things to keep in mind, though, is that a track that sounds full and well-balanced in terms of the frequency spectrum will not necessarily fit within the bigger picture of a mix. A food analogy will be helpful to illustrate this point. Consider a perfectly crafted pizza—the spices in the sauce mix with the flavors of the dough, cheese, and other toppings to create a well-balanced mix. Now consider that artificial smoke flavor has been added to the sauce. Although this flavor could be the basis for an effective barbeque, it would likely overpower the mix of flavors in a pizza. The mixing process can be similar in that a subtle use of individual elements can contribute to a successful mix.

The frequency spectrum is generally divided into four primary regions: bass, low-mids, high-mids, and high. The emphasis or attenuation of frequencies in a given band can have a striking (not always positive) impact on a mix. Keep in mind that your perception of the effect of boosting or reducing frequencies in the bands listed below will likely differ since these descriptions are subjective. However, in researching numerous books on the subject, I noticed that similar adjectives were used and this coincides with my anecdotal experience. I would encourage you to make your own observations on the effect of equalization in the following range of frequencies. This is easily accomplished by setting up a single-band parametric equalizer in your DAW. Use the equalizer to boost and reduce frequencies in each of the following ranges and make notes about your subjective reactions.

Bass (40 Hz to 250 Hz)

The fundamental tones of a four-string bass fall within this register, and engineers often describe this frequency band as providing "punch" and "power." An overemphasis on this frequency band can make a mix boomy and an under emphasis can cause a mix to sound tiny or weak.

Low-Mids (250 Hz to 2000 Hz)

The low-mid frequency band includes the frequencies from middle C on the piano to about three octaves above middle C. Hence, the fundamental tones of many instruments fall within this range. Keep in mind, though, that the fundamental pitches of many instruments such as guitar, alto and tenor saxophone, trombone, and French horn straddle the bass and low-mid range. Emphasis of this band may create a nasal or hornlike sound and can be fatiguing to listen to, but a subtle emphasis between 1 kHz and 3.5 kHz can help to bring a vocal track forward in the mix. Attenuation may create a muted, indistinct, or muddy sound.

High-Mids (2000 Hz to 4000 Hz)

Emphasizing frequencies in this band can provide a sense of clarity and brightness, but may also make a track too strident. Overemphasis of the frequencies creates a small sound similar to a telephone while attenuation can make a recording sound somewhat muted.

High (4000 Hz to 20,000 Hz)

Harmonic information is found in the highest band of frequencies, and a boost of frequencies above 7 kHz can create a crisp or airy sound but may also contribute to vocal sibilance. High frequencies may be divided into two sub categories:[3]

Presence (4000 Hz to 6000 Hz)

A boost of this band can add sizzle to cymbals and provide a better sense of definition to an instrument or voice, but overemphasis may become piercing or twangy. Attenuation may create a muffled or distant sound.

Brilliance (6000 Hz to 20,000 Hz)

Extended high frequencies can provide a sense of detail or clarity and contribute to a sound that might be described as brilliant. Attenuation of this band can mellow or smooth sound but may also make a mix sound dull or lifeless.

Figure 11.2 provides a comparison of frequency ranges and pitch for many common instruments and vocal designations.

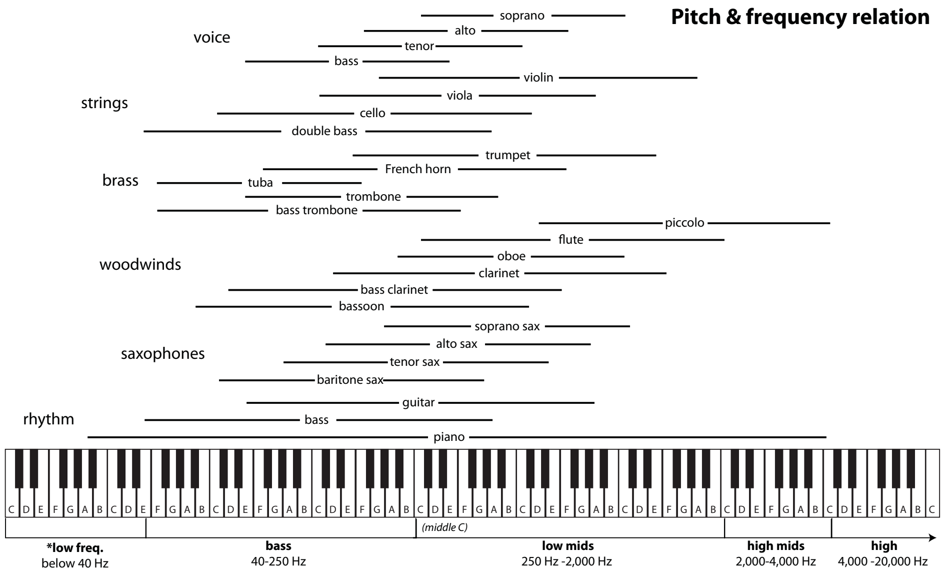


FIGURE 11.2

Pitch and frequency relation

Applying Equalization

One of the biggest challenges in applying equalization is that an equalization change on one track may impact other tracks or the mix as a whole. It's very easy to add a subtle boost of high frequencies to enhance the sound of cymbals and a similar boost in a vocal track that results in an overemphasis of these frequencies. For this reason, it is important to frequently check the way a track fits in the mix as a whole as you apply equalization to individual tracks. Keep in mind that it often works better to attenuate frequencies on other tracks than to boost a band of frequencies on the track in question.

In most cases, the entire frequency spectrum should be represented in a mix without any bands being unduly emphasized or reduced. This may mean that you will need to reduce a range of frequencies in one track after boosting the same range in another track. For example, to better emphasize a vocal track you might subtly boost frequencies in the range of 1.5 kHz to 3 kHz. In order to keep a pleasing or "equalized" balance of frequencies, it may be necessary to attenuate backing tracks in the same frequency range. Although the lead vocal may sound somewhat strident and the backing tracks slightly muted when heard individually, the combination of elements can work together to provide a sound that is well balanced.

When applying equalization, I sometimes find it helpful to use a parametric equalizer with a narrow Q in order to focus my ears on the band of frequencies in question. Adjust the center frequency until you find the range of frequencies that is causing a problem such as vocal sibilance or a boomy bass.

Table 11.1 was provided by PreSonus, a manufacturer of professional audio tools. The table provides a useful breakdown of the effect of boosting or reducing

TABLE 11.1 EQ parameters

Instrument	What to cut	Why to cut	What to boost	Why to boost
Human Voice	7 kHz	Sibilance	8 kHz	Big sound
	2 kHz	Shrill	3 kHz and above	Clarity
	1 kHz	Nasal	200–400 Hz	Body
	80 Hz and below	Popping P's		
Piano	1–2 kHz	Tinny	5 kHz	More presence
	300 Hz	Boomy	100 Hz	Bottom end
Electric Guitar	1–2 kHz	Shrill	3 kHz	Clarity
	80 Hz and below	Muddy	125 Hz	Bottom end
Acoustic Guitar	2–3 kHz	Tinny	5 kHz and above	Sparkle
	200 Hz	Boomy	125 Hz	Full
Electric Bass	1 kHz	Thin	600 Hz	Growl
	125 Hz	Boomy	80 Hz and below	Bottom end
String Bass	600 Hz	Hollow	2–5 kHz	Sharp attack
	200 Hz	Boomy	125 Hz and below	Bottom end
Snare Drum	1 kHz	Annoying	2 kHz	Crisp
			150–200 Hz	Full
			80 Hz	Deep
Kick Drum	400 Hz	Muddy	2-5 kHz	Sharp attack
	80 Hz and below	Boomy	60-125 Hz	Bottom end
Toms	300 Hz	Boomy	2–5 kHz	Sharp attack
			80–200 Hz	Bottom end
Cymbals	1 kHz	Annoying	7–8 kHz	Sizzle
			8–12 kHz	Brilliance
			15 kHz	Air
Horns	1 kHz	Honky	8–12 kHz	Big sound
	120 Hz and below	Muddy	2 kHz	Clarity
String section	3 kHz	Shrill	2 kHz	Clarity
	120 Hz and below	Muddy	400–600 Hz	Lush and full

specific frequencies of many common instruments. Figure 11.3 provides a different way of looking at the effect of emphasizing or attenuating various frequency bands.

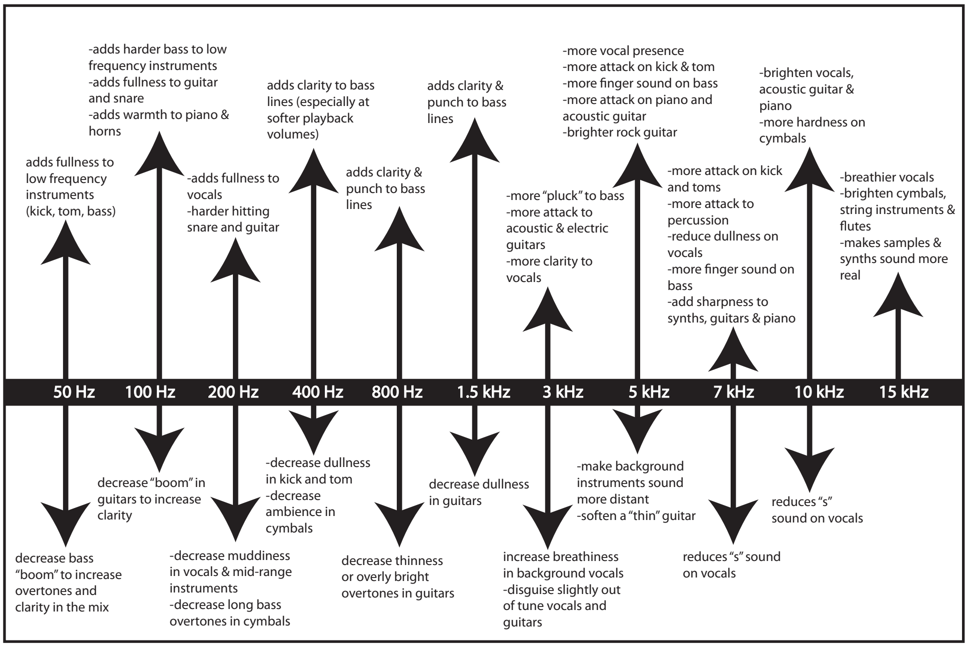


FIGURE 11.3

Applying EQ (Based on an illustration by PreSonus. Used by permission.)

Ambience

The Merriam-Webster definition of *ambience* is "a feeling or mood associated with a particular place, person, or thing." That definition is also appropriate for audio recording, as it provides an insight into why effects like reverb, delay, or chorus might be added to a mix—namely, that effects can provide interest and character to a recording. Effects can offer a sense of dimensionality by mimicking the characteristics of a room or performance hall. They can help place a track forward or backward on the Z-axis. For example, a wet track will tend to sound more distant than a dry track. Thus, when used with panning, effects can help position an element in a three-dimensional space.

There are a number of common approaches to utilizing reverb and delay. One approach is to mimic the sound of a natural acoustic environment. With this approach, reverb may be added to the main bus or an auxiliary send in which multiple tracks utilize the same reverb. The second approach is much more flexible since each track can contribute a varying amount of signal to the auxiliary send. Individual tracks may also be processed with reverb via a track insert, but if the goal is to create a natural sounding space, care must be taken to ensure that the reverberation of individual tracks blends to make a convincing whole.

A contrasting approach is to apply effects to individual tracks so that each element has its own unique sound and space. For example, on a recent project I applied compression and reverb to create a fairly wet saxophone track. In this instance, the backing tracks were fairly dry but the snare was heavily processed to create a big sound. Although this combination of elements would not naturally occur in a performance space, the elements fit together to create a convincing artificial space.

Keep in mind that an auxiliary return is just like a channel strip in that it can be panned, equalized, or processed in some other way. In some situations, it may be advantageous to apply EQ in order to lighten or darken the effect, or pan can be used to move the effect to one side.

In contrast to reverberation, effects such as chorus and flange are typically used to thicken a part or to create interest—for example, flange and delay might be used to process a guitar signal so that it sounds more interesting or chorus could be subtly added to a lead vocal to fatten the track. Bear in mind that, as with equalization, a "bigger than life" track might sound good on its own but will not necessarily fit within the bigger context of a mix so it's a good idea occasionally switch between individual tracks and the full mix when experimenting with effects.

Automation

As your mix takes shape, you will undoubtedly want to automate some parts of the process. For example, you might want to pull down the level of a saxophonist's microphone during a guitar solo or boost the lead vocal during the chorus of a song. In a DAW, most elements such as pan, volume, and mute can be automated so that fader moves and button presses are recorded and automatically "performed" each time the track plays back. In some cases, it will even be possible to automate plug-in parameters. Note that chapter 7 provides an overview of track automation.

Mastering

Once songs have been mixed, a mastering engineer (technically called a pre-mastering engineer)[4] may be involved in preparing tracks for duplication. The job of a mastering engineer is multifaceted: one of the most important functions is to provide an experienced set of ears to evaluate a mix and make necessary adjustments that will improve the mix in the domains of frequency and dynamics and to detect and fix noise and other anomalies. The mastering engineer works in a room that has been sonically treated for the purpose and utilizes an extremely accurate reference system. Historically, mixing engineers prepared the physical master that was used to mass-produce vinyl records—a process that required specialized knowledge and equipment. Although today's mastering engineer will prepare a physical master for digital duplication, the process still requires specialized knowledge and equipment.

When to Hire a Mastering Engineer?

Your decision to hire a mastering engineer will largely be determined by the budget for your project. An experienced mastering engineer can be an invaluable asset and will provide comfort that your work will be in the best possible shape prior to duplication. The other side of this question is one of practicality, and there are many situations for which it is impractical to hire a mastering engineer. For example, you might want to duplicate just a few hundred CDs or prepare a quality demo of your band. It would also be impractical to hire a mastering engineer when writing music for television since deadlines are typically very tight. I remember a comment from the owner of a national music service when he accepted some of my compositions for inclusion in his music library: "I don't have time to work to improve recordings for the library—the tracks need to sound great and be ready for broadcast." Given the do-it-yourself nature of this book, several pages will be devoted to the mastering process.

Mastering with iZotope

Unlike for the other chapters in this book, here I elected to focus on a set of tools from a single vendor for the screenshots. Ozone 4 is a collection of mixing and mastering tools by iZotope that are reasonably priced and offer good performance value. I would stress that specialized mastering tools are not essential, as most DAW applications come with an extensive set of plug-ins. Although the discussion will focus on iZotope, the concepts in this chapter are readily applicable to plug-ins found in all of the major DAWs, as well as from third-party vendors.

The Ozone suite provides six tools: a paragraphic equalizer, mastering reverb, loudness maximizer, multiband harmonic exciter, multiband dynamics processor, and multiband stereo imaging. The following paragraphs are devoted to a discussion of the function and use of each of these tools. If you haven't done so already, be sure to read chapter 10, "Processing Signals," since it provides a necessary foundation for many of the concepts presented in this section.

Main Screen

Figure 11.4 shows the main screen in Ozone. Click the Presets button to toggle between preset view and settings view for the currently selected tool. Tools are selected with the button to the left of each tool name and enabled by clicking on the button just to the right of the tool name. The presets are a good starting point. Try each of the mastering presets and explore the settings in each module to develop a sense of how the modules work together to provide a particular sound, such as "Midrange detail" or "Gentle Tube."

Mid-Side Processing

Some Ozone tools such as the paragraphic equalizer provide the option to utilize mid-side processing. The concept behind mid-side processing is that a mix

FIGURE 11.4

Ozone main screen

has three components: left side, right side, and middle channel. The middle, or "phantom," channel typically consists of primary elements such as bass and voice. Ozone can process a mix in stereo or processing can be independently applied to the mid or sides of the mix. Stereo, mid, and side buttons are provided for the purpose. Note that a bypass button (b) and solo button (s) are also provided (see figure 11.5).

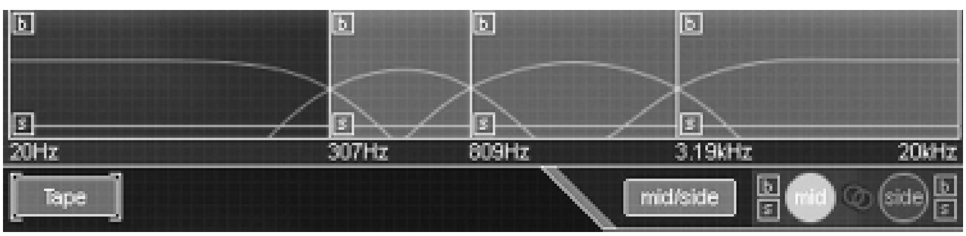


FIGURE 11.5

Selecting the mid channel in Ozone

Paragraphic Equalizer

The paragraphic equalizer provides multiple filters including bell, high-pass, low-pass, high-shelf, and low-shelf that can be applied to the frequency bands at the top of the equalizer. Click one of the green circles to adjust the parameters of a band. Drag the circular control point left or right to change the center frequency or up or down to adjust the gain parameter. The brackets indicate the filter Q or bandwidth. Bandwidth can be adjusted by clicking on a bracket and dragging left or right (see figure 11.6).

Click the Show Info button to change filter types. This action will reveal a window with which you can specify the filter type, as well as frequency, gain, and Q.

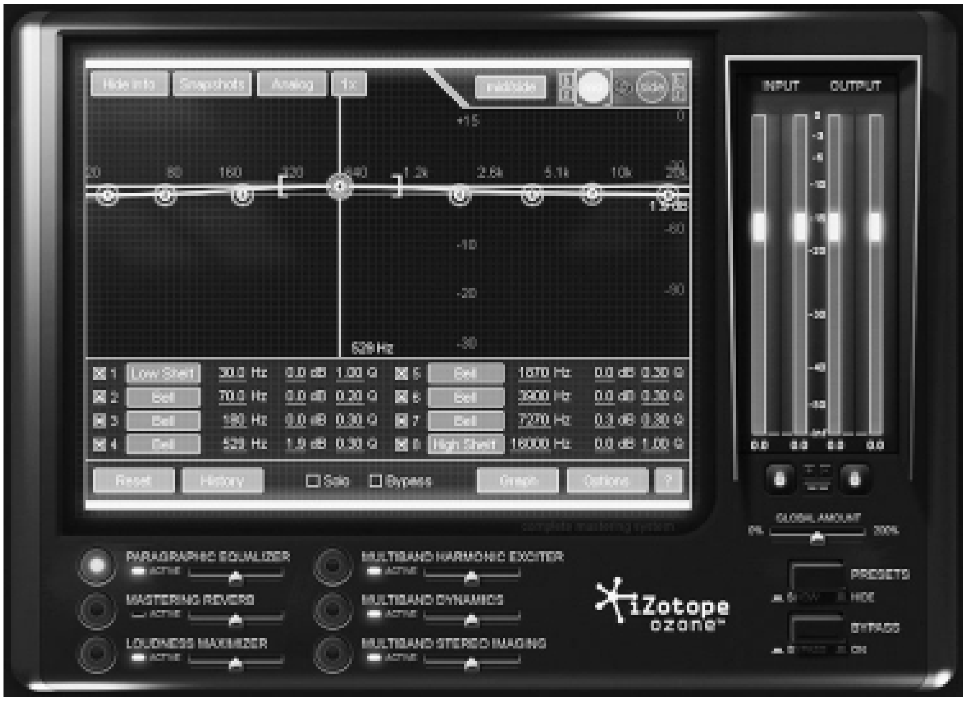


FIGURE 11.6

Adjusting EQ parameters in Ozone

Ozone provides the ability to select digital equalization or an analog mode that models an analog tube equalizer. Another interesting feature called "matching EQ" can be used to analyze the frequency spectrum of a source recording. The EQ of the source file can then be applied as a real-time plug-in.

Earlier in the chapter I mentioned a trick for zeroing in on a specific frequency by altering the bandwidth of a filter and adjusting the center frequency. Ozone provides a convenient built-in method to accomplish the same thing. Simply hold the Alternate key and click in the equalization screen to focus on the range of frequencies under the mouse cursor.

Using the Paragraphic Equalizer

The paragraphic equalizer provides all of the functionality of a multiband equalizer found in most DAWs. You could, for example, use a shelving filter to tone down the bass or a bell filter to boost or reduce mid-range frequencies. In terms of mastering, mid-side processing is very powerful since EQ and other types of processing can be applied to specific elements in the mix without affecting other elements. For example, you could boost frequencies around 3 kHz in the middle channel to bring a vocal forward in the mix and tone down those same frequencies in the side channels. Similarly, a wider sound can be achieved by adding brilliance to the side channels while leaving the middle channel unaltered.

To use mid-side processing, click the Stereo button to make the mid and side channels available. The mid frequencies are shown in orange and the sides in blue. You will want to make frequent use of the solo button as you adjust frequency parameters in order to focus your attention on either the middle or side channels.

Mastering Reverb

The mastering reverb is used to sweeten a mix by providing a subtle ambience. The controls of the mastering reverb are easy to use: the room-size parameter controls the reverb tail and the room-damping and pre-delay control are used to adjust the brightness of the room and the amount of time prior to the start of reverberation. As with other effects, sliders are provided to proportion the amount of dry and wet signal.

One of the more powerful features of the mastering reverb is the ability to apply reverb to a specific band of frequencies on the center channel and another set of frequencies on the side channel. This is accomplished through the use of the Mid and Side buttons in conjunction with the frequency range control at the top of the reverb screen. One application of mid-side reverb is to apply reverb to the side channels in order to create a greater sense of depth while leaving the center channel relatively dry. Note that the toggle switch titled Show Channel Ops in the correlation and phase meter provides buttons for mono mode, channel inversion, channel swapping, and phase rotation (see figure 11.7).

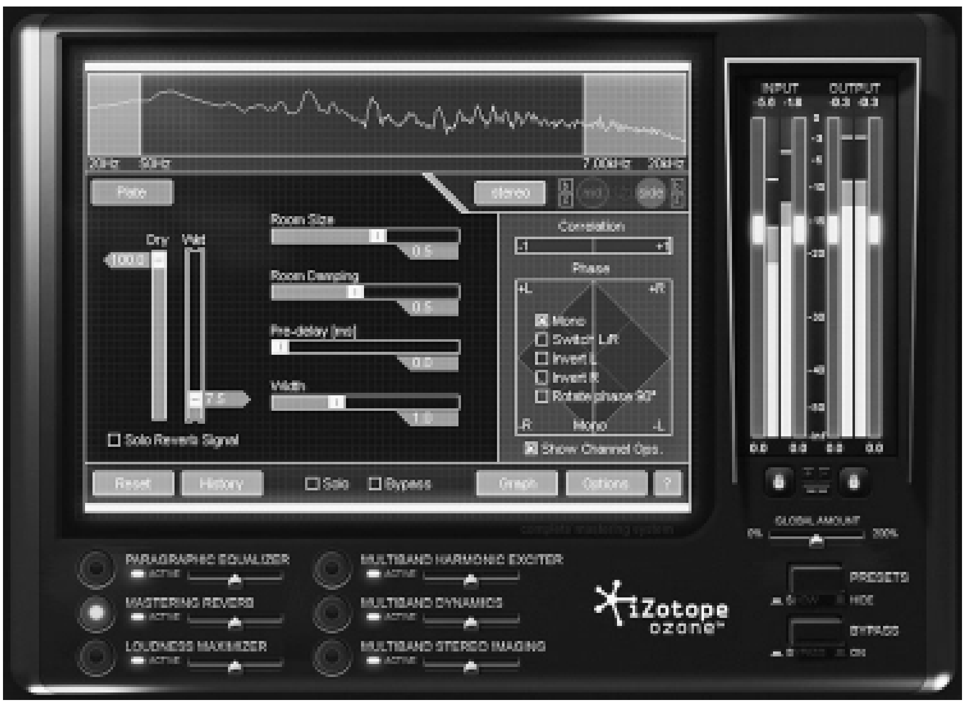


FIGURE 11.7

Using the mono switch in the Ozone reverb plug-in

Multiband Dynamics Processor

Unlike a traditional dynamic processor, the multiband processor can compress, limit, or expand specific bands of frequencies. Each band is selected by clicking one of the four bands available at the top of the screen. Band width can be altered by dragging the vertical bars to the left or right. Note that each band provides a bypass (b) and solo (s) button that will be useful in dialing in an appropriate amount of dynamic processing (see figure 11.8).

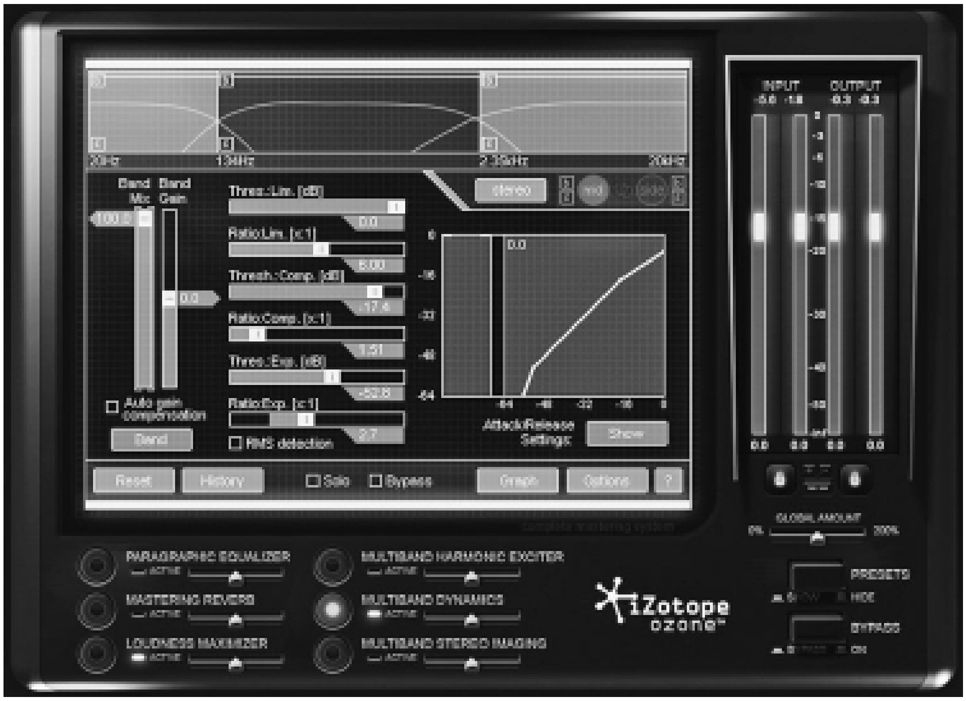


FIGURE 11.8

Multiband dynamics processor

Each band can be dynamically processed with a limiter, compressor, or expander; threshold and ratio controls are available for each of these functions. You will notice that a toggle button titled RMS Detection is provided. *RMS* stands for "root mean square" and is a formula that is used to average instantaneous voltages.[5] The RMS option is useful when trying to increase the overall loudness of a mix. In contrast, the standard peak detection is useful to even out transitory signal spikes.

As with the mastering reverb and equalizer, the multiband dynamics processor can be applied in stereo or separately to the mid and side channels. This provides an astounding level of control but also makes the processor difficult to use. I find it helpful to make frequent use of the solo button in order to focus my ears on a specific range of frequencies and then adjust the ratio and threshold sliders in order to achieve the desired amount of dynamic processing. In general, signal processing should be transparent, but a subtle use of compression or expansion can be difficult to achieve when there are so many parameters. For this reason, you may want to frequently toggle the bypass switch in order to hear the results of adjustments to specific frequency bands in the context of the mix as a whole. The bypass button can be a useful gauge to determine if the given signal processing is having a beneficial or detrimental effect in the context of the mix.

The multiband dynamics processor can be used to correct or enhance a mix in a number of ways. For example, compression could be applied to low frequencies in the center to tighten the dynamics of a bass while middle frequencies are compressed in the side channels in order to smooth out dynamics of an instrument that is panned to the side.

Multiband Harmonic Exciter

The harmonic exciter is used to add a subtle amount of harmonic distortion to a recording. Although distortion is usually something to avoid in recording (distorted guitars are an obvious exception), added harmonics can give a unique color and sparkle to your mix. Ozone provides several models, including *tube*, *tape*, *warmth*, and *retro*. Harmonic processing can be applied to individual bands of frequencies, as can be seen in figure 11.9.

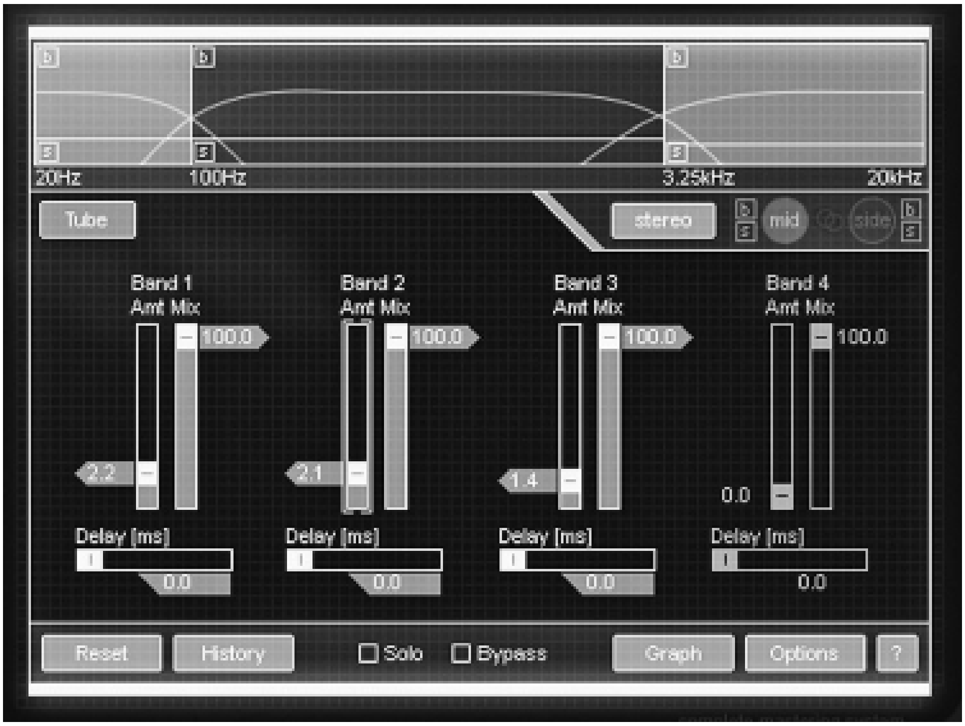


FIGURE 11.9

Using the harmonic exciter

The exciter is similar to an equalizer, but instead of boosting or reducing existing harmonics, new harmonics are added to a signal. So, instead of boosting high frequencies with an equalizer, try using the harmonic exciter on the same frequency band. The effect is subtly different.

Multiband Stereo Imaging

The multiband imaging tool is useful for several tasks. Its primary function is to adjust the spaciousness or width of a mix, but the tool can also be used to narrow the stereo image. One benefit of multiband imaging is that midrange signals can be widened without affecting bass frequencies. This is beneficial since a focused bass sound is typically desirable in popular music.

Another interesting application of the imaging tool is to change the perceived location of a stereo image. This is accomplished with the delay sliders at the bottom of the screen. One of the ways we perceive the location of a sound is by identifying a difference in intensity between our left and right ears.[6] The difference in loudness between a signal on the left and one on the right is the principle behind panning. Location may also be determined by the amount of delay it takes for a signal to reach one ear and then the other. For example, a sound that emanates from our right will arrive at the right ear slightly earlier than at the left and, thus, provides a clue as to the location of the sound. In the multiband stereo imaging tool the delay sliders can be used to create a similar effect. Select the Group All Band Delays toggle button and drag a delay slider to the left or right, and you will hear the image shift slightly to one side.

Loudness Maximizer

The loudness maximizer (figure 11.10) is typically placed as the last stage of the mastering signal flow. Its job is to ramp up the volume of a mix while limiting the signal and preventing distortion, but the maximizer can also be applied in a more subtle way to add body to a mix. As I have mentioned previously, the dynamic range of commercial popular recordings is typically compressed to an extreme. The subjective response of most listeners is that these tracks sound "louder." The maximizer can be used to optimize the level of a recording in this way, but the amount of dynamic processing should be determined by individual taste and, to an extent, musical genre. Threshold is the key parameter of the loudness maximizer. Lower values mean that more of the mix is being limited, providing a more pronounced effect.

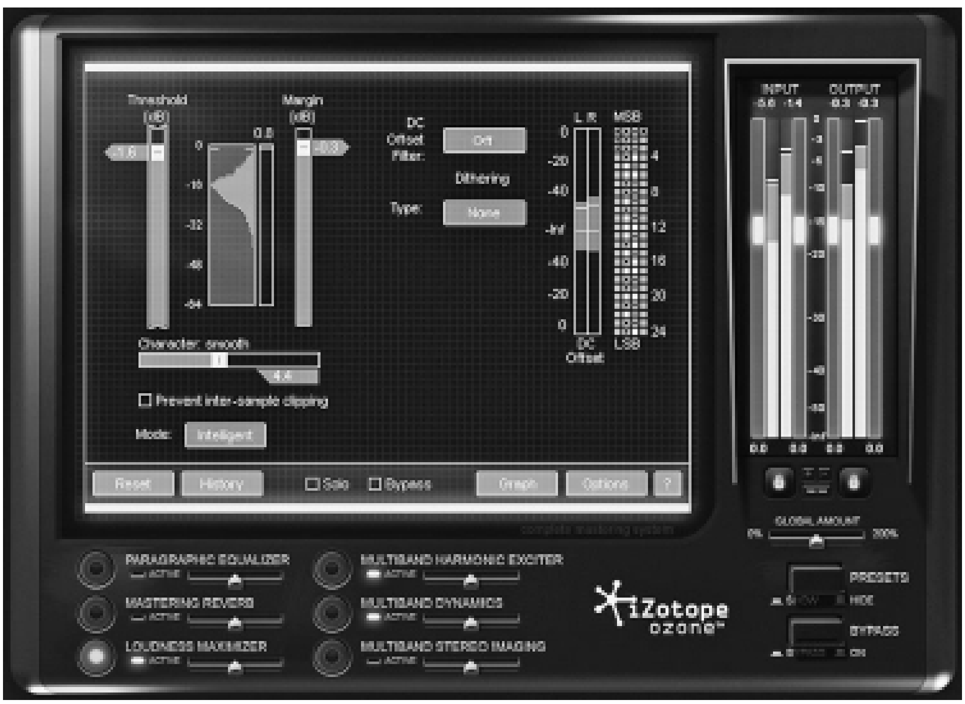


FIGURE 11.10

Loudness maximizer

Practical Application

Up to this point, the focus has been on the function and operation of the tools in the Ozone mastering suite. The following paragraphs are devoted to a discussion of the use of these tools in the context of mastering a song. Keep in mind that there is no "correct" approach to mastering. There are many ways to approach the task, and I would encourage you to experiment in order to develop a sense of the capabilities of the software.

There is a fair amount of overlap between the process of mixing and that of mastering music. The ultimate goal of a mix is that is should sound great on any playback system: no band of frequencies should overpower or be underrepresented in the mix, levels and dynamics should be pleasing to the ear, and each element of the mix should be clear to the listener. Although those considerations are also a part of mastering, a mastering engineer may add an extra element of polish or sweetening. This might be in the form of sparkle that is added to a band of frequencies using a harmonic exciter or a subtle use of ambience to provide a pleasing wash to the mix. The mastering engineer might apply compression to a band of frequencies to tighten the bass or adjust the spaciousness of the mix with an imaging tool. The sweetening process takes a certain amount of objectivity so it is usually best to avoid these decisions during the mixing process. My advice is to focus on making the best possible mix and, when possible, give yourself a few days before you revisit the mix with the ears of a mastering engineer. This can help you to have more objectivity and a fresh set of ears to apply to the mastering process.

At the companion Web site, you will find an excerpt from an original composition in both its raw mix and its premastered form. I utilized Ozone to improve the recording in several ways, as detailed below.

The original track seemed slightly dull, so a first step was to use the paragraphic equalizer to add some sparkle. I engaged the mid-side mode of the equalizer and used a high shelving equalizer to accentuate frequencies starting around 10 kHz. I used a wide bell curve to attenuate frequencies around 1.5 kHz in the center channel because the saxophone seemed too bright in the context of the mix. A subtle amount of compression (in stereo mode) was added to invigorate the track.

I engaged the reverb unit and added a small amount of plate reverb (13% wet) to the signal. Although the mastering reverb is not readily apparent in the recording, it added a subtle and pleasing wash to the track.

I used the tube mode of the multiband harmonic exciter to add some simulated tube color in varying amounts to the three primary frequency bands. I have found that it is a good idea to solo each frequency band when using the harmonic exciter as it is very easy to overdo the effect and cause audible distortion in one of the frequency bands. In this instance, the bass frequencies sounded distorted when I soloed this band in the harmonic exciter.

It was interesting to explore the multiband stereo imaging tool. I used the tool to create a wider image in the middle- and high-frequency bands. This had the effect of making the mix sound bigger or more spacious. A final stage involved the loudness maximizer. A threshold of –1.6 dB with a default margin of –0.3 dB seemed to work well for the track.

Although there are many ways to improve the original track, the steps detailed in this section enabled me to create a commercial sound that seemed appropriate for the genre. You can listen to the premastered and mastered versions of the track at the companion Web site.

EXAMPLE 11.1

Presets are a good starting step for Ozone or any other mastering or mixing tool. I often begin with a preset that sounds good and add subtle tweaks to optimize the settings for the given sound file.

Creating a Red Book CD

A compact disc is the culmination of most recording projects. The CD might be used to duplicate a few "one offs" for friends and family or it might be used by a CD manufacturer to develop a glass master for replication. Free applications such as Windows Media Player and iTunes will be sufficient for casual projects, but you will want to change several settings to optimize these programs for the best audio quality. In Windows Media Player, select the Rip Music tab and select WAV (lossless) for the format. By default, Windows Media Player will compress audio files that are added to the library so this setting forces the program to store the files in an uncompressed format. You will also want to turn off volume leveling from the Burn tab so that the Windows Media Player doesn't automatically alter the level of tracks when they are burned to disk. The process is similar for iTunes: select the Preferences menu and click on the General tab. Select the Import Settings button and select the AIFF or WAV encoder for lossless import. You will also want to turn off Sound Enhancer and Sound Check from the Playback tab to prevent iTunes from altering the EQ or level or your tracks.

Applications such as CD Architect (Windows) and WaveBurner (Mac) are used by many audio professionals and provide a greater measure of control over the burning process. These programs are designed to produce Red Book compliant CDs—the standard audio CD format—that are suitable for use as premasters. Key features of professional CD burning software include the ability to insert arbitrary track numbers, adjust the relative level of tracks, trim and normalize clips, and create crossfades. These applications can also be used to insert Universal Product Code (UPC) and European Article Number (EAN) barcode data as well as International Standard Recording Code (ISRC) subcodes that are used to digitally fingerprint individual tracks. (Visit www.usisrc.org for more information about registering ISRC subcodes.) Programs like CD Architect and WaveBurner can also be used to create CD Text, which is used to display song titles and other information on some CD players. However, it is important to note that most players do not read CD Text. iTunes and other media players obtain song names and other information from an online database called Gracenote.

There are a number of artistic decisions that must be made when creating an audio CD. The order of songs is an obvious consideration, but the amount of space between songs and the relative level of each song is also very important.

For example, a normalized ballad might sound unnaturally loud when placed in the context of a playlist. I find it helpful to play back a few seconds of each track starting at random points in the track in order to get a sense of the average level of the songs. Adjustments can then be made to ensure that tracks flow from one to another in a pleasing way, both musically and dynamically. When possible, it is also helpful to set a CD aside for a few days in order to revisit a project with fresh ears.

Conclusion

Although the process of mixing and mastering can seem daunting, it can also be very rewarding. Unlike a professional engineer, the home recordist can take the time to explore a number of techniques and strategies without worrying about the clock. Just as beginning orchestrators learn to score by writing for duets and trios, you may find that it is helpful to start with smaller projects to hone your skills. You might also consider offering to host a free recording session for a singer/songwriter or group of students in order to gain experience. I still enjoy doing occasional projects *pro bono* when an interesting opportunity presents itself because I know that I will learn something new and improve my skills with each session.

12

Do-It-Yourself Projects

This final chapter consists of do-it-yourself projects for the home studio. One of the many benefits of doing the work yourself is that you can design a rack or build a computer that is perfectly suited to your needs. The process also affords the opportunity to balance variables including form, function, and aesthetics. For example, a perfectly adequate mixing desk can be built for around $100, but a desk with hardwood trim and sliding drawer will cost more to build. Perhaps the biggest benefit of do-it-yourself projects is the satisfaction that comes from building a useful item for a modest price.

I purposely selected projects that would be reasonable for readers with a minimal background in woodworking. With that said, it is important to read and understand the user manuals for any power tools you intend to use, and it is essential to wear appropriate hearing and eye protection. You should also understand that the projects presented in this section are a value-added feature for the reader. I am not a structural engineer or designer and, as such, make no warranty in terms of the design or use of these projects so use these plans at your own risk.

I created the illustrations for most of these projects using a free computer-aided design program called SketchUp. I would encourage you to download the program at sketchup.google.com and use it to create your own versions of the projects. It's great to be able to see what a slightly rakish angle of a rack or curve on a desk will look like prior to spending the time and money to create a prototype.

PC Computer

Building a custom computer can make a great deal of sense for many musicians. Although you won't necessarily save money when comparing your parts list to the budget offerings from manufacturers such as Dell, you will be able select just the right mix of components for your intended application. You will also have the benefit of being able to easily upgrade the computer since the components are not proprietary.

No expensive tools or experience are required to build a custom computer. The hardest part of the process is striking a balancing between your budget and your objectives when developing a parts list. Be sure to read chapter 2 of this book, as the information from that chapter will provide a foundation for this project.

You will find a great deal of information from the online gaming community, but it is important to make some distinctions between a gaming system and one that is intended to be used for digital audio recording. Many gamers *overclock* their computers so that the CPU runs at a faster rate than it was designed for. This can generate additional heat so more cooling (likely involving louder fans) may be necessary, which creates obvious obstacles in a recording environment. Not only can overclocking can cause stability problems, it can also damage components. For these reasons I would recommend against overclocking if your goal is to run a stable recording system.

The selection of a video card is another primary difference between recording and gaming systems. Extremely fast video cards run hot and typically require more aggressive cooling to keep temperatures in a reasonable range. This is often translates to excessive noise from gaming video cards.

One way to select components for a computer system is to reverse-engineer your parts list from the system requirements that are available from hardware and software vendors such as Digidesign or Steinberg. Products like Pro Tools have very specific hardware and operating system requirements so you will obviously want to consider these requirements if you are building a system for a particular application.

Mobo

The motherboard, or "mobo," is a good place to start when developing a parts list for a custom PC. Motherboards are available in two broad categories: those that support AMD processors and those that support Intel processors. Although many motherboards are inexpensive, CPUs can be quite expensive so you might want to consider these items in tandem. For example, a Single-Core Intel Celeron 1.8 GHz processor is slow by current standards but costs only $40. At the other extreme, an Intel Core 2 Quad 3 GHz processor may cost upwards of $300. You will need to carefully evaluate the specifications of the motherboard to verify that the given processor and socket type are compatible. As an example, the LGA 775 socket supports a number of Intel processors.

Form Factor

The term *form factor* refers to the physical dimensions of the motherboard and case and is an important consideration. The ATX form factor is most common, but other options such as Micro ATX are available if you intend to build a computer in a small case.

Expansion

The type and number of expansion ports is a key consideration for musicians. For example, a given motherboard will typically provide one PCI Express 2×16, one PCI Express ×1 slot, and two or more PCI slots. Some audio interfaces require the installation of a PCI card and PCI slots may also used to install components such as a wireless network card.

Other expansion considerations include the type of RAM, as well as the maximum amount of RAM supported by the system. Also, consider necessary rear-panel ports such as FireWire (IEEE 1394) and USB 2, as well as compatibility of data storage (usually PATA/IDE and SATA 3 Gb/s). Once you decide on a mobo and CPU, you will be able to consider the other necessary components, which will minimally include a case, power supply, RAM, hard drive, optical drive, and operating system.

Computer Case

A computer case and power supply can be purchased as a unit or separately. For convenience, I elected to purchase a computer case with a pre-installed power supply. A power supply in the range of 450 W should be adequate for a basic setup, but many gamers purchase additional power to support multiple drives, high-end graphic card, and the like. Case ventilation is also a consideration. Most cases come with several fans that are used to keep temperatures an appropriate level within the case. One of the things I learned in building my first computer is that larger fans (120 mm) are better since smaller fans tend to generate more noise.

Video Card

I do lots of orchestral writing so I wanted a video card that would support a dual-link display should I decide to add one in the future. Many readers could save money by purchasing a less expensive card since audio recording is not a graphically intensive activity.

Table 12.1 lists the parts that were the basis for my computer build. My original intent was to test a chip called the EFi-X that purports to offer Macintosh compatibility, so most of my hardware selections were informed by the

TABLE 12.1 Parts list

Description	Item	Cost
Case	CASE ROSEWILL|R5717 BK 450W RT	$60
Motherboard	MB GIGABYTE GA-EP45-UD3R P45 775 R	$120
CPU	CPU INTEL|C2D E8400 3G 775 45N R	$168
Memory	EM 2Gx2|GSK F2-8500CL5D-4GBPK R	$59
Wireless card (PCI)	WL ADAPTER ENCORE|ENLWI-G(2) 54M R	$13
Hard drive	SATA 500 GB, 7200 RPM	$50
Operating system	Windows Vista Home Premium*	$100–$225
Thermal paste		$10
Video card	9800 GT 512 MB video card	$105

*Many audio professionals preferred to stick with Windows XP when Windows Vista was released since device drivers were generally reliable in Windows XP. However, Windows XP is at the end of its lifecycle, so I would recommend Windows 7 for most applications. Linux would be a good choice for some readers, but lack of support for audio hardware is still a problem. As you can see, the operating system is an expensive item. Academic pricing may be available for some readers or you might want to consider an original equipment manufacturer (OEM) version. An OEM operating system is designed for manufacturers and system builders and is typically sold with at least one hardware item from vendors such as Newegg.

specifications listed by the manufacturer of the compatibility chip. Unfortunately, the manufacturer of the EFi-X chip was unable to loan one of the chips to me as part of my research for this topic. However, you might want to explore this option if you are interested in building a system to run OS X.

There are many other items to consider, such as a CD/DVD burner or secondary hard drive. I happened to have an old IDE CD burner and DVD player and was able to recycle those parts by using them in this project.

Building the Computer

It goes without saying that you should never work on a computer or any other electronic device that is connected to a power source. You will also want to consider wearing an antistatic wrist strap to avoid damaging an electrical component with an accidental static discharge.

A first step for me was to apply thermal paste to the top of the CPU prior to installing the CPU in the motherboard. (Make sure you apply the paste to the correct side of the CPU!) It was also necessary to remove a socket protector from the motherboard prior to installing the CPU (see figure 12.1). The CPU came with a cooling fan, which was installed by inserting four pegs into special holes surrounding the CPU.

One of the questions I had about building a computer concerned the relationship of the USB, FireWire, and other ports on the motherboard with the cutouts on the computer case. This became clear once the parts arrived, as the manufacturer of the motherboard provided a metal plate with cutouts for all of the ports.

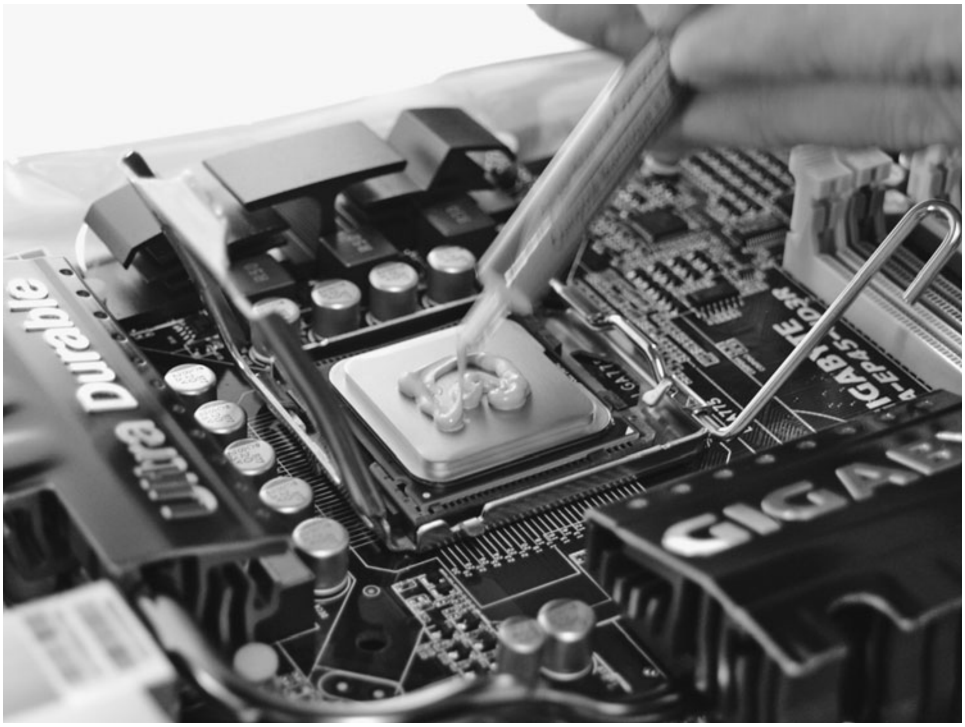


FIGURE 12.1
Installing the CPU

Although the instructions that came with the motherboard didn't indicate that risers were necessary, it seemed fairly obvious that I should screw the included risers to the chassis and attach the motherboard to these lifts (see figure 12.2).

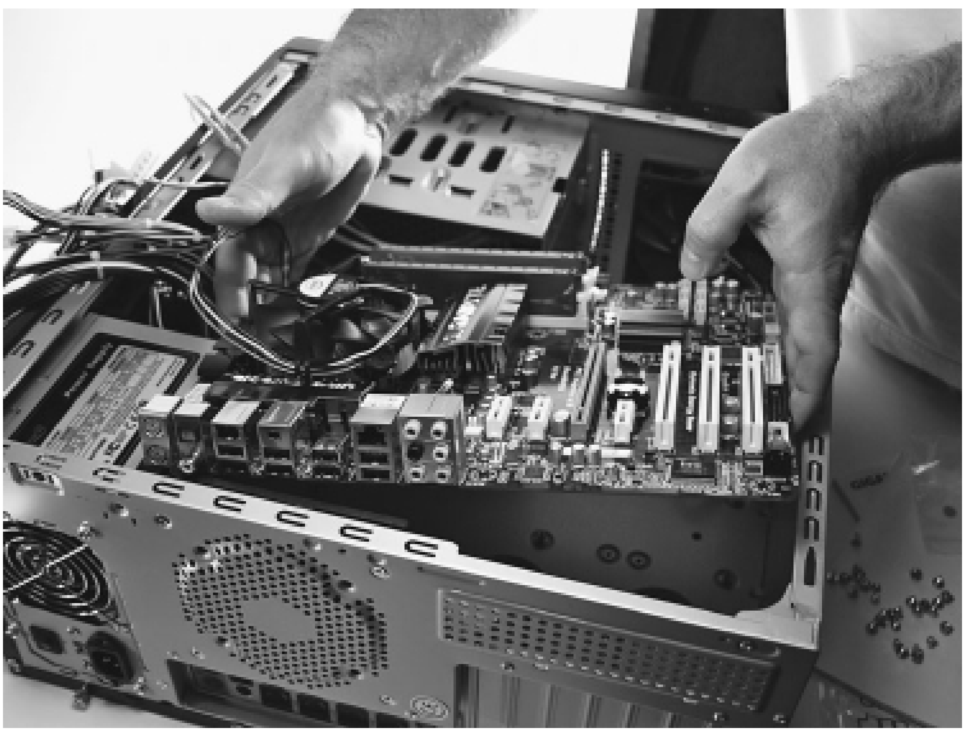
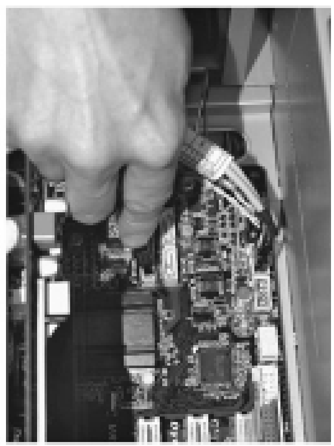


FIGURE 12.2
Installing the motherboard

A next step was to install RAM and the hard drive. The instructions indicated that, for a pair of RAM, the RAM should be installed in either the first and third slots or the second and fourth slots. Installation was simply a matter of aligning the RAM and pressing firmly until the part snapped in place. A hard drive cage made it easy to install the SATA hard drive. Screws were used to connect the drive to the cage, and the cage was inserted in the case and screwed down so that it wouldn't shift.

Once the hardware was installed, it was time to connect the components with cables. This was largely an intuitive process as the motherboard was clearly marked with labels to indicate where cables for the CPU fan, drives, and other items should be attached. IDE drives were attached with the flat drive cables, shown in figure 12.3. Note that it may be necessary to change a jumper pin on your hard drive when attaching an IDE drive. The jumper settings are usually listed on the drive itself. SATA drives are connected via the smaller cables shown in figure 12.3.

IDE cable



SATA cable

FIGURE 12.3

IDE and SATA cables

Once the hardware was installed and connected, it was time to test the machine. I plugged in the power, monitor, keyboard, and mouse and crossed my fingers as I turned on the power switch. The motherboard fans powered up but the computer did not start. After some additional research I found that it was necessary to make two connections between the motherboard and power supply.

Installing the OS

Once the computer was able to boot into BIOS, it was time to install the operating system. I inserted the installation disk and rebooted the computer, and I was presented with the installation screen for Windows Vista. At this point, installation was simply a matter of following the prompts. After Windows was successfully installed, I inserted the disk from my motherboard manufacturer and installed the necessary drivers. It was then possible to configure the wireless card and retrieve the latest drivers for the video card from the manufacturer's Web site. The final step was to download the latest drivers for all of my audio hardware and install software so I could utilize the computer in my recording studio.

Other Observations

At first I was dismayed at how loud the computer was until I realized the noise was coming from the inexpensive hard-drive tray I had installed in one of the drive bays. The computer was much quieter after I removed the tray and, though the computer is somewhat louder than my Mac desktop, it is not unduly loud. I may install better fans in an attempt to tone this down even more.

This project would be reasonable for anyone with a basic knowledge of the components of a computer. The entire assembly took an evening (and I am sure a second build would be much faster). I was particularly pleased with the Gigabyte motherboard because it appeared to be of high quality and came with reasonably detailed instructions. The final computer assembly can be seen in figure 12.4.

FIGURE 12.4

Final computer assembly

Speaker Stands

Attractive speaker stands are easy to build and require a minimum number of tools. In fact, a circular saw, drill, and screwdriver are the primary tools needed to build the speaker stands shown in figure 12.5.

For this project I purchased a 4 × 8-foot sheet of ¾-inch medium-density fiberboard (MDF) since it would provide plenty of material for two stands with enough left over for other projects. A first step was to rip the material into the widths listed in figure 12.6. You might want to experiment with other proportions:

Ripping can be accomplished with a circular saw and a straight edge. As is evident in figure 12.7, a second sheet of MDF can be used as a fence.

After ripping the parts, I used a circular saw and straight edge to cut the parts to length. Most commercial stands range in height from 38 inches to 40 inches and I elected to cut the

FIGURE 12.5

Finished speaker stands

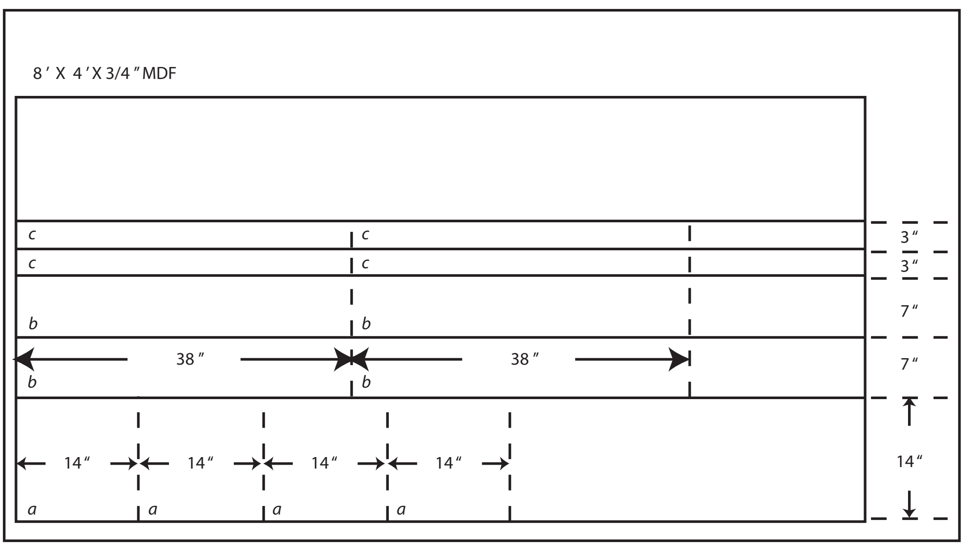


FIGURE 12.6

Speaker stand cutout list

FIGURE 12.7

Ripping MDF with a circular saw

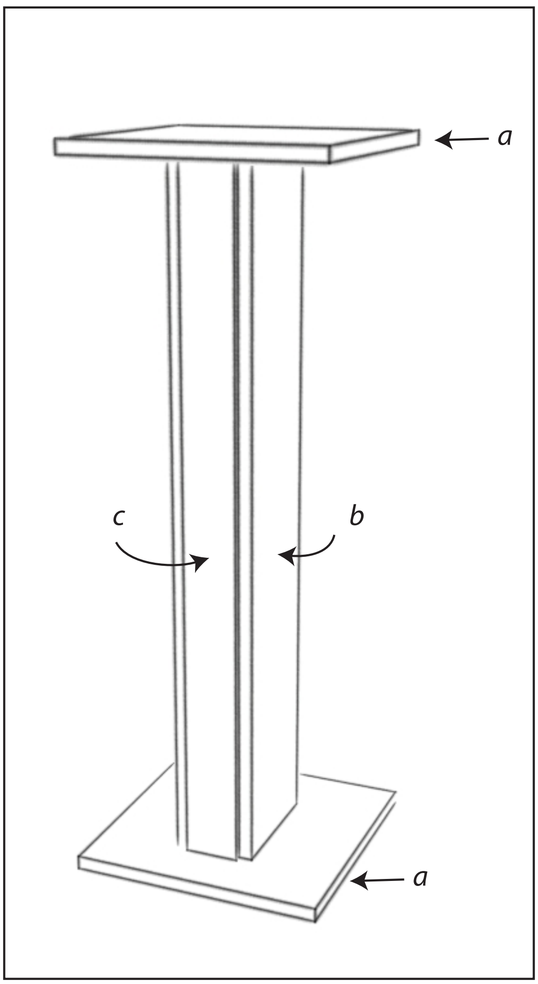


FIGURE 12.8

Stand assembly

parts for the vertical column to 38 inches to yield a total height of 39½ inches including the base and top.

Once the parts were ripped and cut to length, it was time to assemble the vertical column, as shown in figure 12.8.

The vertical parts could be joined together in a number of ways, including a combination of glue with nails, 2½-inch wood screws (or longer), or biscuit joints. In their book *Woodworking Simplified*, David and Jeanie Stiles recommend using "a screw that is two to three times as long as the thickness of the wood into which you are screwing."[1] A very strong joint, called a dado, can also be made by routing a ¾-inch groove and using screws and glue to fasten a second piece of MDF perpendicularly in the groove. Whichever method you select, it is essential to use a combination of glue and fasteners of an appropriate length so that the columns are securely joined. I happen to own an inexpensive air compressor and nail gun, and the combination glue and nails worked very well for the speaker stand column. Wood screws also work well and the countersunk holes can be covered by applying multiple coats of wood filler. Be sure to sand between coats in order to create a smooth surface for finishing.

Before attaching the base and shelf, it is necessary to center the column on the base in order to do some drilling. Start by marking the center of each side of the column and scribe two lines marking the center of the base and shelf. With those markings in place, it is simple to align the column in the center of each part (see figure 12.9).

FIGURE 12.9

Aligning a column with the center of the base

Place the column in position and scribe a line around the center column. These markings can be used to drill guide holes in the base, but remember to drill about ⅜ inch *on the inside or toward the middle* of each guide line. At this point, the assembly can be turned on end so that guide holes can be drilled through the base and tops into each end of the speaker column. I used glue and six 2½-inch wood screws on each end and the assembly feels very stout (see figure 12.10). Guide or pilot holes are important not only as an aid in drilling screws but also to create stronger connections.[2] I *would not* trust nails for this part of the assembly as nails do not have the holding power of appropriately sized screws.

FIGURE 12.10
Attaching a top to a column

You might want to consider painting the pieces or adding a veneer prior to assembling the parts. Other enhancements would include a rubber surface on the top of each stand or a cable grommet in the middle of each shelf or near the top of the column so that cables can be run through the inside of each column. Decorative molding could also be applied to the top shelf. I elected to use a chamfer router bit (see figure 12.14) to add a decorative touch to the top of the base and speaker shelf. Be sure to thoroughly test each speaker stand prior to trusting the stand to hold an expensive studio monitor.

Mixing Desk

Purchased mixing desks can be very expensive but a simple desk can be built for a modest price. I debated over the best way to approach the base of the desk. One approach is to build two equipment racks of suitable height and fasten the top to the racks. I have used that approach in the past, and it works very well. However, I wanted to present a useful project that would be within the means of even the most modest woodworker so I elected to use commercially available legs for the base of the desk. Keep in mind that this approach has less lateral support than using a pair of racks. I ordered four desk legs from Closet Masters Inc. and was very pleased with the quality of the legs and the price was a modest $13 per leg. I used MDF to build the prototype shown in figure 12.15. Even though the prototype was reinforced, it still exhibited a ⅛-inch sag. Sag could be minimized with a combination of reinforcement and hardwood trim. Another option would be to build a base consisting of a pair of equipment racks. With that said, sag is not excessive and the prototype has stood up well to continuous use.

A first step is to consider the dimensions of the top of the desk. My goal was to have enough space for two racks and a 23-inch monitor (approximately 20 inches on the horizontal) so I chose a width of 5 foot 6 inches and depth of 2 feet 4 inches. The cutouts for the desk are shown in figure 12.11. Start by rip-

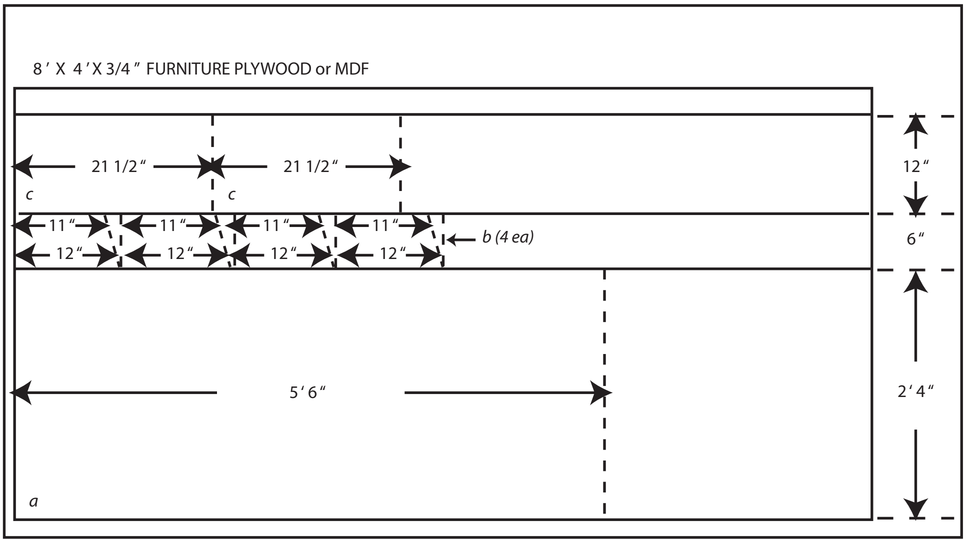


FIGURE 12.11
Cutouts for a mixing desk

ping the ¾-inch material lengthwise to create a piece that is 2 feet 4 inches × 8 feet. Cross-cut this board to create a desk top that is approximately 5 feet 6 inches wide.

Rip another full-length piece with a width of 6 inches (or wider if you wish to have taller racks). Crosscut the 6 inch × 8-foot board to create the sides for four racks. I like a slightly angled look so I elected to cut the racks with a width of 12 inches at the base and 11 inches at the top. You should be aware that there is a tradeoff here between aesthetics and function since an angled rack rail will allow room for only a very narrow item in the bottom rack space. Practically speaking, the small racks have room for a two-unit rack item. Alternatively, just make a straight crosscut so that equipment can be racked horizontally.

Rip the remaining board to make two rack tops. I would suggest an internal width of 19⅛ inches for the racks so add an additional 1¼ inches, as well as an additional amount for an overhang. At this point the rack can be assembled with screws and glue. The parts will fit together as shown in figure 12.12.

I suggest installing four L brackets in each rack for extra support. For a stronger joint, router two grooves in the bottom of each rack top and fasten the sides to this dado joint with glue and screws. It is important to be accurate in measuring the internal width of the racks as your equipment will not fit if the opening of the rack is too small. I have found an internal width (the distance between the inner walls of the rack) of 19⅛ inches works well and leaves a small margin for error.

MDF has a tendency to sag so it is essential to add some extra support if you use MDF to make a desk of this dimension. Channel strut, available in the electrical section of most hardware stores, is relatively inexpensive and provides a reasonable amount of support. I purchased a 10-foot length and used a hacksaw to cut the piece into two lengths in order to make the two supports shown in figure 12.13.

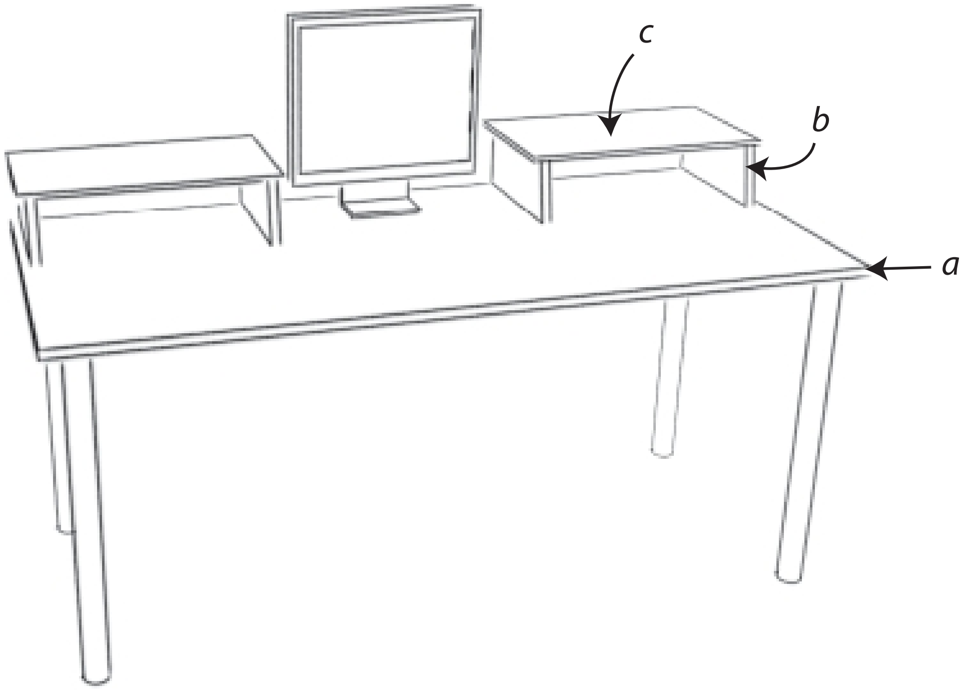


FIGURE 12.12
Desk and rack assembly

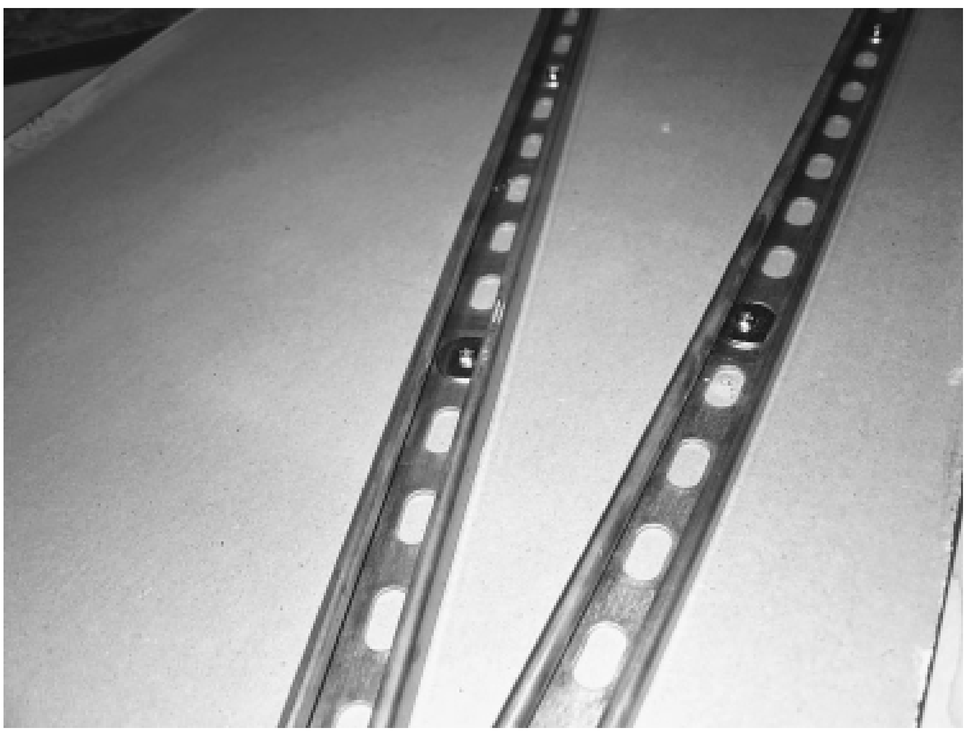


FIGURE 12.13
Attaching supports to the underside of the desk

The pre-drilled holes on the channel strut were large so I used washers and wood screws to fasten the hangers to the underside of the desk. Hardwood trim would provide additional support but is expensive.

At this point the desk is nearly complete. Mark the location of the legs on the bottom of the desk and attach the legs with fasteners.

There are many ways to approach the finish of the desk. I used a round-over bit (see figure 12.14) to add a smooth edge to the desk. The desk could be trimmed in hardwood or a wood skirt could be added to provide a more finished appearance. Another option is to use leftover material to make a sliding tray for a typing keyboard.

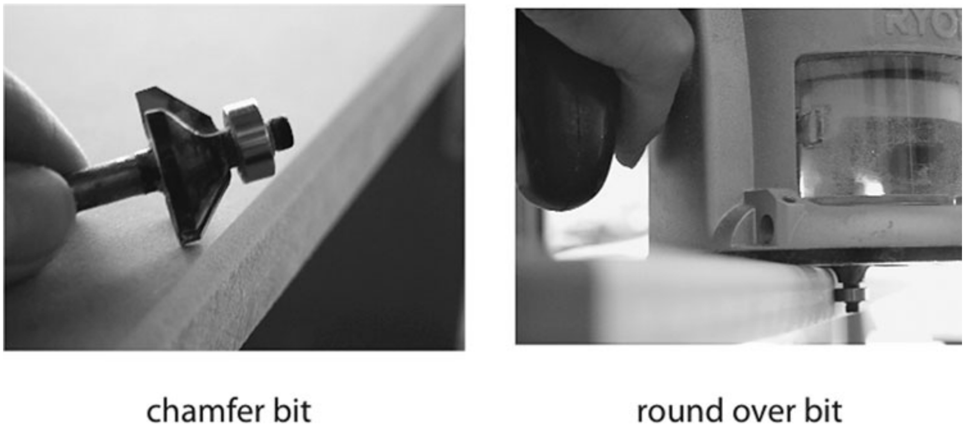


FIGURE 12.14
Chamfer and round over bits

One simple finish method is to apply a coat of primer paint and then several coats of an attractive high-gloss paint. The completed project is shown in figure 12.15

FIGURE 12.15
Completed desk

Equipment Rack

It is easy to build an attractive equipment rack. As with the previous projects, the only required tools are a circular saw, drill, and screwdriver. I decided to make two small racks from a single sheet of ¾-inch MDF, but I would encourage you to consider if a different size would better suit your workspace. The only dimension that really matters is the internal width of the rack, but it is also a good idea to check the depth of any items you intend to rack. The cutting list is shown in figure 12.16.

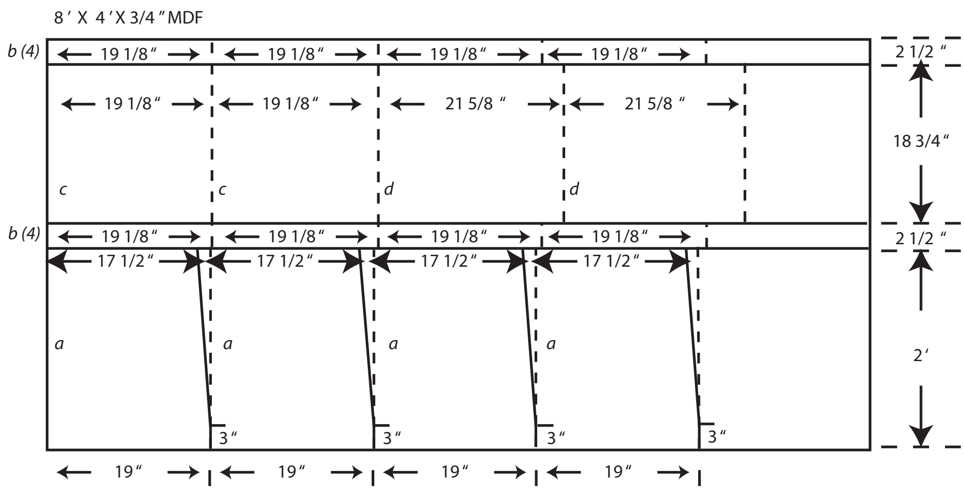


FIGURE 12.16
Cutting list for two racks

Start by ripping a length of MDF in order to make the sides of the rack. I decided to make the sides 2 feet high so I used a straight edge and circular saw to rip a 2 foot × 8-foot board. I cut the board to make four 2 foot × 19-inch sides. An optional step is to rake the front edge of the rack sides. I measured up 3 inches and angled each side from 19 inches at the base to 17½ inches at the top.

Rip a 2½-inch strip from the MDF to make four braces 2½ inches × 19⅛ inches. It is important to make the braces at least 19⅛ inches so that your rack will be wide enough for equipment to fit within the rack. Although a standard rack is 19 internal inches, the extra ⅛ inch is negligible and provides a small margin of error to guard against building a rack that is too narrow.

Rip another strip to leave a piece that is 18¾ inches × 8 feet. The large piece will be used to cut the tops and internal shelf, and the small strip can be used to make four more braces that are 19⅛ inches in length. The height of the braces need not be exact as long as you pair braces of equal height when you assemble the rack, but it is important to be accurate when cutting the braces to length. The parts fit together as shown in figure 12.17.

To assemble the rack, stand one of the sides on a flat surface and use a square or a factory-cut corner of MDF to align one of the braces so that the brace and side form a right angle. Set one of the 19⅛ inch × 18¾-inch shelves

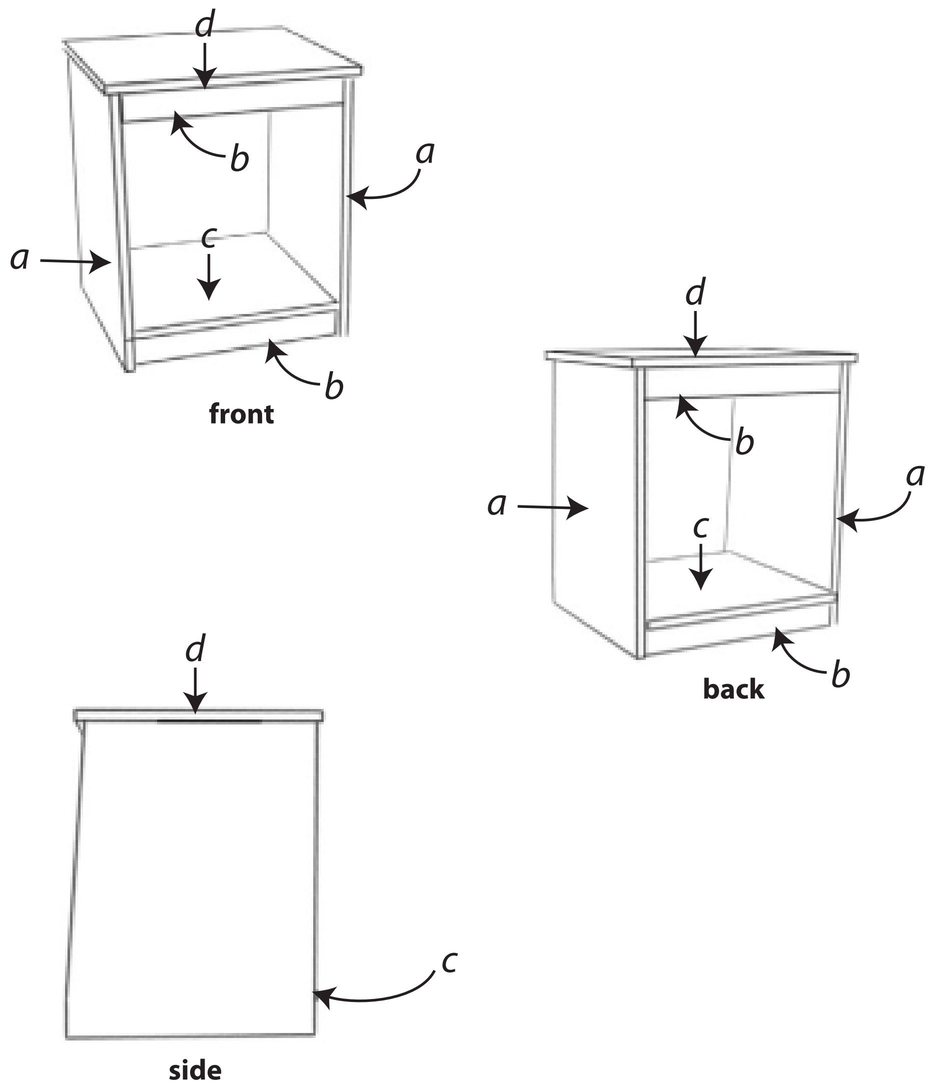


FIGURE 12.17
Rack assembly

in the rack to use as a guide for the placement of the bottom braces. I elected to place the bottom brace back a few inches from the front of the rack in order to create a more attractive look. Note that you might need to trim the bottom shelf if you rake the front of the rack.

Use a drill to create two guide holes for 2½-inch screws and fasten the braces with glue and screws. Alternatively, use a nailing gun and glue to fasten the parts. Check frequently with a square or factory-cut corner to make sure the rack is square during the assembly process.

Attach the bottom shelf to the top of the braces with glue and screws. Align the top of the rack so that there is an even overhang and fasten to the racks sides and braces. An optional step is to use a chamfer bit to router a beveled edge on the top of the rack. Alternatively, hardwood trim can be installed for a very attractive look. You might also want to consider installing wheels to the underside of the bottom shelf. The completed rack is shown in figure 12.18.

FIGURE 12.18
Completed rack

Rack Rails

Rack rails can be constructed from 1/16-inch aluminum angle offset, which is available in most hardware stores. Note that aluminum angle offset stock is suitable only for fairly light rack items and the holes are easily stripped if you use a power drill to fasten the bolts. I would advise purchasing commercial rails if you need support for heavier items as it is very difficult and time-consuming to drill and tap holes in heavier stock.

Use a hacksaw to cut the aluminum bar to an appropriate length. It will be necessary to use a 10 × 32 tap set so that standard rack screws can be screwed into the holes. I purchased a 10 × 32 tap for around $5 and spent an additional $5 for a tap handle.

The easiest method for marking the position of the holes is to use an existing rack rail as a template. If you don't already own a rack rail you can try a center hole spacing of 1¼–½–1¼–½ inch, but be sure to check these measurements against the equipment you intend to rack before spending the time and expense to manufacture a rack rail. Use a felt-tip pen to mark the location of the holes and drill the holes with the drill bit that comes with your tap set. Next, use the tap to create threads in each of the holes (see figure 12.19).

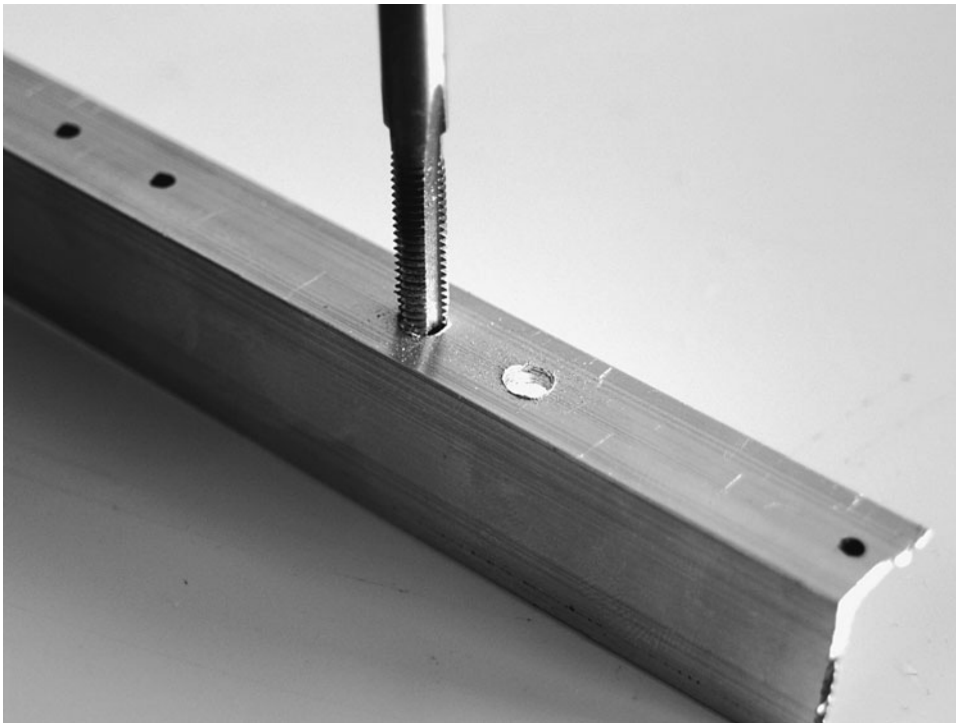


FIGURE 12.19

Preparing rack holes with a tap

Absorber

An attractive midrange absorber can be created by building a simple frame around a length of Owens Corning 703 insulation. A frame can be built from inexpensive pine stock, or you might want to purchase hardwood from a specialty store. I do not own a table saw but was fortunate that my father could rip wood to a suitable thickness. Many wood stores are willing to cut and rip stock for a fee so you should be able to gather parts for the project for a reasonable cost.

Use a hand or electric miter saw to cut a 45 degree angle on one end of the wood. Mark an appropriate length from the inner (short) edge of the cut and make another 45 degree cut *away* from this mark (e.g., the short sides of the cut will determine the internal dimension of your frame). Repeat this process for the other sides and fasten the pieces with glue and nails (see figure 12.20).

Cut some attractive cloth so that it overlaps each edge of the frame and use a heavy-duty stapler to attach the cloth to the frame. It is difficult to make the fabric cover taught using only cloth and staples, but plumber's pipe strapping can be used to stretch the material for a more professional look (see figure 12.21a). The extra cloth can be trimmed or tucked inside the frame.

Next, cut a length of Owens Corning 703 and place it in the frame (see figure 12.21b). Staple retaining straps on the back of the frame if necessary and hang the completed absorber in your studio. Although the wood frame shown in figure 12.22 is attractive, OC 703 insulation is rigid so another option would be to sew an attractive slip cover to place around the insulation.

FIGURE 12.20

Absorber frame

FIGURE 12.21a

Installing cloth inside of the absorber frame

Balanced Cables

Balanced XLR and ¼-inch cables are relatively easy to make. Mogami and Canare are popular brands of cable, and I have been very pleased with Canare Star Quad, which sells for 40 to 50 cents per foot. Quality connectors by Neutrik can be purchased from many online vendors for a modest price.

FIGURE 12.21b

Continued

FIGURE 12.22

Completed absorber

Cut the cable to length and slide a bushing and cable relief chuck onto each end of the cable as shown in figure 12.23. It's easy to forget to slip these parts on the cable prior to soldering so I am in the habit of installing the parts on *both* ends of the cable prior to stripping and soldering the wires.

FIGURE 12.23

Bushing and cable relief chuck

Next, carefully use a utility knife to remove the outer shield from the end of the cable. One technique is to bend the cable and gently score the outer loop with a knife. Repeat on four sides and the shield can be easily removed.

Carefully use a wire stripper to remove a length of shield from the end of each of the inner cable wires. Note that the Neutrik assembly instructions indicate that 18 mm (0.71″) and 4 mm (0.16″) should be removed from the outer and inner shields respectively (see figure 12.24).

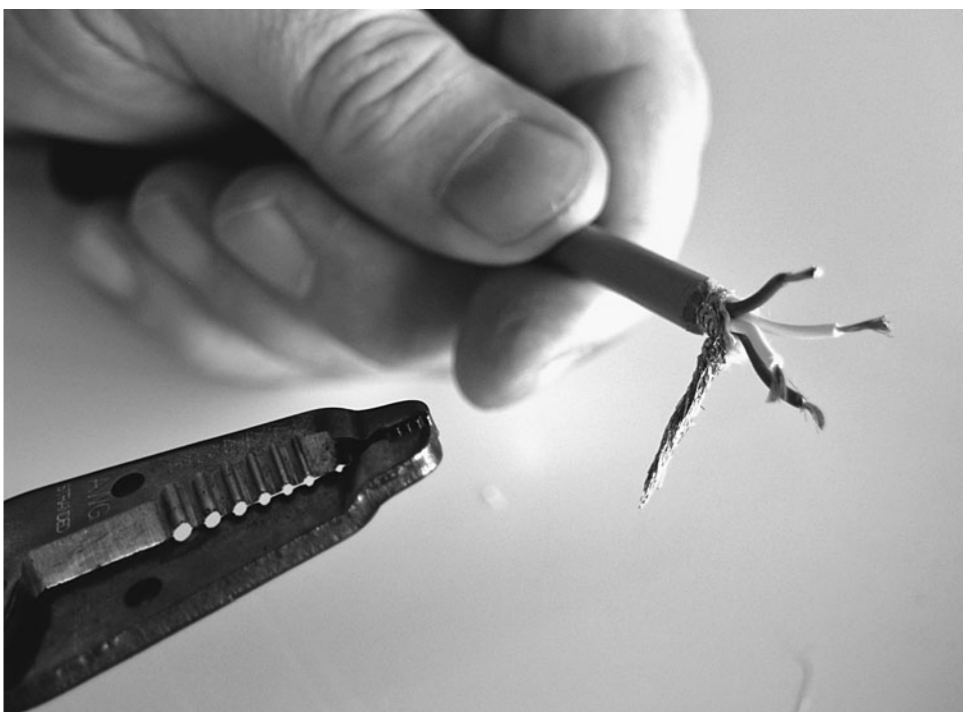


FIGURE 12.24

Removal of the inner and outer shields

Next, "tin" the bare ends of the wires by applying a small amount of solder with your soldering iron. (Twist pairs of wires together prior to tinning if you are using a "quad" cable such as the Canare Star Quad.) The key to soldering is to use the soldering iron to heat the wire or receptacle as opposed to trying to melt the solder with the iron. Hold the iron against the wire and touch the solder to the wire. The solder should flow onto the wire, thereby tinning the end of the wire.

A final step is to insert the wires into the appropriate receptacle in the plug or jack and solder them in place. A small vice or clamp can be used to hold the plug in position and an alligator clip can be used to hold the wires in place. Once the wires are in position, apply the iron to a receptacle and hold the solder so that it melts and fills the wire receptacle (see figure 12.25).

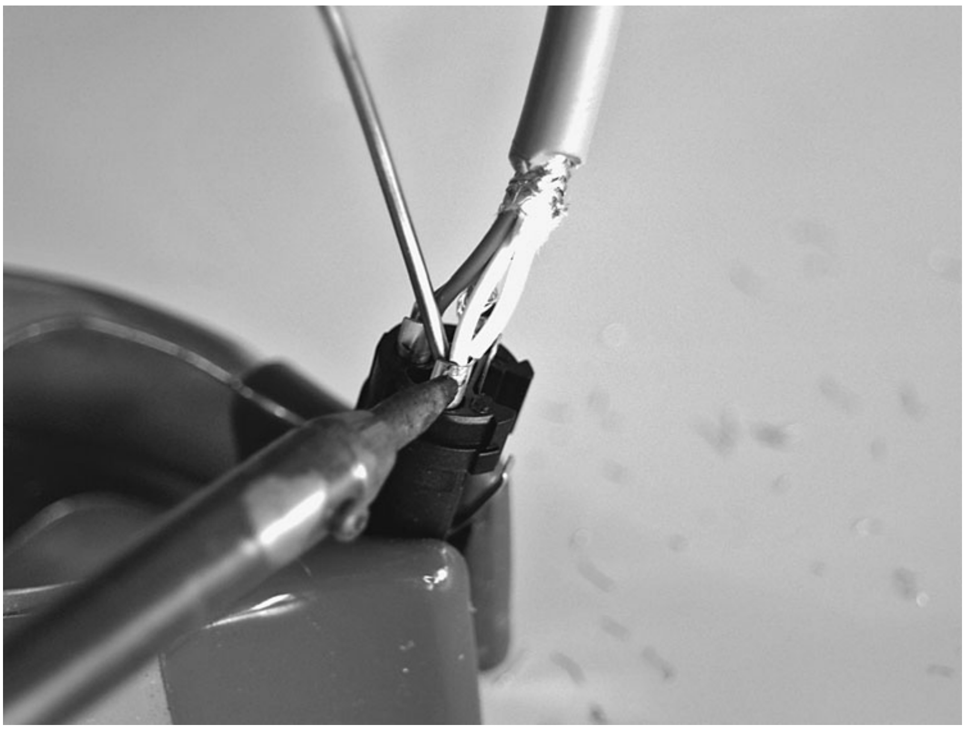


FIGURE 12.25

Soldering a wire to a plug

Neutrik cites the IEC 60 268-12 standard in recommending the following pin configuration:[3]

Pin1 --> ***X****ternal* of cable (shield/ground)
Pin2 --> ***L****ive* ("Hot" /+ polarity)
Pin3 --> ***R****eturn* ("Cold" /– polarity)

It is essential to be consistent in your wiring as a cable with reversed polarity can cause problems when recording. For example, you will experience electronic phase cancellation if two microphones are used to record a sound source in stereo and the polarity of one of the cables is reversed. A cable tester such as the Behringer CT100 costs around $30 and is very useful in identifying a variety of problems associated with cables.

Custom Wall Plate

Wall plates can be expensive to purchase but it is not hard to build an attractive patch panel. Consider the type and number of jacks that you will need on each wall and purchase an appropriate number of XLR and ¼-inch mounting jacks. I mounted the jacks to a piece of fir, but jacks could also be mounted to an appropriately sized metal plate.

Next, cut lengths of inexpensive pine or MDF to create a small box. The box will be installed in the wall and provides a convenient way to attach the front plate, as well as adding a measure of sound proofing for the plate assembly. Screws can be used to attach the plate directly to the box or a hinge can be used to make it easier to access the back of the plate (see figure 12.26). As you can see in figure 12.27, I installed a secondary piece of material that functions as a stop for the mounting plate.

FIGURE 12.26
Front of wall plate

Use a utility knife to remove Sheetrock from the wall and insert the frame. Ideally, the box will be installed next to a stud or on top of the bottom plate so that screws can be used to fasten the assembly securely in place. Assemble another box and front plate for the adjoining room and use the instructions from the previous tutorial to solder cables to connect the jacks in each room (see figure 12.27).

Pop Screen

Although professional pop screens are not big-ticket items, an attractive pop screen can be built for just a few dollars. Start by cutting a pair of old pantyhose to fit an embroidery hoop. (Embroidery hoops can be purchased at any sewing or craft store.)

FIGURE 12.27

Soldering jacks in a wall plate

Next, attach a Thomas & Betts ADR6-B2 one-hole mount to the embroidery hoop, as shown in figure 12.28. Eight-gauge copper wire is flexible and works well as a connection between the microphone stand and embroidery hoop. For a more professional look, insert the copper wire inside of ⅜-inch split-flex tubing. The copper wire is attached to the hoop with the one-hole mount shown in the previous step and a grounding clamp works well to connect the assembly to the microphone stand (see figure 12.28).

attaching mount

finished pop screen

FIGURE 12.28

Attaching a mount to an embroidery hoop

Conclusion

The projects in this chapter represent just a few of the possibilities for a home studio. Other projects such as a rolling mixer stand, keyboard workstation, microphone storage cabinet, gobos, and cable snakes can be constructed using the materials and techniques described in this chapter. As I mentioned in the introduction to this chapter, Google's free SketchUp application is a useful tool for working out project dimensions and developing virtual mockups, so I encourage you to utilize the program as a part of the design process. Finally, some community colleges offer evening woodworking classes. These classes can provide an excellent opportunity to learn about more advanced woodworking techniques and, in many cases, you will be able to use professional woodworking tools to complete a project for your home studio.

Conclusion

Like any complex topic, a mastery of recording can be a lifelong pursuit, and it is my sincere wish that this book has provided a useful introduction to the topic and will serve as a foundation for further study and exploration. A good recording can open many doors for performers, composers, songwriters, and others, and you will find that the proactive approach to listening that comes from studying the recording arts will open your ears to an entirely new musical spectrum.

Looking forward, we see that the next step is to move from the theoretical to the practical. Audio recording can be daunting owing to the many variables involving performers, instruments, audio equipment, recording technique, and artistic goals, but a helpful bit of advice comes from John Marchese's wonderful book, *The Violin Maker*. Marchese quotes a master violin maker, who says, "Part of making decisions when you're building a fiddle is going from general ideas of what would probably be good to very specific details of what would be good in *this* situation."[1] His advice is also appropriate to the topic of audio recording. To stress one of the central themes of the book, experiment with microphone placement, signal processing, and other aspects of audio production and you will develop techniques that are appropriate to *your* unique situation.

To emphasize another central theme of this book, you do not need to spend a large sum of money to get started in audio recording. The process can begin with a single microphone and a simple recording device. Not only will you develop a useful new set of skills but you will also develop a different perspective on the *sound* of music, and this awareness will have a profound impact on the way you create and listen to music.

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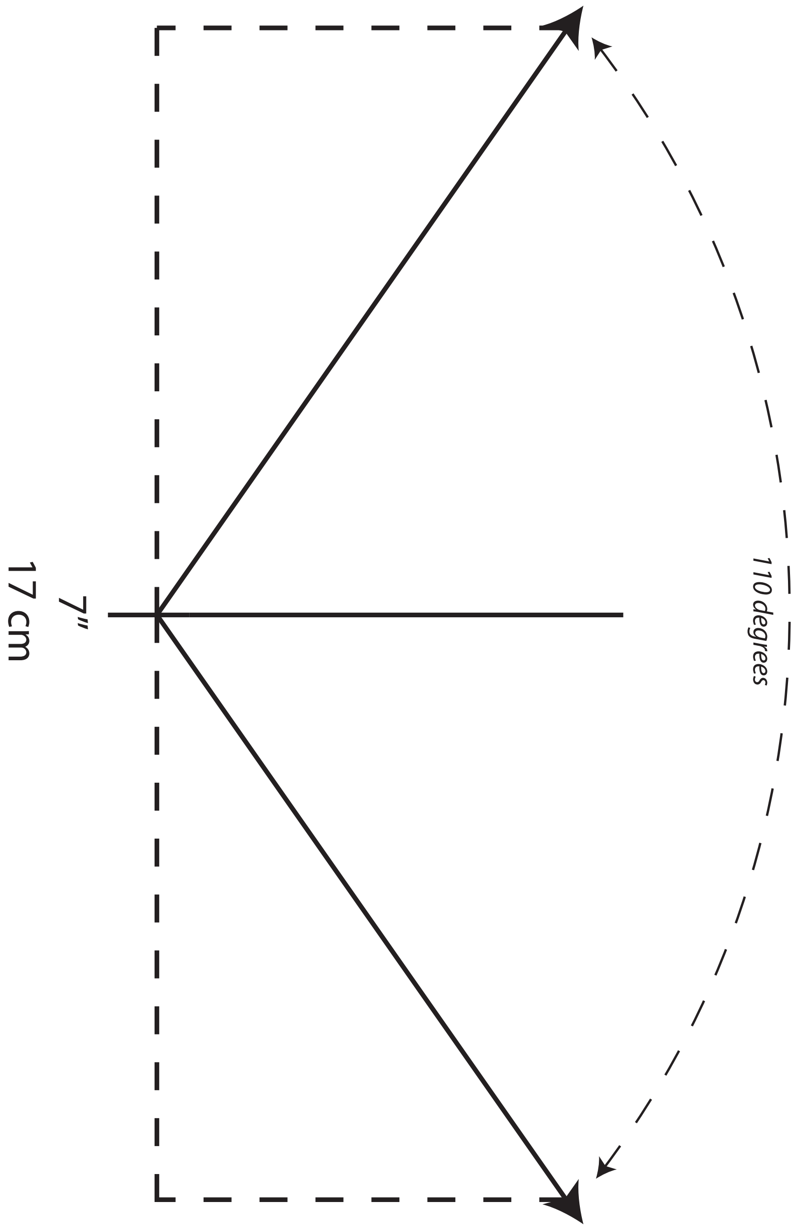
Appendix 1

ORTF Template



Appendix 2

Audio Evaluation Template

Global observations
Room (dimensionality, reverberance)

Tone color:

Dynamics:

Clarity:

Foreground:

Background:

Other observations:

Song name:

Anomalies: (e.g. tape hiss, distortion, intonation, etc.)

Sound stage
(position instruments left to right, front to back, top to bottom)

background

Level

foreground

Pan

Appendix 3

List of Online Recordings by Callout

All callouts are located in the corresponding chapter folder on the Web site. The subfolders for callouts in chapter 4 are given in parentheses following the title.

Chapter 4: Using Microphones

4.1 SM57 and SM58 (Acoustic Bass, WCU Studio Recordings, Voice, Trumpet, Saxophone, Electric Guitar)
4.2 Cascade Fat Head II (Piano, Saxophone, Trumpet, Voice,Violin Duo, Acoustic Bass)
4.3 Stereo techniques (various) (Acoustic Guitar, Violin Duo)
4.4 Ensemble and choral excerpts (Choir, Orchestra, Jazz Band)
4.5 WCU session (WCU Studio Recordings)
4.6 Vocal microphone excerpts (Voice)
4.7 Acoustic guitar excerpts (Acoustic Guitar)
4.8 Electric guitar excerpts (Electric Guitar)
4.9 Electric bass excerpts (Electric Bass)
4.10 Acoustic bass excerpts (Acoustic Bass)
4.11 Bass drum excerpts (Drums)
4.12 Drum set excerpts (Drums)
4.13 Drum excerpts: ambient microphone (Drums)
4.14 Piano excerpts: stereo techniques (Piano, WCU Studio Recordings)
4.15 Tenor saxophone excerpts (Saxophone)
4.16 Flute excerpts (Flute)
4.17 Trumpet excerpts (Trumpet)
4.18 Violin excerpts (Violin Duo)
4.19 Jazz band recording (Choir, Orchestra, Jazz Band)

Chapter 5: Podcasting

5.1 Podcasting examples: Cardioid, omni, figure 8

Chapter 8: Active Listening

8.1 "I'll Be Seeing You" from Frank Sinatra, The Capitol Years
8.2 "Beethoven String Quartet No. 127, Maestoso-Allegro," Emerson String Quartet
8.3 Tower of Power: "Spank-a-Dang"
8.4 "Blue In Green" from Kind of Blue
8.5 Pat Metheny Group: "(It's Just) Talk," from *Still Life (Talking)*
8.6 Stevie Ray Vaughan: "Pride and Joy"
8.7 Filter: "What's Next"
8.8 Mariah Carey: "Bye Bye"
8.9 Sergio Mendes and Brazil '66: "Mais Que Nada"
8.10 Duke Ellington: "All Too Soon" from Unknown Session

8.11 "Deve Ser Amor (It Must Be Love)," Antonio Carlos Jobim, Herbie Mann & Joao Gilberto
8.12 Joni Mitchell: "Free Man in Paris" from Dreamland
8.13 Duke Ellington's "Mood Indigo"
8.14 Pink Floyd: "Us and Them" from Dark Side of the Moon
8.15 David Sanborn: "As We Speak"
8.16 Marvin Gaye: "I Heard It Through the Grapevine"
8.17 Steely Dan: "Do It Again"
8.18 Lyle Mays: "Street Dreams 4"
8.19 The sound of a piano dampening mechanism
8.20 Guitar solo with and without compression
8.21 Jimi Hendrix: "Once I Had a Woman"
8.22 Compression pumping
8.23 Mariah Carey: "Migrate"
8.24 Distorted reverb tail
8.25 Digital distortion

Chapter 10: Processing Signals

10.1 Electric bass with and without compression
10.2 Snare with compression
10.3 Vocal track with compression
10.4 Main bus with and without compression
10.5 Using a limiter on bass
10.6 Main bus with and without expansion
10.7 Guitar amp with and without a noise gate
10.8 Poor application of EQ vs. a subtractive approach
10.9 Using delay to pan an instrument
10.10 Chorus effect
10.11 Phase shift effect
10.12 Flange effect
10.13 Phrase with and without pitch correction
10.14 Application of amplifier simulator to direct guitar
10.15 Bass with and without compression and EQ
10.16 Guitar amplifier simulator
10.17 Guitar with compression, chorus, and delay
10.18 Saxophone with reverb and compression
10.19 Bright and warm piano EQ
10.20 Voice with compression and EQ to tone down sibilance
10.21 Kick with compression and EQ
10.22 Snare with compressor and EQ
10.23 Processing an entire drum kit with compression and reverb
10.24 Jazz trio excerpt: reverb on main bus, mellow EQ
10.25 Classical duo: reverb on main bus

Chapter 11: Mixing and Mastering

11.1 "Homeward" pre and post master

Appendix 4

List of Online Recordings by Category

Choir, Orchestra, Jazz band

Choral: ORTF with spaced pair
Choral: mid-side
Choral: Oktava ORTF
Jazz Ensemble: EX-1 (24 tracks)
Jazz Ensemble: EX-2 (24 tracks)
Orchestra: ORTF
Orchestra: ORTF with spaced omni pair
Orchestra: spaced omni pair
Strings: ORTF with spaced omni pair

Acoustic Bass

Acoustic Bass: cardioid angle up to bridge
Acoustic Bass: cardioid below bridge
Acoustic Bass: cardioid near top
Acoustic Bass: direct
Acoustic Bass: figure 8 below bridge
Acoustic Bass: figure 8 near top
Acoustic Bass: omni angle up to bridge
Acoustic Bass: omni below bridge
Acoustic Bass: omni near top
Acoustic Bass: ribbon below bridge
Acoustic Bass: small diaphragm condenser below bridge
Acoustic Bass: small diaphragm condenser between bridge and F hole
Acoustic Bass: SM57 angle up to bridge
Acoustic Bass: SM57 below bridge

Acoustic Guitar

Acoustic Guitar: Blumlein 24 inches
Acoustic Guitar: cardioid neck and body
Acoustic Guitar: figure 8 neck and body
Acoustic Guitar: Blumlein 24 inches
Acoustic Guitar-Finger: cardioid 10 inches
Acoustic Guitar-Finger: cardioid near neck
Acoustic Guitar-Finger: figure 8, 10 inches
Acoustic Guitar-Finger: figure 8 near neck
Acoustic Guitar-Finger: omni 10 inches
Acoustic Guitar-Finger: omni near neck
Acoustic Guitar-Finger: ORTF 24 inches

Acoustic Guitar-Finger: small diaphragm condenser 2 inches from bottom
Acoustic Guitar-Finger: spaced pair cardioid
Acoustic Guitar-Finger: spaced pair omni
Acoustic Guitar-Finger: XY below with ambient omni
Acoustic Guitar-Finger: XY
Acoustic Guitar: omni-neck and body
Acoustic Guitar: ORTF 24 inches
Acoustic Guitar: Rode cardioid 10 inches from body
Acoustic Guitar: Rode figure 8: 10 inches from body
Acoustic Guitar: Rode omni 10 inches from body
Acoustic Guitar: spaced pair cardioids 12 inches
Acoustic Guitar: spaced omni and cardioid
Acoustic Guitar: XY-24 inches

Active Listening

8-1 Frank Sinatra: "I'll Be Seeing You," *The Capitol Years*
8-2 Emerson String Quartet: "Beethoven String Quartet Op. 127," *Beethoven: The Late String Quartets*
8-3 Tower of Power: "Spank-A-Dang," *Rhythm & Business*
8-4 Miles Davis: "Blue In Green," *Kind of Blue*
8-5 Pat Metheny, "(It's Just) Talk," *Still Life (Talking)*
8-6 Stevie Ray Vaughan: "Pride and Joy," *The Real Deal: Greatest Hits, Vol. 1*
8-7 Filter: "What's Next," *Anthems for the Damned*
8-8 Mariah Carey: "Bye Bye," *E=MC²*
8-9 Sergio Mendes & Brasil '66: "Mais Que Nada," *Four Sider*
8-10 Duke Ellington: "All Too Soon," *Unknown Session*
8-11 Antonio Carlos Jobim: "Deve Ser Amor," *Recorded In Rio de Janerio*
8-12 Joni Mitchell: "Free Man In Paris," *Dreamland*
8-13 Duke Ellington: "Mood Indigo," *Duke Ellington's Greatest Hits*
8-14 Pink Floyd: "Us and Them," *Dark Side Of The Moon*
8-15 David Sanborn: "As We Speak," *As We Speak*
8-16 Marvin Gaye: "I Heard It Through the Grapevine," *The Complete Collection: Marvin Gaye*
8-17 Steely Dan: "Do It Again," *Can't Buy a Thrill*
8-18 Lyle Mays: "Street Dreams 4," *Street Dreams*
8-19 Pedal noise
8-20a Guitar with no compression
8-20b Guitar with compression
8-21 Jimi Hendrix: "Once I Had a Woman," *Blues*
8-22 Compression pumping
8-23 Mariah Carey: "Migrate (feat. T-Pain)," *E=MC²*
8-24 Distorted reverb tail
8-25 Digital distortion

Drums

Drums: EV ND868-beater side
Drums: EV ND868-front
Drums: EV ND868-inside

Drums: Rode ambient-omni
Drums: small diaphragm condenser: spaced pair overhead-no snare or kick
Drums: small diaphragm condenser: XY overhead-no snare or kick
Drums: SM57 on snare
Drums: Spaced pair small diaphragm condenser: ribbon ambient, SM57, EV ND868
Drums: Spaced pair small diaphragm condenser with SM57 and EV ND868
Drums: XY small diaphragm condenser with ambient Rode omni, SM57, EV ND868
Drums: XY SDC-SM57-EV ND868

Electric Bass

Electric Bass: direct
Electric Bass: direct amp simulator
Electric Bass: direct with compression
Fretless: no compression
Fretless: with compression and limiting

Electric Guitar

Electric Guitar Amp: large diaphragm condenser: cardioid side of cone
Electric Guitar Amp: large diaphragm condenser: omni side of cone
Electric Guitar Amp: center of cone
Electric Guitar Amp: SM57 off axis
Electric Guitar Amp: SM57 side of cone
Electric Guitar: direct

Flute

Flute: cardioid
Flute: small diaphragm condenser
Flute: figure 8
Flute: omni
Flute: ribbon

Piano

Piano: bidirectional over strings: spaced
Piano: bidirectional spaced: 4 feet
Piano: Blumlein 18 INCHES
Piano: cardioid over strings
Piano: cardioid spaced 4 feet
Piano: cardioid over strings with reverb
Piano: Fat Head II: spaced pair, close
Piano: mid-side inside with reverb
Piano: mid-side 18 inches
Piano: mid-side over strings
Piano: Oktava cardioid spaced 4 feet
Piano: Oktava over strings with reverb
Piano: Oktava over strings
Piano: omni spaced pair: over strings
Piano: omni spaced pair: 4 feet

Piano: ORTF at rim
Piano: ribbon spaced pair over strings
Piano: XY at rim

Podcasting

Podcast: one large diaphragm condenser: figure 8
Podcast: two large diaphragm condensers: cardioid
Podcast: one large diaphragm condenser: omni

Signal Processing

10-1a Fretless no compression
10-1b Fretless with compression and limiting
10-2a Snare dry
10-2b Snare with compression, EQ, and reverb
10-3a Vocal no compression or EQ: pop
10-3b Vocal with compression and EQ: pop
10-4a Main no multi compressor
10-4b Main multi compressor
10-5a Electric Bass direct: no limiter
10-5b Electric Bass: with limiter
10-6a Main no compression and expansion
10-6b Main compression and expansion
10-7a Guitar Amp no noise gate
10-7b Guitar Amp with noise gate
10-8a EQ bad example: additive
10-8b EQ better example: mostly subtractive
10-9a Mono center no delay
10-9b Mono off center with delay
10-10 Direct Guitar with chorus
10-11 Direct Guitar with phase shift
10-12 Direct Guitar with flanger
10-13a Flute with no pitch correction
10-13b Flute with pitch correction
10-14a Electric Guitar no amp
10-14b Electric Guitar amp simulator-heavy
10-15a Electric Bass and combo no compression
10-15b Electric Bass and combo with compression
10-16 Direct Guitar with amp simulator
10-17a Electric Guitar no chorus, comp, delay
10-17b Electric Guitar chorus, comp, delay
10-18a Sax no reverb comp or EQ
10-18b Sax reverb comp and EQ
10-19a Piano bright with reverb
10-19b Piano warm with reverb
10-20a Voice no comp and EQ
10-20b Voice comp and EQ
10-21a Kick dry
10-21b Kick comp and EQ

10-22 Snare with compression, EQ, reverb
10-23a Drum Kit no processing
10-23b Drum Kit with processing
10-24a Full Mix no verb
10-24b Full Mix with reverb
10-25 String Duo reverb on main
11-1a "Homeward" pre master
11-1b "Homeward" master

Saxophone

Sax: cardioid 12 inches above bell
Sax: SM57 12 inches
Sax: figure 8, 12 inches
Sax: omni 12 inches
Sax: ribbon 12 inches

Trumpet

Trumpet: large diaphragm condenser 24 inches wet
Trumpet: SM57 24 inches
Trumpet: large diaphragm condenser 24 inches
Trumpet: large diaphragm condenser omni wet
Trumpet: large diaphragm condenser omni
Trumpet: ribbon 24 inches wet
Trumpet: ribbon 24 inches
Trumpet: SM57 24 inches wet

Violin Duo

Violin Duo: Blumlein
Violin Duo: XY cardioid
Violin Duo: mid-side
Violin Duo: ribbon-Blumlein
Violin Duo: spaced pair cardioid
Violin Duo: spaced pair figure 8
Violin Duo: spaced pair omni
Violin Duo: spaced pair figure 8 wet

Voice

Vocal: AT3035
Vocal: SM58
Vocal: cardioid off axis no pop screen
Vocal: Cascade center
Vocal: Cascade mid-body angled up
Vocal: no compression
Vocal: Rode cardioid
Vocal: Rode figure 8
Vocal: Rode omni
Vocal: SM57

WCU Studio Recordings

AEA-R84
AKG 414: cardioid high, omni low
AKG 414: close pair, Focusrite
AKG 414: close pair, Mbox
AKG 414: close pair, PreSonus
AKG 414: Focusrite
AKG 414: PreSonus
AKG 414: spaced pair, Mbox
AKG 414: with C12 close pair, PreSonus
AKG 414: close pair with C12
AKG 451: Focusrite
AKG 451: Mbox
AKG 451: PreSonus
AT STEREO MIC: close pair, PreSonus
AT STEREO MIC: close
AT815-close: MBox
C12VR: omni, Focusrite
C12VR: omni, Presonus
KM 184: Focusrite, ORTF
KM 184: Mbox, ORTF
KM 184: PreSonus, ORTF
NEUMANN U87: close pair, Focusrite
NEUMANN U87: close pair, Mbox
NEUMANN U87: close pair, PreSonus
NEUMANN U87: Focusrite
NEUMANN U87: PreSonus
NEUMANN U87: close pair with C12
NEUMANN UA87: Mbox
RODE NT4: Focusrite
RODE NT4: PreSonus
RODE NT5: Mbox
ROYER R122: Focusrite
ROYER R122: PreSonus
ROYER R122: spaced pair, Mbox
ROYER R122: with C12-Focusrite
ROYER R122: with C12-Presonus
SM57: close-PreSonus
SM57: close-Focusrite
SM57: close-Mbox

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