Chapter 1

Discovering What You Need

In This Chapter

- Understanding the components of a computer-based studio
- Discovering how each component contributes to the final sound

Whether you use a Windows or Macintosh computer for your home recording studio, your system of choice employs much of the same basic technology. In fact, your simple computer-based studio consists of the same basic components as a typical million-dollar professional studio complex — they're just in a different physical format. For example, instead of the huge mixing board that you see in a commercial studio, you're going to be working with a piece of software in your computer. It might not look as imposing, but it performs the same functions (and then some).

In this chapter, you discover the purpose of each individual component of a computer-based recording studio — and you also discover how each of these components relates to the quality of sound that you ultimately get from your studio. This knowledge will definitely help you when it comes to spending the right amount of money on the right stuff. (See Chapters 3, 4, and 8 for more on purchasing gear.)

I use the term component pretty broadly in this chapter and include everything you may use in your studio from preamps and microphones that you need to capture the sound to the software you use for editing to the blank CDs you use to store your musical data on.

Looking at the Larger Picture

At first glance, trying to figure out what you need to record your music with a computer can be confusing. Taking a quick look through this chapter will probably reinforce this perspective — at least initially. Before you get a brain cramp trying to figure out the more arcane jargon, here's a short list of what a typical computer-based home studio consists of:

- **Audio interface:** The audio interface is a piece of hardware that allows you to get sound from the outside world into your computer. Most audio interfaces contain everything you need to accomplish this task, including preamps, direct boxes, AD and DA converters, and a sound card. Confused by all these components? You won't be for long — as you read this chapter, all these items will start making sense to you. Many types of interfaces are available, each with different features. At least one option will surely meet your particular needs.
- **Computer:** This is an obvious one, but it's an item that confuses a lot of people. Of course you know you need a computer to make music with a computer, but the question that always comes up is, "What kind of computer?" I describe the basics of an audio-recording computer later in this chapter and even dedicate an entire chapter to this seemingly innocent question (see Chapter 3).
- **Input device:** Input devices include instruments, mics, and any other device that lets you input sound into your computer.
- **Mastering media:** The mastering media is where you put your finished music. This can be CDs, or it can be in the form of computer files such as MP3, WAV, AIFF, and others.
- **Monitors:** Monitors consist of speakers or headphones. These are important because you need to hear what you're recording or mixing.
- **Software:** Music software can vary considerably. You can find simple programs that let you assemble pieces of pre-recorded music (called loops) such as Apple's GarageBand or basic recording programs such as Guitar Tracks Pro by Cakewalk. You can also find more sophisticated programs such as MOTU's Digital Performer or Steinberg's Cubase — ones that allow you to record and mix hundreds of audio and MIDI (Musical Instrument Digital Interface) tracks, software synthesizers, and limitless effects.

  Software can also exist in the form of synthesizers, samplers, and digital signal processors (DSPs) such as effect plug-ins.

REMEMBER

As you're probably aware, a recording studio can have lots of gear — from a locker full of microphones and roomful of instruments to a pile of electronic gear such as preamps, compressors, mixers, and speakers. All this equipment is seductive, and you could spend all your time fussing with gear and not get any recording done. (In fact, this happens a lot.) Try not to focus too much on the equipment. Instead, put your energy into making music.

TIP

As you get more and more involved in recording, you'll find that you can add almost any of the individual components that I describe in this chapter to your existing system to expand and enhance what you can do. For example, even if your audio interface comes with a preamp (or two, or four, or more), you might want to get hold of a dedicated external preamp to improve the sound or create a specific effect from your microphones.

Interpreting Input Devices

All your expensive recording gear is useless if you have nothing to plug in to it. This is where the input device comes into play. An *input device* is, simply, any instrument, microphone, or sound module that produces or delivers a sound to the recorder.

Instruments

Your electric guitar, bass, synthesizer, and drum machines are typical of the instruments that plug in to the interface and represent most of the input devices that you use in your studio. The synthesizer and drum machine can plug directly into the Line In inputs of your interface, whereas your electric guitar and bass need a direct box (or its equivalent) to plug in to first. (Most audio interfaces allow you to plug directly into one of the preamps, so you don't need a separate direct box.)

A *direct box* is an intermediary device that allows you to plug your guitar directly into the mixer without going through your amp first. (For more on direct boxes, see the upcoming section "Deciphering direct boxes.") Check out Figure 1-1 for an example of an instrument-input device.

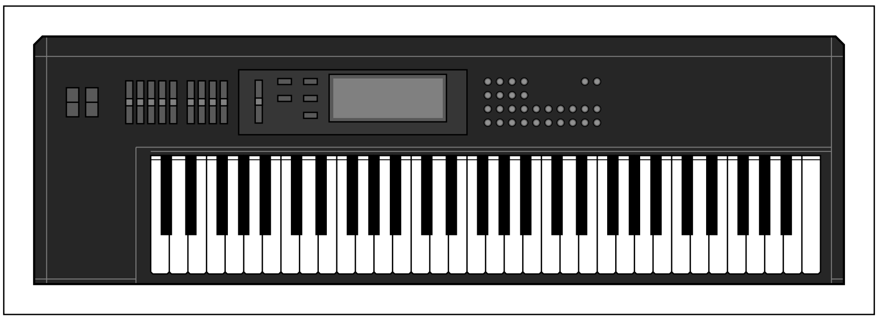


Figure 1-1: An instrument-input device, which you can plug right into your audio interface.

Microphone

A microphone (abbreviated *mic*) enables you to record the sound of a voice or an acoustic instrument — sound sources that, last time I checked, couldn't be plugged directly into the interface. A mic (shown in Figure 1-2) converts sound waves into electrical energy that can be amplified by the preamp and understood by the interface. As you find out in Chapter 3, a lot of different types of mics are available, and choosing the best one for a particular application is like choosing the color of paint to put on a canvas.

Figure 1-2: A mic. Use a mic when your instrument can't plug directly into your audio interface.

Sound modules and soft-synths

Sound modules are special kinds of synthesizers and/or drum machines. What makes a sound module different from a regular synthesizer or drum machine is that these contain no triggers or keys that you can play. Instead, sound modules are controlled externally by another synthesizer's keyboard or by a Musical Instrument Digital Interface (MIDI) controller (a specialized box designed to control MIDI instruments). Sound modules have MIDI ports (MIDI jacks) to enable you to connect them to other equipment.

TIP

Often sound modules are *rack-mountable,* meaning they have screw holes and mounting ears so that you can put them into an audio component rack. Some controllers, however, are not rack-mountable. Figure 1-3, for example, shows a drum module that rests on a stand or tabletop.

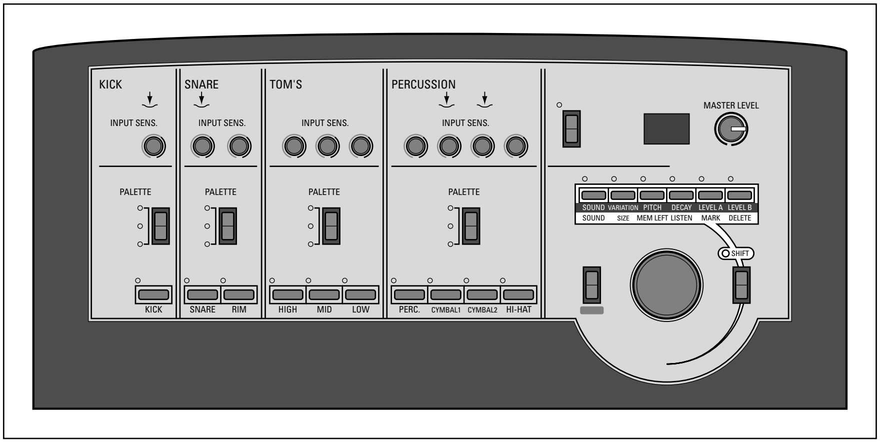


Figure 1-3: The sound module can be plugged right into the mixer but has to be played by another source.

Soft-synths are software versions of sound modules, housed in your computer as programs. These software programs have no need of external MIDI connections because they're *virtual* sound modules — they live on your computer's hard drive. You just insert them into your recording program as plug-ins.

Examining the Audio Interface

In order to record into a computer, you need a device called an *audio interface.* The audio interface is a piece of hardware that acts as an intermediary between the analog world of your mics and instruments and the digital world of your computer. Traditionally (if there can be traditions in such a new technology), you needed the following pieces of gear:

- **Sound card:** This card — also called a *PCI card* because it fits in the Peripheral Component Interface (PCI) slot in your computer — allows your computer to read and understand the digital information coming from and going to the converters. Without a sound card, your computer doesn't know what to do with the musical data that it receives (or that your hard drive stores).
- **Analog-to-digital (AD) and digital-to-analog (DA) converters:** These converters allow you to get the sound from your instrument, direct box, or preamp to the sound card (the AD converter's job) and from your computer back out to your monitors (the DA converter's job).

REMEMBER

Nowadays, both the sound card and converters are usually part of the audio interface, so you don't have to buy them separately.

An audio interface also generally contains everything else you need to get your instruments and mic signals into your computer properly. These components include:

- **Direct box:** A direct box (technically called a Direct Induction Box or DI box for short) lets you plug your guitar directly into your recording device (in this case your computer) without having to go through your amp first.
- **Microphone preamp:** This is a requirement if you want to plug your mic into your recording device. The preamp amplifies the signal coming from your mic so that it can be recorded.

TIP

Audio interfaces have different ways of handling both these components. Some contain preamps that can act as both a microphone preamp and a direct box, but others also have instrument inputs that function like a direct box and allow you to plug guitars directly into them. Most audio interfaces have at least two and as many as eight inputs with preamps. (Chapter 3 shares more on instrument inputs and preamps.)

Audio interfaces come in many varieties — varieties that use one of the three following ways to connect to your computer:

- **Through a PCI card connected to your computer's PCI slot:** This method is the old standard for getting audio in and out of a computer. PCI-based interfaces come in several varieties, which include the following:
  - Separate sound cards with no analog inputs and outputs.
  - Analog inputs and outputs within the card.
  - Analog inputs and outputs housed in a separate box (called a *break-out box*).

  TIP

  If you want to go the PCI route, make sure that your computer has PCI slots that are compatible with the PCI interface that you're considering. (Not all are; the Digidesign 001, for example, doesn't work in Mac G5 computers.)
- **Through an interface connected to the USB port:** This method is handy because most computers have at least one USB port. The only problem with USB for recording audio is the relatively slow transfer speed. USB 1.1 generally limits you to two inputs (although you can find some interfaces with as many as six inputs) and two outputs. It also introduces more *latency* — the delay from the audio going in and coming back out of your computer — than the PCI or Firewire (see the next bullet) options. (Dealing with latency definitely adds some steps to the recording process; I spell out the steps for overcoming this problem in Chapter 10.)

USB 2.0–compatible interfaces are just beginning to hit the market. USB 2.0 offers a much faster transfer rate than USB 1.1, so the limited input and latency issues won't be a problem in the not-too-distant future (assuming you have both a computer and USB audio interface that function with USB 2.0).

- **Through your FireWire port:** FireWire is preferable to USB because the transfer speed is fast enough to keep latency down to a minimum. FireWire ports are inexpensive and available on laptop computers as well as desktop ones, which makes Firewire interfaces more versatile than PCI-based systems.

TIP

Because you have so many audio interface options to choose from, I detail what to look for in Chapter 3.

Singling out a sound card

A sound card is necessary for your computer to be able to record or playback digital audio data. All computers come with a basic sound card, but for recording music you most likely need to get a better one. Keep in mind, though, that buying a soundcard separately isn't your best option because you'll still need to get the AD and DA converters and other components that are included in an audio interface. And all audio interfaces come with a (usually pretty good) soundcard so a separate sound card isn't necessary.

Examining AD and DA converters

When you play your instrument or sing into a mic, the *signal* that you're producing is an analog one. It consists of electrical impulses representing sound waves. In order for you to record, store, or playback these impulses in your computer, you need to convert these impulses into and out of digital bits. (You know, 1s and 0s.) You do this with AD and DA converters.

REMEMBER

The quality of the sound of your recordings is hugely influenced by the quality of your sound card and converters. Because this is such an important part of the recording puzzle, I explain the intricacies of digital audio conversion, recording, and playback in Chapter 3. (I make it easy to understand, I promise.) Before you go out and buy an audio interface, I highly recommend that you read Chapter 3.

Deciphering direct boxes

A direct box (or DI box, short for *D*irect *I*nduction) is traditionally used to connect your guitar or bass directly into the mixer without having to run it through your amp first. A direct box's purpose is twofold: to change the guitar's impedance level (a guitar has a high impedance, and a mixer has a low impedance) so it matches your mixer to create the best sound possible, and to change the nature of the cord connection from unbalanced (quarter-inch) to balanced (XLR) so you can use a long cord without creating noise. (For more on cord types and balanced signals versus unbalanced signals, go to Chapter 5.)

Because you're unlikely to need a long run of cords from your guitar to your mixer (the main reason for going from an unbalanced to a balanced connection), your main purpose in using a direct box is to act as an impedance transformer (it changes your guitar's signal from high to low impedance). Without a direct box changing your impedance levels, your guitar signal might sound thin or have excess noise.

Depending on how many mics and guitars you want to plug in to your audio interface (and depending on which interface you have), you might not need to buy a separate direct box. Most audio interfaces have inputs that can handle any and all impedance-transforming chores associated with the signal from your guitar or bass. These are usually the inputs with the preamps already built in. (See the next section.)

Perusing the preamp

Microphones produce a lower signal level than do line-level devices (synthesizers, for example); thus they need to have their signal level increased. For this purpose, you need a *preamp,* a device that boosts a mic's output. Preamps can be internal or external, meaning they could reside within your audio interface or exist as a separate unit that you plug in between your mic and audio interface.

The preamp is one of the most crucial elements of a recording system. It can affect your instrument's sound significantly. Most professional recording studios have a variety of preamps to choose from, and engineers use a particular preamp based on the type of sound they're trying to capture.

The three basic types of preamps available are solid-state, tube, and hybrid.

Solid-state

Solid-state preamps use transistors to boost the level of the mic or instrument. Top-quality (expensive) solid-state preamps are generally designed to produce a sound that's clear and accurate (GML and Crane Song brands, for instance). Solid-state preamps can also be designed to add a pleasing distortion to the music (Neve, API, and Neve-clone preamps, for example). Many recording professionals prefer the clear and accurate sound of a solid-state preamp for acoustic or classical music or any situation when capturing a very natural sound is important. The preamps in your audio interface are solid-state — though certainly not as high a quality as many of the more expensive external preamps — and are usually designed to more on the “clean sound” side of the spectrum, rather than the “pleasingly distorted” side.

Tube

Since the beginning of the digital recording revolution, professionals have been complaining about the harshness of digital recording. As a result, many digital-recording pros prefer classic *tube* preamps because they can add warmth to the recording. This warmth is actually a distortion, albeit a pleasing one. All-tube preamps are generally very expensive, but they’re highly sought after among digital recording aficionados because of their sound. Tube preamps work well with music when you want to add color to the sound (for example, adding some distortion to your sound source or enhancing certain pleasing tones in your instrument). No wonder they show up a lot in rock and blues — and they’re great for recording drums. You can also find tube preamps that are clean and open, such as those made by Manley Labs.

Hybrid

A *hybrid* preamp contains both solid-state and tube components. Most of the inexpensive tube preamps that you find in the marketplace are actually hybrids. (These are also called *starved-plate* designs, because the tubes don’t run the same level of voltage as expensive tube designs.) These types of preamps are usually designed to add the classic tube warmth to your instrument’s sound. How much the sound is colored by the tubes — and how pleasing that colored sound is to the listener’s ears — depends on the quality of the preamp. Most hybrid preamps allow you to dial in the amount of *character* (pleasing distortion) that you want. You won’t find a hybrid preamp that sounds as good as a great (or even pretty decent) solid-state or tube preamp, but you might find one that works well enough for your needs.

Your audio interface comes with a limited number of solid-state preamps (usually two to four, but sometimes as many as eight). If you want to plug in more mics than the number of preamps you have or if you want to be able to produce different sounds from your preamps, you need to buy one or more external preamps, such as the one shown in Figure 1-4.

Figure 1-4: An external preamp.

Clueing In to the Computer

No matter which platform of computer you choose, Mac or PC, the stuff you find inside your computer plays a major role in determining how smoothly (or how less-than-smoothly) your computer recording system runs. (Chapter 3 details the best computer setups for audio.)

To set up a computer to record audio properly, you need several things:

- **A computer** (Preferably with a speedy processor.)
- **Bunches (BIG bunches) of memory** (The words *too much* don't apply.)
- **Dual hard drives** (One just won't cut it.)
- **An audio interface** (See the "Examining the Audio Interface" section earlier in this chapter and also see Chapter 4.)
- **The software** (See the "Signing On to Software" section later in this chapter and also see Chapter 3.)

The following list clues you in on the various pieces of hardware that you find in your computer:

- **The CPU:** The CPU (processor) is the heart of your computer studio. The speed of your CPU ultimately dictates just how well any program runs on it. As a general rule, for audio, get the fastest processor that you can afford. For most audio software, you need *at least* a Pentium III for the PC or a G3 for Mac. If you can afford it, get a *dedicated* computer — one that you have specifically set aside for recording audio — because running other types of applications (such as home finance, word processors, or video games) can cause problems with your audio applications and reduce the stability of your system.
- **Memory:** Computer-based audio programs and all their associated plug-ins are RAM (random access memory) hogs. My advice: Get a lot of RAM. Okay, that's not very specific, but how much you really need depends on your recording style. If you do a lot of audio tracks and want reverb or some effect on each track, you need more RAM (and a faster processor).

  Many recording software programs recommend a minimum of 384MB of RAM, but you should really get a lot more. And I mean a *lot* more. At least a gigabyte, but you can never have too much. Also, don't skimp on the quality of the RAM you use. Cheap RAM is worse than no RAM at all, so I recommend that you buy name brand RAM.

  REMEMBER

  Regardless of the platform you choose (PC or Mac), keep in mind that you can never have too fast a processor or too much memory.

- **Hard drives:** To record audio, be sure you get the right type of hard drives. Notice how I said hard *drives* (plural). Yep, you should get more than one if you want to record more than a few tracks of audio. You want one hard drive to hold all the software and the operating system — and *another* drive just for the audio data. Having two greatly increases the likelihood that your system remains stable and doesn't crash on you, especially if you try to run 16 or more tracks.

  As for the size of the hard drive, bigger is better, at least in the audio drive where you store your music. For the core system drive, you can get by with a 10GB (gigabyte) drive; for the audio, 20GB is still pretty conservative because audio data can take up a ton of space. For example, a 5-minute song that has 16 audio tracks recorded at a 24-bit resolution and 44.1-kHz bandwidth would take up about 600MB of hard drive space (that's about 7.5MB per track minute).

  Choose your hard drives wisely. For the software hard drive, you can get away with a stock drive (usually the one that comes with your computer). But for the audio, you need a drive that can handle the demands of transferring audio data at high speed. The main things you want to look for are

  - **Spindle speed:** Also called *rotational speed,* this is the rate at which the hard drive spins. For the most part, a 7,200 rpm (rotations per minute) drive works well for recording and playing back audio.
  - **Seek time:** This is the amount of time the drive takes to find the data stored on it. You want an average seek time under 10 ms (milliseconds).
  - **Buffer size:** Often called *cache buffers,* these memory units store data as it's being transferred. You want a buffer size of at least 2MB.

The track count that your system can handle is directly related to the speed of your hard drive — the faster the drive, the more tracks you can record and play back at once. (Of course, the type of drive you get determines how large a role your processor plays.) My current choice for a drive is a Maxtor 7,200 rpm ATA IDE drive with an 8.5 ms seek time and 2MB cache buffer. A 100GB drive currently costs around $100.

Signing On to Software

One thing I guarantee is that you won't have a hard time finding a piece of software that meets your musical needs. Heck, I'll even go so far as to guarantee that choosing the best software among the plethora of options won't be easy. I'm sorry if this bursts your bubble, but someone had to say it. Yep, the options for audio are endless. (Well, almost endless.) Even though I explore audio software in detail in Chapters 7 and 8, here's a quick rundown on the basic components of audio recording software:

- **The mixer:** The mixer lets you adjust the level (volume) of your instruments, route your signals where you want them, and add effects or other digital signal processing to your tracks. The mixer is one of the most important pieces of gear in a recording studio — even a computer-based one. All audio recording software contains a mixer.
- **The arranger:** The arranger is where you can organize your musical ideas. All audio recording software has some sort of arrange function.
- **The editor:** One of the best things about computer-based recording is that you can generally do sophisticated editing of the audio data. The editing capabilities of the many software programs vary considerably. Some — such as Pro Tools — have very powerful audio editing while others — can you say "Logic Pro" — have very powerful MIDI editing.
- **MIDI sequencer:** MIDI (short for Musical Instrument Digital Interface — a communication protocol for musical instruments) is often part of audio recording software and — like editing capabilities — the MIDI capabilities in the different programs vary. Pro Tools, for example, is known for having rudimentary MIDI capabilities, whereas SONAR and Logic Pro have powerful MIDI features.
- **Digital signal processor:** Digital signal processing (DSP) is anything you do to the sound of your audio data other than adjust the volume. This includes equalization, dynamics processing, and effects processing (and many other kinds of tweaks).

Meeting the mixer

The mixer is the heart of any recording system. Although the mixer might seem daunting with all its knobs, buttons, sliders, and jacks — take a look at Figure 1-5 to see what I mean — it's really one of the most interesting and versatile pieces of equipment in your studio. With the mixer, you can control the volume level of the incoming signal, adjust the tonal quality of an instrument, blend the signals of two or more instruments together, and a host of other things. And don't worry; as you read through this book, you get the hang of all those knobs in no time.

For the computer-based home recordist (that's you), the mixer is incorporated into your computer software. (Of course, you can always use an external hardware mixer if you want, but it's not necessary.)

The mixer in all the audio recording software programs does the job well enough that you don't need an external mixer, although some people prefer having physical faders and knobs to mess with. If you're a knob-turner and like to physically touch the instrument you're playing (or, for that matter, the gadget you're tweaking), I heartily recommend that you get a dedicated computer-control surface, such as the Mackie Control (shown in Figure 1-6). A *computer-control surface* is a unit that lets you get your knobs and faders while still using the internal mixer in your recording software. This can be an advantage because it eliminates the need for lots of analog-to-digital conversions (ADCs) and digital-to-analog conversions (DACs). (And that's not counting the actual converters, which can cost a lot of money.) On the other hand, if you prefer clicking a mouse or typing on a keyboard (the kind with letters on the keys), choose a software version.

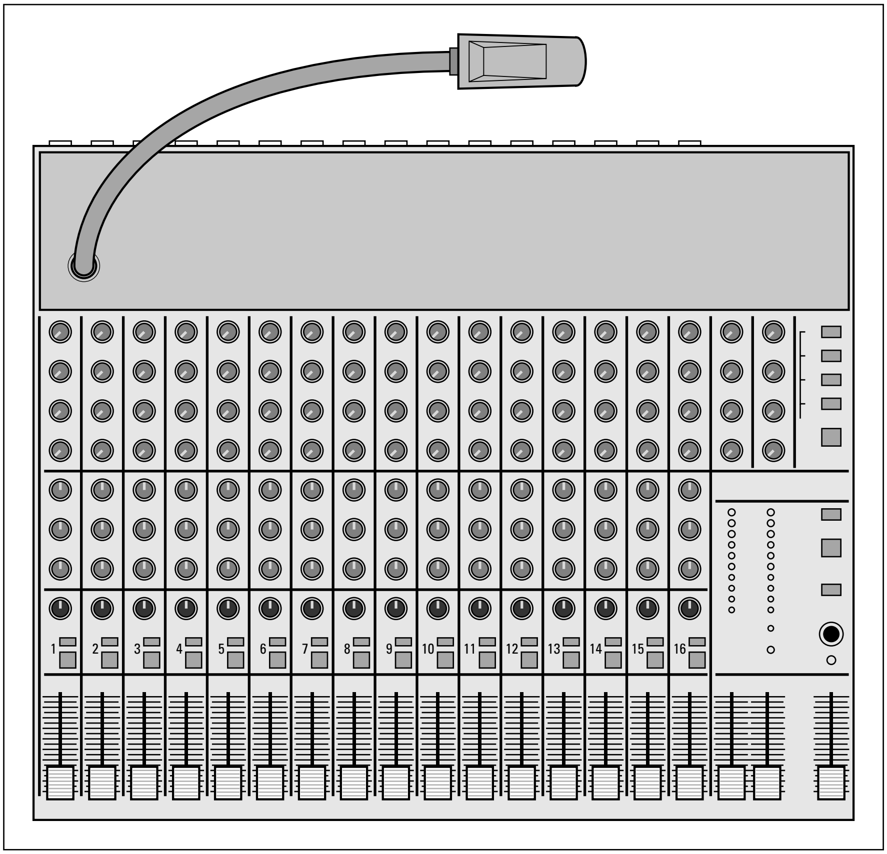


Figure 1-5: The mixer is the heart of your home studio system.

Figure 1-6: A computer-control surface offers you real knobs and faders and still uses the mixer that's part of your software.

Accessing the arranger

Audio software makes arranging your musical ideas very easy. Usually with just a click-drag of your mouse you can move sections of data around. Figure 1-7 shows a screen shot of a typical arrangement window. From this window you can perform any number of functions, including trying alternate arrangements for your song; taking the best parts of your performances and placing them exactly where you want them; and countless others.

Exploring the editor

Being able to make miniscule edits to your recorded performances is one of the reasons many people want to record on a computer. Between the usually large video monitor that most people have and the functions of most audio recording programs, editing is a breeze. Check out Figure 1-8 for a look at the editing window for a popular brand of software. From this window you can improve the quality of your music dramatically with the help of a few editing tricks — getting rid of unwanted noise (those darned chair squeaks, for example) or fixing the occasional bad note in an otherwise inspiring performance, to name a few. Of course the downside is editing too much and sucking the life out of a performance, but you wouldn't do that (would you?). Chapter 6 talks about editing capabilities in some of the more popular brands of software, and Chapter 11 explores how to use these capabilities with your music.

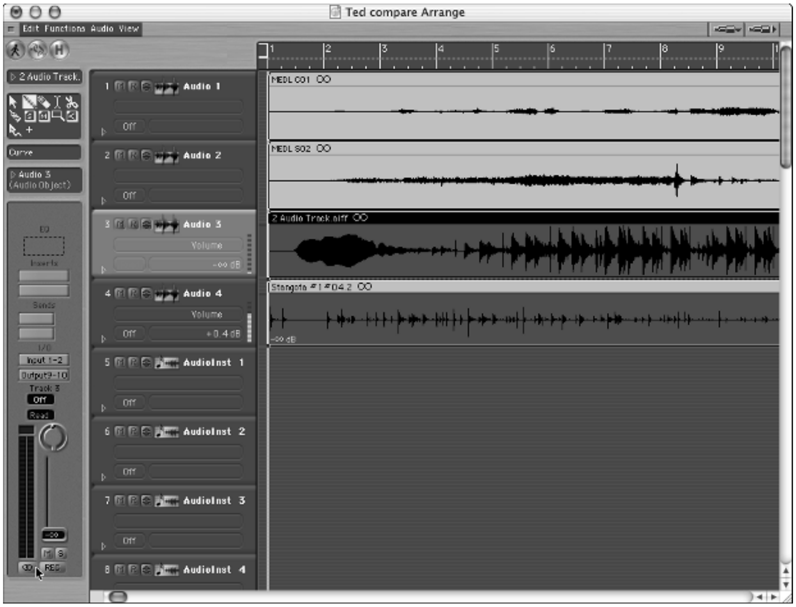


Figure 1-7: An arrangement window in audio software programs lets you perform many useful functions.

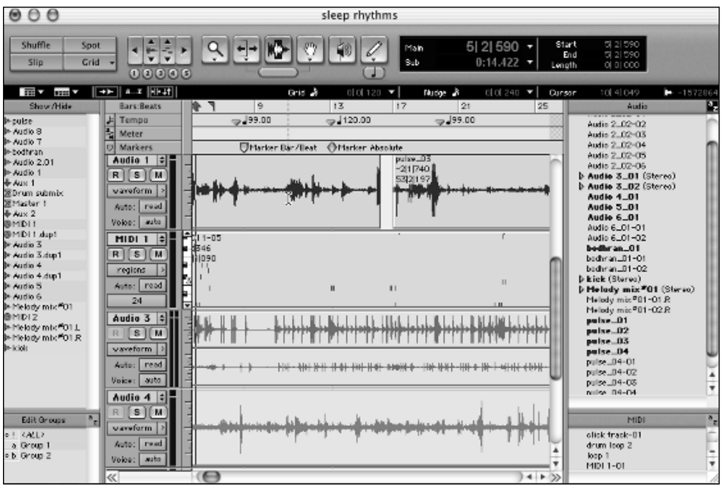


Figure 1-8: Editing in audio software programs makes it easy to fix problems with your performances.

Managing the MIDI sequencer

A MIDI sequencer is where you can record MIDI performance data into your computer for playback and editing. MIDI sequencing allows you to record your performance and choose the sound that accompanies it later on. The MIDI data that you record is simply performance information, such as when a note is played, the velocity (volume) of the note, and when the note was released, to name a few.

TECHNICAL STUFF

MIDI (Music Instrument Digital Interface) is a protocol that musical instrument manufacturers (in a rare moment of cooperation) developed to allow one digital instrument to communicate with another. MIDI uses binary digital data, in the form of 1s and 0s, to tell an instrument to play or release a note, to change sounds, and a host of other messages. Chapter 4 explains more about the other gear you need to do MIDI in your studio, and Chapter 11 lays out how to use MIDI for your musical ideas.

Most audio recording software contains some sort of MIDI sequencing capabilities. Some are fairly simple and let you record, playback, and edit the data, but others offer much more advanced features that let you work with the data in intricate ways. Check out Figure 1-9 to see a couple MIDI windows in a MIDI powerhouse program.

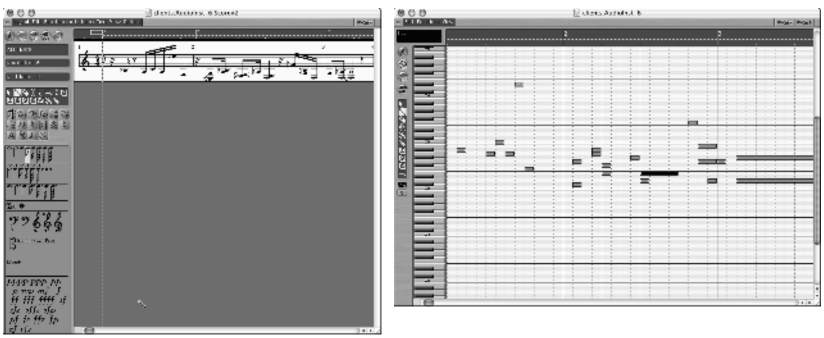


Figure 1-9: MIDI is easy to use in most audio-recording software programs.

Digging into digital signal processors (DSPs)

Part of the recording process involves making adjustments to a sound either before or after it's been recorded. This is the job of the *signal processor.* Signal processors come in three varieties — equalizers, dynamic processors, and effects processors. They can be incorporated into the system or work as separate, standalone units. For most home recordists, the signal processors of choice are integrated into the software as plug-ins, although you can also use external processors by sending the audio out of your computer and back in again.

Equalization (EQ)

Equalizers enable you to adjust the frequencies of a sound in a variety of ways. In effect, you tell the frequencies to

- **Go away:** You can get rid of unwanted noise or an annoying ringing by reducing select frequencies.
- **Come hither:** Add *life* or *presence* to an instrument by bringing the best characteristics of that instrument forward.
- **Scoot over:** You can make room within the frequency spectrum for each of the instruments in your mix by selectively boosting or cutting certain frequencies.

You can find out more about EQ (and discover some great EQ tips and tricks) in Chapter 16. The three main types of EQ are graphic, shelf, and parametric. Here's the rundown:

Graphic EQ

Use *graphic equalizers* to choose a specific frequency to increase or decrease by a certain amount, generally up to 6 or 12 decibels (dB). Doing so enables you to eliminate an offending frequency from the signal or make other adjustments to the tonal quality of the source signal. The graphic EQ will have a certain number of frequency bands that you can adjust. You're limited to only those frequency bands that your EQ has available. Figure 1-10 shows a typical graphic EQ.

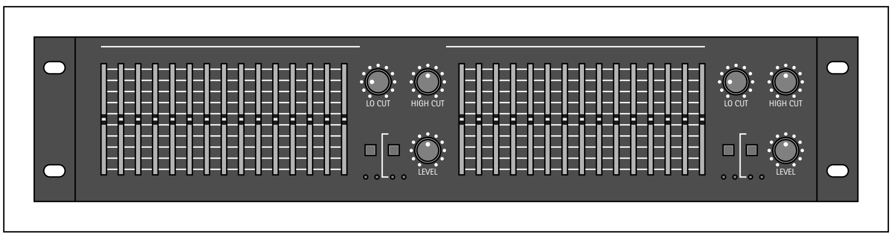


Figure 1-10: A graphic equalizer.

Shelf EQ

A *shelf equalizer* affects a range of frequencies above or below the target frequency. Shelf EQs are generally used to *roll off* the top or bottom end of the frequency spectrum. For example, you can set a shelf EQ to roll off the frequencies below 250 Hz (hertz) to reduce the amount of *rumble* (low-frequency noise) in a recording. You can see how this looks in Figure 1-11. Notice how the shelf EQ gradually reduces the amount of energy (sound) below the set point and then levels off: hence, the *shelf* in its name.

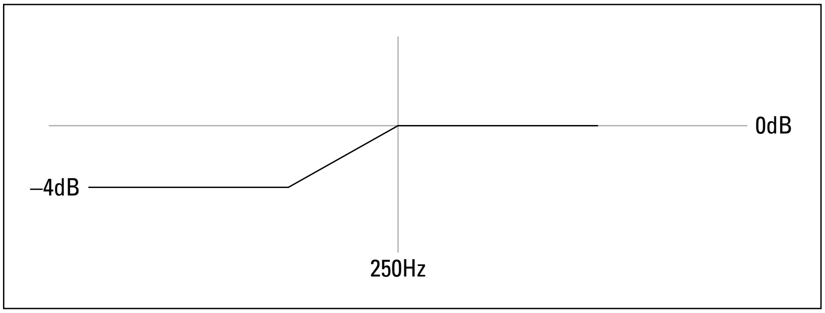


Figure 1-11: A shelf equalizer works this way.

Parametric EQ

The *parametric equalizer* enables you to choose the specific frequency that you want to affect, as well as the range of frequencies to put around that frequency. With a parametric EQ, you dial in the frequency that you want to change, and then you set the range (referred to as the *Q*) — that is, the number of octaves that the EQ will affect. Check out Figure 1-12: The two diagrams show how the Q relates to the range of frequencies affected. (The higher the Q setting, the narrower the band of frequencies affected.)

The beauty of a parametric EQ is that you can take a small band (range) of frequencies and add or cut them. Doing this allows you to get the various instruments in a mix to fit in with one another (called *carving out frequencies*). For example, a lot of times the bass guitar gets lost behind the booming frequencies of the kick drum (200–500 Hz). By setting the Q to cover this frequency range and setting the level at about –6dB (reducing by 6 decibels), you effectively make room for the bass guitar to be heard in the mix.

You'll find that the parametric EQ is one of your most useful tools when mixing all your individual tracks into a stereo pair (part of the mixing process covered in Chapter 14). I describe this tool in detail in Chapter 16.

Dynamic processors

Dynamic processors are devices that regulate the amount of sound (energy) that passes through them. This amount is defined as the *dynamic range* — the range between the softest sound and the loudest sound. Dynamic processors come in three varieties: *compressors/limiters, gates,* and *expanders.* I explain each variety in the following sections.

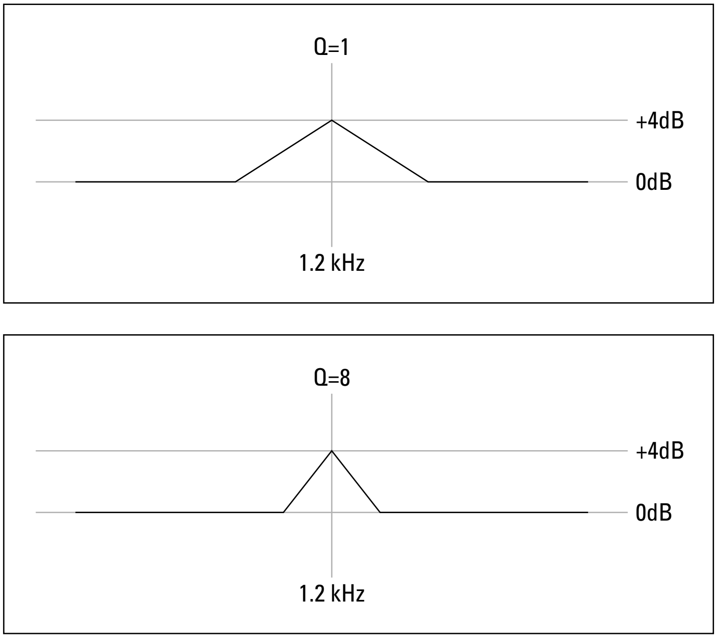


Figure 1-12: A parametric equalizer in a digital system. Top: Using a small Q. Bottom: Using a large Q.

Dynamic processors are useful in a variety of ways. Use them to

- Control the signal going into the mixer and recorder.
- Tame the levels and correct the effects of an erratic musical performance when mixing.
- Optimize the levels of the finished stereo tracks when mastering — the final step in making your music. (Chapter 14 covers mastering in greater detail.)

Dynamic processors are some of the most useful tools that you have in your home studio. See Chapter 18 for more on dynamic processors.

Compressors/limiters

The compressor's job is to compress the dynamic range of the sound being affected. The purpose of the compressor is to eliminate *transients* (unusually loud notes) that can create *clipping* (digital distortion). The compressor limits not only how loud a note can be, but it also reduces the difference between the loudest and softest note (compressing the dynamic range).

Compressors are used extensively on vocals; the device keeps transients at bay by gently reducing the highest level that goes through it. Compressors are also used in mastering to raise the overall volume of a song without creating distortion. The device does this by reducing the overall dynamic range, and as a result, a compressor effectively raises the volume of the softer notes without allowing the louder notes to be too loud and distorting.

A *limiter* works much like the compressor, except that it severely limits the highest level of a sound source. The limiter is basically a compressor on steroids: It gives you beefed-up control over volume. Any signal above a certain level, called the *threshold,* gets chopped off rather than compressed (as with the compressor). A limiter is a good choice in extreme situations, such as when you want a really in-your-face snare drum sound. In this instance, the limiter essentially eliminates any dynamic variation from the drummer's snare drum hits and makes them all as loud as possible.

Gates

A *gate* is basically the opposite of the limiter. Rather than limiting how loud a note can get, the gate limits how *soft* a note can get. The gate filters out any sound below a certain setting (the threshold) while allowing any note above that threshold to be heard.

TIP

Recordists often use gates on drums to keep unwanted sounds from the cymbals from bleeding through to the tom-tom or snare drum mics, or on guitars by refusing to allow the noise generated by guitar effects to be heard when the instrument isn't playing.

Expander

The *expander* is basically the opposite of a compressor — instead of *attenuating* (reducing the volume of) the loudest notes in a performance, an expander attenuates the softest notes. For example, if you have a singer whose breath you can hear in the mic and you want to get rid of that particular blemish, just set the expander to go on at a level just above the annoying breath sounds, and it will subtly drop out the offending noise.

Effects processors

Effects are historically used to mimic real-world situations. As a home recordist, you'll likely discover a great affinity toward your effects processors because they enable you to create sonic environments without having to rent some great recording room. For example, imagine dragging your drums and all your recording equipment into a large cathedral, setting them up, and spending several hours getting the mics placed just right. Sounds like a lot of work, right? (I get tired just thinking about it.) Well, how about recording your drums in your modest home studio and simply choosing the "cathedral hall reverb" patch instead? Now that's much easier.

I can practically guarantee that you'll use effects processors all the time in your studio. Scope out Chapter 17 for how to use them effectively.

In the world of effects processors, you have many choices, and many more show up every year. The most common effects processors are (in no particular order) reverb, delay, chorus, flanger, and pitch correction. Read on for the lowdown on each type.

Reverb

Reverb is undoubtedly the most commonly used effects processor. With reverb, you can make any instrument sound as if it were recorded in almost any environment. Reverb, a natural part of every sound, is the result of the sound bouncing around inside a room. The larger the room, the more pronounced the reverb. The purpose of a reverb in audio production is to make an instrument sound more natural (especially because most instruments are recorded in small, non-reverberant rooms) or to add a special effect. Reverb can make almost any recorded instrument sound better — if used correctly.

Delay

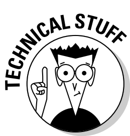
Think of delays as the recording studio's version of an echo. The delay can be set to happen immediately after the original sound or be delayed much longer. Delay can sound natural or be used as a spacey special effect. You can have a single echo or multiple delays (very common with the snare drum in reggae music, for instance). Delays are commonly used on vocals and guitar, although you can hear them on just about any instrument, depending on the style of music.

Chorus

A chorus effect can make one instrument sound like several. Chorus effects add very-slightly-off-tune versions of the unaffected sound, which results in a fuller sound. You find chorus effects used on vocals, guitars, and lots of other melodic instruments.

Flanger

A flanger (pronounced *flanj*-er) effect is similar to a chorus effect in sound except that the flanger gets its sound from *delaying* part of the affected sound in relation to the original, rather than altering its pitch. Recordists sometimes use flangers on background vocals and solo instruments to add an interesting texture. This is a unique sound that you recognize almost immediately upon hearing it.



The flanger effect comes from the early days in recording. You create the flanger effect the old-fashioned way by using a two-track recorder to record a duplicate track of the track that you want to flange. You then play the two identical parts back at the same time and gently press against the edge of the two-track tape (the one with the duplicate part) while it's running. This delays certain parts of the sound slightly and drastically changes the character of the instrument.

Nowadays, you can just choose the flanger patch (sound) on your effects processor to get this sound. Isn't technology great?

Pitch correction

Pitch correction, like its name suggests, is used to correct an out-of-tune note. You can use this effect to help a singer or an instrument player sound better by fixing those notes that are slightly out of tune. Pitch correction (also called *auto-tune*) has gotten a bad rap lately (mainly from its overuse and potential for abuse with a singer who can't sing in key). When used sparingly and appropriately, pitch correction can make an otherwise decent vocal performance really shine. It can also be used to create some interesting effects, such as that robotic vocal sound that you hear on so many of the pop songs on the radio nowadays. An easily distinguished example would be the lead vocal on Cher's "Believe."

Monitors

To record and mix music, you need to be able to hear it. (Hey, obvious facts need love, too.) Monitors make this happen. You can use headphones or speakers as monitors; most home studios use both. Monitors are an essential part of a recording studio because you need to get what you're recording and mixing into your ears for you to make sure that it sounds good.

Without good speakers, you won't know what your mixes are going to sound like on other speakers. (Find out more about mixing in Chapter 14.)

Headphones

Chances are that your first home studio will be in a spare bedroom or a corner of your garage or basement. All your recording, monitoring, and mixing will be done in this room. If that's the case, you'll find that a set of headphones is indispensable. When you use headphones, you can turn off your speakers and still hear what's being (or has been) recorded. When you go to record a guitar with a mic in front of the guitar amp, you want to hear only the guitar — not the guitar amp *and* the guitar amp coming back through your monitors. A good pair of headphones will allow you to do this. (See Figure 1-13.)

Figure 1-13: Studio headphones.

Speakers

For most home recordists, the first set of monitors consist of the home stereo system, but sooner or later you're gonna want a real set of monitors. Studio monitors come in many varieties, but the home recordist's best bet is a set of near-field monitors. *Near-field monitors* are designed to be positioned close to you (which is often the case anyway because most home recordists have very little room in which to work).

Near-field monitors come with or without an amplifier. The amplified monitors are called *active monitors,* and the non-amplified monitors are referred to as *passive monitors.* Which type of monitor you choose depends on your budget and whether you like the idea of the amp coming with the speakers or prefer to purchase the amp separately. Figure 1-14 has a picture of an active near-field monitor. The amplifier is located inside the speaker cabinet.

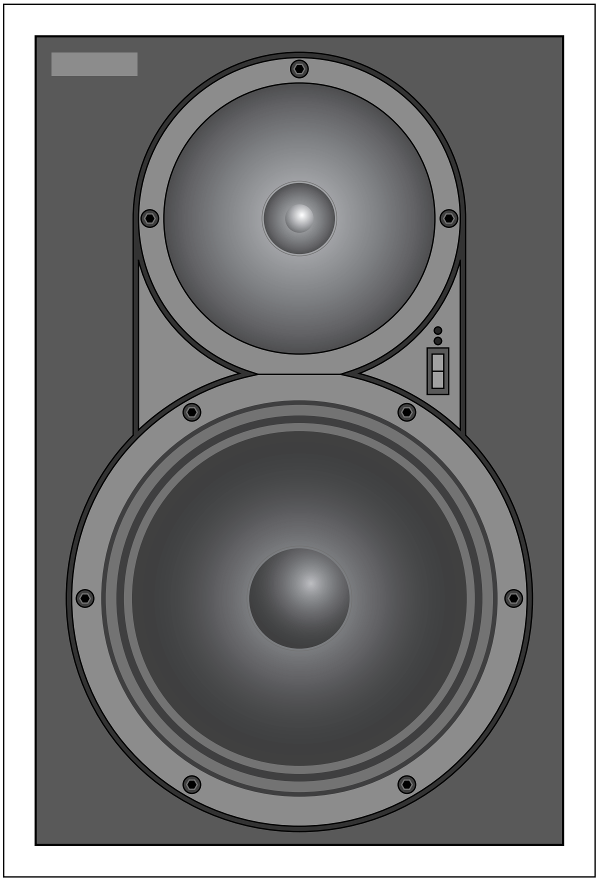
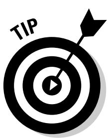


Figure 1-14: An active near-field monitor: The amplifier is located inside the speaker cabinet.



If you end up getting passive monitors, you need to buy an amplifier to send power to the speakers. The amplifier connects to the outputs of the mixer and boosts the signal to the speakers. A good power amp should be matched in power to work well with whatever speakers you have.

Mastering Media

After you mix your music, you need to put your final music on something. The two most common media for home recordists are CDs and raw computer files. Which medium you choose depends on what your goals are. For instance, if you intend to send your finished mix out to a mastering house, you're better off saving it as raw audio data in a computer file. On the other hand, if you master your music yourself and just want to have it duplicated (or you want to give copies for your friends to play), you would want to use a CD.

CD

With the cost of CD-R (write) and CD-RW (rewrite) drives plummeting, CD mastering is the only choice for most home recordists. With CDs, you can back up large amounts of data at a very low cost and burn audio CDs that can play in any CD player. You can even send out your mastered CD to be duplicated and packaged for retail sale. (Chapter 14 explains the process for burning CDs.)

Computer files

Sometimes you won't want to master your music directly to CD. You might decide to have a professional mastering house do it, or maybe you want to put your music on the Internet. In those situations, store your recordings as computer files. The following sections describe the most commonly used file formats for storing recorded music.

WAV and AIFF

WAV and AIFF (*A*udio *I*nterchange *F*ile *F*ormat) files are the formats for audio files found with most professional audio software. The advantage to saving your music to WAV or AIFF is that when you hand over a CD containing your WAV and/or AIFF files to a mastering house, the recorded sound is actually in a higher-quality format than that of the finished CD (provided your recorder records in 20 or 24 bit, which most do). You can also take your music files to any other studio that supports these file formats and work with them there.

MIDI

A MIDI file isn't an audio file; rather, it's a data file that contains MIDI information that can be transferred from one computer to another. An advantage to MIDI files is that they take up less room than an audio file. The disadvantage is that they contain only the MIDI information and no sound. To play a MIDI file, you have to have a sound module. And the sound you get from the file depends entirely on what sound source you use, such as the particular keyboard or soft-synth.

MP3

MP3 is a file format that has become quite popular on the Internet. Its advantage over audio CDs and other computer files is that it takes up less room. Its disadvantage is that the data is compressed, and the sound quality isn't nearly as good as that of commercial CDs (contrary to what MP3 proponents claim).