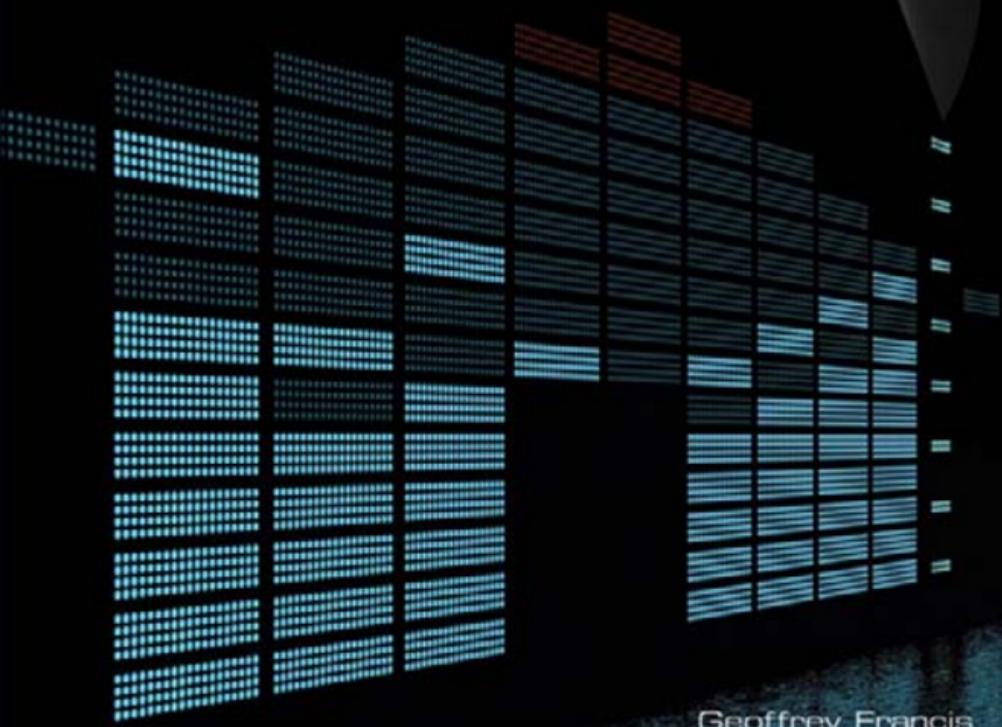


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# HOME RECORDING

## for Beginners



Geoffrey Francis

# Home Recording for Beginners

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Geoffrey Francis

**Course Technology PTR**  
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**Technical Reviewer:** Craig Richardson

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**Interior Layout Tech:** Macmillan Publishing Solutions

**Cover Designer:** Mike Tanamachi

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## About the Author

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Geoffrey Francis is a seasoned veteran of home recording with the best part of 20 years' experience. He graduated from Sydney's Professional Audio School in 1991 and rapidly discovered that this represented the beginning, not the end, of the learning process. He has built and successfully run his home studio in a variety of environments as diverse as an eight-track analog studio right under Sydney airport's main flight path and a state-of-the-art (well, almost) digital studio tucked away in the Tasmanian bush.

He cheerfully admits that along the way he has made just about every mistake in the book (as well as a few that aren't) and that he has learned more from these than from anything else. His approach is firmly rooted in commonsense reality. His underlying philosophy is that successful home recording is largely about rising to the challenge of doing the best with what you've got and in the circumstances in which you find yourself. For most people, that's the only option they have.

He is also the author of *Up and Running*, the official REAPER User Guide.

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# Introduction

I still remember my first experiences with digital audio. I had the advantage of many years' experience recording in an analog (tape) environment. I understood pretty well what were the main issues involved in recording and mixing. I also had been a reasonably high-level Windows user for quite some time. Surely it would be a simple task to marry the two together.

Boy, was I ever wrong! For starters, everywhere I turned, I got conflicting advice.

“You don’t want to use a computer. Get a dedicated hard disk recorder, like a Yamaha, or a Tascam, or a Roland. They’re better. They’re reliable, stable, and focused. No software conflicts. No more PC crashes!”

“Don’t even think of getting a dedicated hard disk recorder. The screen’s so small you won’t be able to see it, and it’ll be out of date before you even open the box. You need a Mac with Pro Tools.”

“Sure, if you want to spend a lot of money, go ahead. But a PC with Cubase will do the job just as well for about half the price.”

“Cubase, you must be joking! You want Logic. It’s much better.”

“Forget Logic. Cakewalk has more users worldwide than Cubase and Logic combined. They can’t all be wrong.”

Mercifully, that was 1996. Today, you could add, “...or Digital Performer, or Ableton Live, or Samplitude, or Tracktion, or n-Track, or SAWStudio, or REAPER or...”

Back to 1996. I settled on Cakewalk Pro Audio 8. I checked out the specs, system requirements and all, and purchased a brand-new PC to suit—about \$2,500 all together. Now I was ready to go.

I installed the program, opened it up, and...what the heck was this? Those menus, that screen layout...they were a million miles away from the Word, Excel, and PowerPoint that I was used to. What do I do next? Where do I start? I stared at the screen. It just stared back. I browsed the menus. The only commands I understood were huddled together for safety over on the left, on the File and Edit menus—you know, Save, Save As, Cut, Copy, Paste, and so on. Oh yes, and there was Help over there on the right. Not that it was much help to me. And as for the rest of it in between, it might as well have been written in Greek...time to phone a friend.

A week passed. I muddled through and finally plucked up the courage to start recording. Oops. How do I get the sound from my microphone into the computer? What's that? I need a PCI multi-in, multi-out soundcard (whatever that is)? And I have to watch out for latency? What the heck's latency? The last time I heard that word was in the physics class back in my school days (and I never really understood it even then). Not sure why, but there goes another \$1,000 or so.

Wait a minute, I'm using condenser microphones. Where's the phantom power coming from to drive those? No problem, sir, you just need a desk with a preamp to boost your microphones' signals before they go into the soundcard. Another \$500 later, and I thought I was starting to understand.

After that, I think it took me about four weeks of constant and intense anguish, agony, and gnashing of teeth before I managed to achieve much of anything. When I finally managed the mini-triumph of recording a simple ditty with four tracks and then attempted to create a mix (of sorts), I realized that my PC was seriously underpowered for the demands I would be making of it. Another \$1,000 for the RAM and CPU to fix that problem.

I could go on, but I think by now you get the point.

The purpose of this book is to help ensure that your transition to digital audio is a much smoother and more pleasant experience than mine was. I hope it helps you. Because if there's one thing guaranteed, it's that you're going to need all the help you can get at first.

## What You'll Find in This Book

In this book you'll find:

- A gentle introduction on a down-to-earth and practical, need-to-know basis of the issues involved in setting up a home studio.
- A CD with fully functional, uncrippled, non-expiring evaluation software (including a PDF User Guide), sample files, projects, and exercises for you to work on.
- A step-by-step, lesson-by-lesson, 30-day foundation course that will enable you to work through manageable examples that will introduce you to the main tasks involved in successful home recording, all the way from setting up a basic system, to recording your first instrument, right through to burning your first CD.
- A solid foundation from which to progress in your own time, at your own rate, and according to your own needs.
- The opportunity to grow in knowledge, competence, and confidence as the jargon and the recording environment are demystified.

- A structured learning program that lets you do this *before* laying out large sums of money. Most musicians will already have in their homes the minimal equipment needed to complete this course. For those who don't, the outlay required will be no more than about \$100.
- The opportunity to evaluate the issues involved in home recording and the extent to which you get involved before you need to spend any of your hard-earned cash on equipment.
- Information and advice on how to arrive at purchasing decisions that strike the optimum balance between cost and quality for you.
- A strategy to identify just what equipment you need, what you don't need, and how to get the best overall recording equipment to suit your particular budget and circumstances.
- A strategy for further growing, developing, and improving your home studio.

## Whom This Book Is For

*Home Recording for Beginners* is for anyone who is considering setting up a home recording studio or who has previously tried this and found it too confusing, too frustrating, or too difficult. Although written and designed primarily around the needs of the PC user, Mac users may also find it of benefit because 90 percent of its content is equally applicable in either environment.

No previous experience with recording is required to benefit from this book. A reasonable level of computer proficiency and familiarity with your operating system (for standard tasks such as installing software and organizing and backing up files) is, however, assumed.

## How This Book Is Organized

*Home Recording for Beginners* is divided into just five chapters plus an appendix. The appendix contains a glossary of terms.

The first two chapters have a practical, hands-on approach with an emphasis on learning by doing. The remaining three chapters use that experience as a foundation on which to graft an understanding of the technical issues that you will also need to come to grips with and enable you to spend your money wisely when purchasing equipment.

- **Chapter 1, “Your Pathway to Home Recording.”** This chapter introduces you to home recording and helps you to distinguish between the realities and the many myths that are prevalent in this area. It lays out clearly what you will need to work through the 30-day foundation course.
- **Chapter 2, “The 30-Day Foundation Course.”** This chapter introduces you systematically, step-by-step, and in a logical sequence to the many disciplines and skills required

for successful home recording. Most importantly, it will guide you to an understanding of how all these pieces fit together. Among the key topics covered in this section are:

- Understanding digital audio and home recording
- Preparing yourself and your computer for home recording
- Understanding and navigating the software interface
- Creating a recording project
- Preparing and recording a simple track
- Recording further tracks: layering and overdubbing
- Identifying, understanding, and resolving potential problems when recording
- Using the clever stuff: auto-punch recording and multiple takes
- Understanding and working with MIDI
- Creating and editing MIDI tracks
- Creating and using loops
- Understanding and using samplers and sampling
- Mixing essentials: volume and pan faders
- Editing your recorded material
- Using essential plug-ins and FX: EQ, compression, and delay basics
- Understanding essential mixing techniques: using submixes and folders
- Understanding more essential mixing techniques: using track routing and busses
- Using the fun stuff: pitch and play rates
- Using more of the clever stuff: markers, macros, and automation
- Rendering for CD audio or Internet distribution

At the end of the 30-day course, there is a simple checklist to help you draw up a focused development plan. It will enable you to identify the primary areas on which

you will need to concentrate most of your energies and efforts, the secondary areas you will also need to consider, and those areas you can ignore altogether.

- **Chapter 3, “What Else Do I Need to Know?”** Having rolled up your sleeves and worked through a series of real examples, you’ll be taken behind the scenes to understand some of the science that underpins it all. This chapter will provide you with a foundation of essential technical knowledge that you will need if you are going to progress. It demystifies terms such as *audio formats, bit depth, sample rate, audio drivers, impedance*, and more.
- **Chapter 4, “Setting Up Your Studio.”** This chapter shows you where you will need to go from here. It guides you through the different equipment that is available and helps you to identify for your own situation what you absolutely need, what might be worth considering, and what you absolutely don’t need. It will help you to achieve the best possible outcome for your budget, as well as offer sound advice on how physically to go about setting up your studio at home.
- **Chapter 5, “The Road Ahead.”** If you’re serious about home recording, it will be an ongoing commitment. This chapter discusses the road ahead and helps you to develop longer-term goals and perspectives.

# 1

# Your Pathway to Home Recording

Today's technology means that almost all musicians can have in their homes the tools they need to produce recorded material to a standard that even 15 years ago could only have been possible in a professional studio.

## Myths and Realities

Sound good? Well, potentially it is, but unfortunately home recording is being sold to a largely uneducated consumer market. By "uneducated," I don't mean unintelligent, ignorant, or stupid, but simply lacking sufficient knowledge to make informed decisions about what they are doing. Too often, alas, this is encouraged by those in the industry who have a vested interest. Here, for example, is just one quote from an article in a well-known audio magazine a while back:

"Today's cheap computers combined with affordable software means that any musician can quickly and easily produce professional-sounding recordings in their own home."

Statements like that are absurd. Having the tools alone isn't enough to do the job, not by a long shot. Put quite simply, the tools are of no use to you if you don't know how to use them. And the best tools don't always come cheap. Let's put aside the myths and start by looking at some of the realities of home recording.

### How Difficult Is It and How Long Will It Take Me to Learn?

Anything is easy once you know how to do it. The trouble is that getting to that point isn't always quite so easy. Whether you're talking about recording, hairdressing, brain surgery, or anything else, there is no getting away from the fact that it takes time to become proficient.

In the case of recording, if you're doing this for the first time, you can't expect to get good outcomes until you have put in the time. How much time depends on many factors, but as a rule of thumb, it'll likely be five years before you really get up to anything

like a professional standard. Nevertheless, if you apply yourself, you can be pretty sure that in 30 days you'll learn enough to obtain:

- A solid grounding in what home recording involves. Hopefully, the mystique and the mystery will have been stripped away. You'll have a good idea of what you're in for if you decide to carry on.
- Whether this home-recording caper is for you. You might find that you have a natural flair and aptitude for this sort of thing. You might find it harder going than that, but well worth the effort. On the other hand, you might conclude that the reward-to-effort ratio simply does not make it worthwhile for you. In any event, you'll be only a little older and a lot wiser without being any poorer!
- If you do decide to go on, you'll know where you should be going from here. Instead of plodding along blindfolded and hoping you're not missing anything important, you'll have a clear plan of action.

## **What Do I Need to Know before I Start?**

The short answer to this is not a lot. This foundation course makes only a limited number of assumptions about where you're at now. In particular, I assume that:

- You are comfortable operating your computer and getting around its interface. I'm not talking about advanced technical knowledge here, just things like managing files and folders; using standard features such as drag and drop, mouse clicks, and menus; and finding your way on to and around the Internet.
- You have a basic understanding of music. For example, I won't attempt to teach you how to sing or play a guitar or keyboard.
- You have absolutely no experience whatsoever with recording or with editing, mixing, or otherwise working with recorded material. If you have any such experience, that's a bonus.

## **How Much Will I Need to Learn?**

The short answer is one heck of a lot. You'll need to develop a multidisciplinary knowledge of a wide range of topics and disciplines. It's part science, part art, part common sense, and part intuition. But don't worry; we'll take it step by step and proceed on a need-to-know basis. I won't throw everything at you at once.

One of the biggest barriers to progressing with your home recording skills is simply that you don't know exactly what you need to know in order to progress. There are just so

many blind alleys and diversions. In which areas should you be concentrating your energies, and which should you leave alone for now? It's not uncommon for people to get far along the track, only to find out that they have overlooked something absolutely crucial. It starts to get a bit like a game of Snakes and Ladders.

The first two chapters of this book aim to prevent that. They will help you acquire all the knowledge you need to get started. You will learn just enough about a large number of topics without exploring any single topic in depth. Eventually, there will come a time for you to be adventurous and follow your instincts to explore those aspects of recording that are of most interest to you in greater depth and detail. That time is not now. You'll progress much better if for the time being you are able to curb your natural curiosity and focus on laying down the solid foundations on which you can later build. Trust me!

## How Much Time Will I Need to Spend?

You need to spend as much time as you're comfortable with, without risking mental overload. The good news is that most of this course has a practical emphasis. This means the more you do, the more you will reinforce what you are learning. In other words, the more you put into it, the more you'll get out of it.

Ideally, you should seek to spend at least one—and sometimes two or three—hours each day working through these examples, exercises, and assignments.

## How Much Money Will I Need to Spend?

This is an easy one to answer: *Very little at first, but eventually quite a lot!*

To be serious, how much money you'll end up spending is largely up to you. I'll return to this topic in more detail in Chapter 4, "Setting Up Your Studio." You should find that there are home recording options for most budgets, although generally speaking, if you want to get the best results, you'll need quality equipment (such as microphones and studio monitors), and that doesn't come cheap.

One thing you definitely don't want to do is waste a lot of money up front on a lot of gear that you don't need and won't use. Look through any audio store's catalogue or visit their website, and you'll see what I mean.

I'll start you off with the absolute minimum outlay you'll need to complete a 30-day foundation course. I can absolutely guarantee that with this gear, you won't get a professional-sounding product or anything like it. That's not the object of this course. The object is for you to learn the basic principles and practices of home recording and to be able to evaluate whether this caper is for you.

If it is, then we'll go on to the next steps together. If it isn't...well, instead of wasting thousands of dollars, you'll have blown a hundred or so at most, and you'll have had a bit of fun along the way.

## **What Equipment Will I Need?**

Assuming that you already have your computer, you'll be surprised at how little you will need to get in order to work through this foundation course. We'll come back to this point soon and identify some inexpensive items that are good enough to get you started.

However, you should also be aware that the more seriously you take this recording business, the more equipment you will eventually need. You will probably never reach the point where you have every piece of equipment that your heart could possibly desire.

## **Will I Need to Build a Dedicated Studio?**

The short answer to this one is probably not.

At this point in time, don't even consider it. Although it is true that a well-designed and purpose-built studio environment should offer you the opportunity to produce better outcomes, at this stage in your learning that should be the very least of your concerns. You'll be much more limited right now by the boundaries of your own skills and knowledge and by the equipment that you'll be using than by not having a vocal booth or an acoustically treated room.

For the time being, keep this in mind: Many absolutely dreadful recordings have been produced in professional studios, and many first-class recordings have been produced in the corner of somebody's bedroom. It'll be years before you'll need to cross this particular bridge.

## **What You Will Need**

The deeper you get into home recording, the more you'll find yourself wanting to spend on various items of equipment. To get started, however, you need surprisingly little. Following is a summary shopping list—but don't spend a single penny until you've read right through to the end of this chapter.

- A computer with an Internet connection. I assume you already have this. This book is written around using a PC, but you may use a Mac if you prefer. The notes are based on Windows XP, but they can, with some modification, be applied to other versions.
- An internal computer soundcard and a pair of speakers. These could be external speakers connected to the soundcard by means of cable, or they could be built into the computer.

- A microphone and cable to connect to your soundcard, together with a microphone stand and clip.
- Headphones possibly (but not necessarily) with a built-in gooseneck microphone that can also connect to your soundcard.
- Recording software. I'll say more about that shortly.
- Ideally, a friend who is also interested and who is also happy and willing to act as your partner and guinea pig in this venture. You'll usually progress better this way than you would on your own. This might cost you a few beers, but what the heck—it's worth it! Don't worry, though, if you are going to be on your own. You won't be the first person to have done this.

This equipment is available from a wide range of sources and at a wide range of prices. For example, a cheap basic microphone from RadioShack will cost you no more than about \$20. On the other hand, a top-quality Sennheiser from a recognized audio dealer might set you back several thousand dollars. But you don't need that yet!

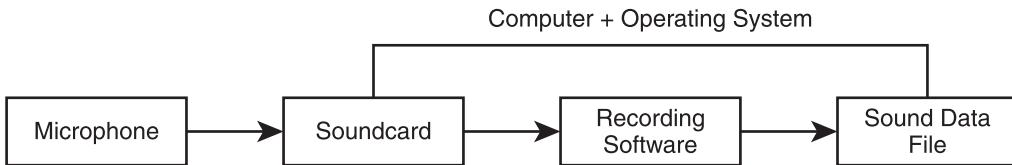
If you already have any quality gear, it's fine to use it for this course. If you don't have such equipment, you'd be well advised to think very carefully before spending your money. You could end up with some very expensive equipment that just isn't the best gear for you.

Instead, I'm going to suggest that for the purposes of learning, you invest in budget equipment only. Assuming that you already possess a PC with a built-in soundcard and a pair of speakers, you should be able to kit yourself out at a store such as RadioShack or Tandy for about \$50 or \$60. Of course, you will never achieve professional results with such equipment (or anything like it), but at this stage you won't know nearly enough to get professional results anyway. What's more, when you've finished the course, you will still be able to use this gear for your Internet phone calls and voice communications with Skype and so on.

Before continuing, it's worth taking a few moments to understand why these items are necessary and how they all fit together. You're going to be using your computer and this equipment for two main purposes:

- To record music
- To play it back

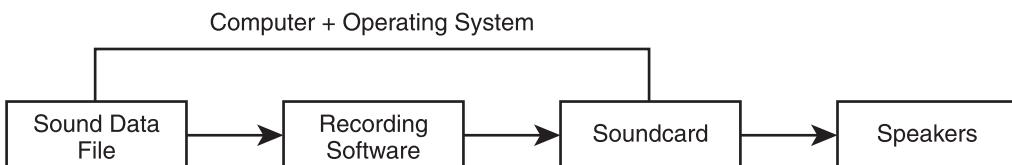
Of course you'll be doing much more than just this, but these are the core functions on which we'll be focusing, at least at first. Let's take a look at how this happens in Figure 1.1.



**Figure 1.1** Basic audio flow into the computer.

The diagram in Figure 1.1 shows the basic signal flow when you record an audio signal into your computer. The sound waves captured by the microphone are passed to your soundcard, where (under the control of the computer's operating system) they are passed on to the recording software application you are using. This software writes the audio to your hard disk as a sound file.

Now let's see what happens when this recorded sound file is played back (see Figure 1.2).



**Figure 1.2** Basic audio flow out of the computer.

In this case, your software application reads the contents of the recorded data file from your hard disk and generates an audio data stream from it. This signal is passed, under the control of your computer's operating system, to the soundcard and on to the speakers.

Sounds simple enough, doesn't it? Well, get ready for this.

No matter how simple this sounds, getting the interaction between these various items right is undoubtedly the greatest single cause of grief, anguish, and heartache to beginners—and not only to beginners. There is a host of issues involved in not only getting the different items to communicate with each other at all, but also in getting them to communicate in an effective and efficient manner.

During this course, I'll be helping you to identify and resolve these issues. Don't worry—you won't be hit with them all at once, and once you understand what's going on, it isn't difficult. But it does take a little time to get there.

Now, on to that shopping list...

## Home Recording: Getting-Started Shopping List

Assuming that you already have a home computer, you should not need to spend a great deal of money on the various items of equipment that will be essential for you to be able to follow this 30-day course. *Please remember, however, that for the most part we are talking about inexpensive consumer-standard equipment here.* If, after you complete this course, you decide to get seriously into recording, then it becomes a different story. In Chapter 4 you will find this issue discussed in some depth.

### The Soundcard

Your computer should already have an onboard soundcard. If it does, this is perfectly adequate for the purposes of our 30-day foundation course. We'll be spending a whole lesson exploring your soundcard to make sure that you are familiar with its functions and characteristics.

For now, just make sure it's there. If you are using Windows XP, open the Windows Control Panel and go to Sounds and Audio Devices. Click on the Audio tab. You should see a display similar to the one shown in Figure 1.3. Make sure your PC's soundcard is selected as the default device for both playback (output) and recording (input). In the example shown here, the PC has a Realtek soundcard. Yours might be something different (it probably is).



Figure 1.3 Windows XP Sounds and Audio Devices Properties dialog box.

If you are using Vista, look for the Control Panel item called Sound and open it (see Figure 1.4). You will then be able to identify by the information shown on the Sounds and Recording tabs whether a soundcard has been installed.

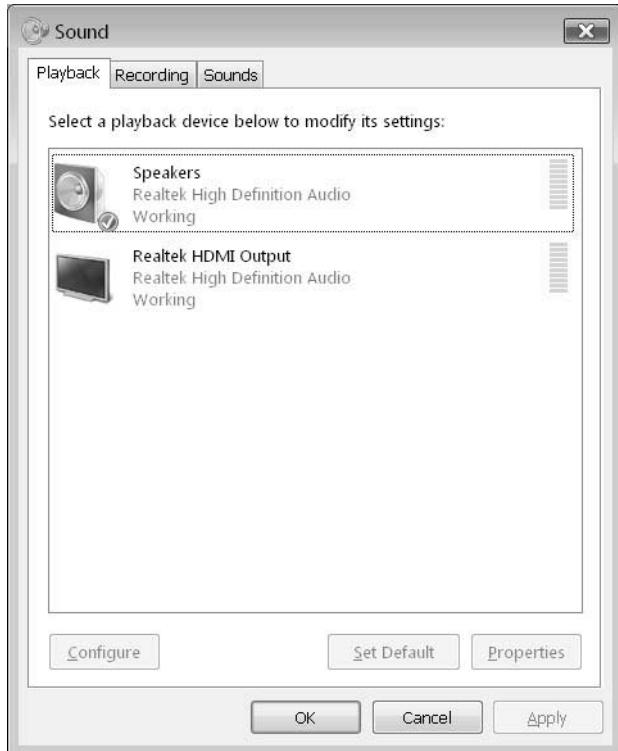


Figure 1.4 Windows Vista Sound dialog box.

If no device is available to you, it's time for a visit to your computer dealer to have your system checked. It's possible that there is one there physically, but for some reason Windows is not recognizing it. This might happen, for example, if the device has been disabled. If there is no physical card, you will need to have one installed. Go for an ultra-budget-priced model at this stage, \$20 at the most. If, after completing the foundation course, you decide that you want to go on to do some serious recording, you will need to replace it with a good-quality (and more expensive) soundcard. I'll discuss this in Chapters 3 and 4.

## The Microphone

Microphones come in all sorts of shapes, sizes, and types. One of the essential elements of producing a good-quality recording is knowing which microphone to use for recording different instruments and voices in different situations. Before you can do this, you will need to understand the main classes of microphones, together with the characteristics and differences of each type.

Don't worry about this for now. This, too, will be discussed in Chapters 3 and 4. If you already have something like a Shure SM58, that's great. If you don't even know what that is, don't worry. A consumer-standard all-purpose dynamic microphone will be fine for now.

Figure 1.5 shows a simple unidirectional cardioid dynamic microphone from Radio-Shack, costing about \$20 or \$30. I'll explain what these terms mean later, so don't worry if they are new to you; for now just make sure that the microphone is dynamic and that its pattern is cardioid. You won't win any awards using the example shown here, but it will serve you just fine for your 30-day foundation course. It comes with a lead, but you may also need an adaptor to ensure that it can plug into your soundcard.

## The Microphone Stand

If you're used to gigging, you've probably got a microphone stand or two already, maybe a bit worse for the wear and held together with gaffer tape. If so, it will be fine to use it for this course, but down the road, if you're going to be doing any serious recording, why not treat yourself to a new stand or two just for that purpose?

This is one area where it probably does pay to go for a reasonably good-quality item from the outset. Cheap microphone stands tend to be flimsy, unstable, and short lived. Get a decent, robust model. You'll also find that a stand with telescopic booming offers you more flexibility and options for recording, especially if you wish to record instruments when sitting down.

If you don't already have any sort of microphone stand at all, about \$30 should buy you something basic to get you started.

## The Microphone Clip

This is only a small item, but don't overlook it. In many cases, a microphone comes supplied with a clip, and it's even possible that this might be the case with your microphone stand. If not, you'll have to buy one. Make sure that the clip you purchase screws neatly onto the stand and will also hold the microphone in position firmly. This isn't a case of one size fits all.



**Figure 1.5** An inexpensive consumer-standard microphone.

## The Headphone Set

Two important parts of the recording process that we'll get to later in the course are overdubbing and layering. Both of these techniques involve recording new material over the top of existing material. This can happen, for example, if you have recorded a guitar track and you wish to add a vocal to it.

To be able to do this, you'll need a pair of headphones. The item shown in Figure 1.6 comes in at about \$20. Again, if you're going to go on to do serious recording, light-weight headphones like these won't be of much use to you. If you already possess a decent pair of headphones, then use them. Otherwise, something like the item shown here will do for now—and when the time comes to move on and up, they'll still be great for Skyping!



**Figure 1.6** An absolutely minimalist headphone set like this will be good enough for use in this course.

## Hardware Limitations: A Disclaimer

Unless you already possess a computer with a fair bit more grunt than the typical home PC, and perhaps a \$500 pro soundcard rather than the consumer-standard card that comes with most PCs, you'll likely find that some of the clever things I cover in this foundation course will be beyond the capabilities of your humble system. We'll deal with these matters as they arise (there aren't many of them), but this shouldn't prevent you from understanding what's happening, nor from achieving your learning objectives. Of course, if you already possess a professional-standard soundcard, by all means use it for this course. In that case, you should disable any possible conflicting device, such as the computer's inferior onboard soundcard.

## The Recording Software

We've left the most interesting item to last—the choice of which recording software to use for this course. Jump onto any audio web forum, and you'll soon discover that many audio users are as passionate about their software platform of choice as others are about their football team, their politics, or their religion. More often than not, the one true recording package they swear is “the only one to use” is just that—the only one they have ever used. This doesn't really put them in a position to give fair, balanced, and objective advice, does it?

Let's clear up some of the misinformation and confusion. Popular home recording software programs for use on the PC include (in alphabetical order) Ableton Live, Cubase, n-Track, Nuendo, Podium, REAPER, Samplitude, SAWStudio, SONAR, Tracktion, and more. They are often called *digital audio workstations*, by the way—or DAWs for short. The truth is that any of these programs will give you all of the basic functionality that you need for your home recording studio. Probably, you could get along quite happily with any of them.

However, as well as a substantial core of common features, each also has its own individual characteristics, its own strengths, and its own weaknesses. Every single one of them misses some bells and whistles that you will find in one or more of the others, and most can boast features of their own that are absent in many others. Every one will contain all sorts of wondrous features that you will never need or use. Each one also has its own characteristic signature, by which I mean a design philosophy, or a method of working. Which you prefer is very much a matter of individual taste.

When you are starting out, you cannot possibly reach an informed decision about which one will ultimately be the best for you. Moreover, to suggest that you try them all to see would, at this stage, not only be impractical, it would also be a bit like taking cars out for a test drive before you learn how to drive.

Nevertheless, we do need to standardize on a product for the purposes of this course. It simply isn't possible to give nine or ten different sets of instructions for every task that you might possibly wish to perform. So, for learning purposes, I've settled on REAPER as the preferred software for this course. My choice of REAPER is not in any way intended to denigrate any other software; it has been made for the following reasons:

- You can legally install and use a fully functional, fully featured, uncrippled, and unexpiring version of the software for up to 30 days before you are required to pay for a license. A copy of the REAPER install file can be found on the CD that is included with this book.
- If you do decide to purchase a home user's license, it will cost you only \$50, compared to several hundred dollars for most of the other products mentioned.
- Unlike most other DAW software, REAPER writes nothing to your Windows registry. If you later decide to uninstall it, you can do so without leaving any footprint.
- You can get a PC version or a Mac OS X version. (At time of this writing, the Mac version is available in beta only.)
- REAPER is my own DAW of choice. It is the one that I use in my own studio, the one that I am most familiar with, and the one I like the best.

## What about the Technical Stuff?

You might have noticed that up until now, you have encountered very few technical terms, and I have made very few references to technical issues. The term digital audio workstations is about as technical as we've gotten so far.

So what about the technical aspects of recording?

It is true that there are a number of technical terms of which you will sooner or later need to develop a degree of understanding. I can throw a few at you now if you'd like:

- Audio drivers
- Sample rates
- Bit rates
- Rendering
- Buffers and buffer size
- Latency

- Clipping
- Plug-ins
- Decibels
- VST

There are a few to begin with. Now forget about them—at least for now—because technology cannot be taught or understood in a vacuum. I'll introduce these terms (and others) to you gradually, as we go along. You'll be learning by doing. In many cases, this means you'll come to understand the concept and how it is applied before you even know what it's called.

There's just one technical issue we'll need to get sorted out before we begin, though, and that's to make sure your computer's specifications are good enough to at least let you use it to take your first steps into the world of home recording. Another reason for choosing REAPER, by the way, is because it is less fussy in its requirements in this regard than are most of the other programs you might choose to use.

Table 1.1 lists the minimum and recommended specifications for running REAPER on a PC. At the minimum end of the spectrum, you'll likely find that your computer might not be able to handle some tasks or that it will perform them poorly. At the recommended end (and above), you're far less likely to have this problem.

**Table 1.1** Minimum and Recommended System Specs for Use with REAPER

Minimum System	Recommended System
500-MHz processor	Multiple processors
Windows 98/Me/2000/XP/Vista/Wine	Windows 2000/XP/Vista
128-MB RAM	1-GB RAM
10-MB free disk space	More than 1-GB free disk space
Resolution 800×600	Resolution 1280×1024 or higher
256 colors or higher	16.7 million colors
Windows-compatible sound hardware	Professional-standard soundcard with ASIO drivers

Don't worry if your computer's specifications resemble those listed in the first column more closely than they do those in the second. You'll still be able to complete this course. If you do decide at some stage to get seriously involved in home recording, you'll probably end up getting a separate computer dedicated to this purpose. Now let's move on to the 30-day challenge....

## Problems: How to Avoid Them and How to Solve Them

Digital audio can be a CPU-intensive application. I recommend that when you are working through the exercises and examples in this book, you close unnecessary applications and stay offline (except, of course, when the instructions require you to be online or to open another application).

You might find that regardless of how careful you are, there may be times when the instructions in this book just don't seem to work as they should. This could be for a number of reasons. It could be because of your hardware setup. It could be because there is just one little point that you've misread. It could be because of some change in the software since this was written.

In any event, help is on hand anytime from the REAPER User Forums. You can log on to the General Discussion Forum at [www.cockos.com/forum/forumdisplay.php?f=20](http://www.cockos.com/forum/forumdisplay.php?f=20) and ask your questions. You'll need to register first, of course, but registration is free.

## Getting Started in 30 Days

How far do you think you can get in 30 days? You're about to find out. Stick with this plan, and you'll be amazed at how far you'll travel in such a short time.

Remember, the more you put into it, the more you'll get out of it.

**Table 1.2** The REAPER 30-day Challenge: An Overview and Checklist

Day	Activity Done
Day 1	Digital audio and home recording—an introduction and overview
Day 2	Windows and your soundcard
Day 3	First steps in REAPER
Day 4	Projects and files—the basics
Day 5	First steps in recording

**Table 1.2** Continued

<b>Day</b>	<b>Activity Done</b>
Day 6	Second steps in recording
Day 7	Recording additional material
Day 8	Consolidation exercise
Day 9	Overdubbing
Day 10	Auto punch recording
Day 11	Recording and working with multiple takes
Day 12	Basic MIDI recording and editing
Day 13	Creating and using loops
Day 14	Sampling: where audio meets MIDI
Day 15	Consolidation exercise
Day 16	First steps in mixing: pan and volume
Day 17	Introduction to audio editing
Day 18	Plug-ins and FX: EQ
Day 19	Plug-ins and FX: compression
Day 20	Plug-ins and FX: delay
Day 21	Submixes and folders
Day 22	Basic routing: sends, receives, and busses
Day 23	Consolidation exercise
Day 24	Fun with pitch and play rates
Day 25	Markers and custom actions
Day 26	Automation envelopes
Day 27	Bringing it together in the mix
Day 28	Rendering for CD or MP3
Day 29	Consolidation exercise
Day 30	Consolidation exercise

Some of these topics might already mean something to you, while others at this stage might make no sense at all. Don't worry—by the time you've finished the 30-day course, it will all make sense.

# 2

# The 30-Day Foundation Course

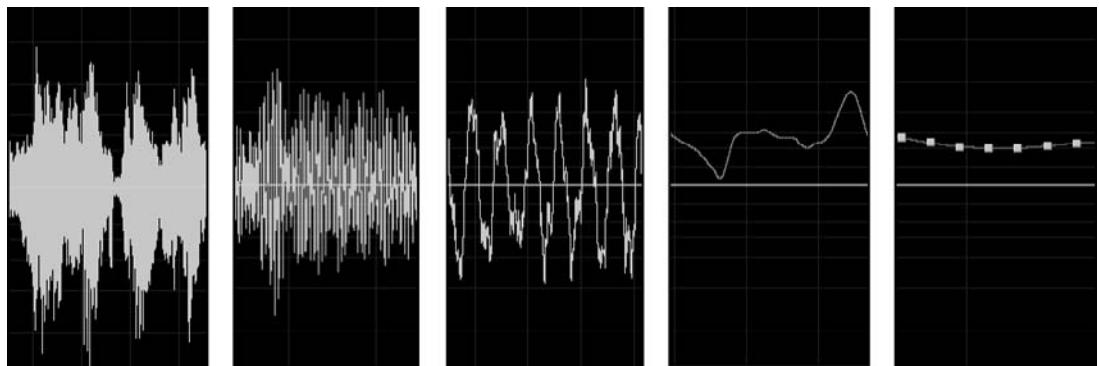
The last few decades have witnessed an extraordinary rate of technological change. Today you are likely to see working side by side in the same studio people who have 20 or 30 years of experience in recording with analog tape and people whose only exposure to this technology is through a book or a museum visit.

## Day 1: Digital Audio and Home Recording

In this first lesson, you will be introduced to the most basic of concepts of what digital recording is, together with a very basic overview of what steps are involved.

You probably already know that sound is made up of a series of waveforms. With analog (tape) recording, these waves form a continuous stream. With digital recording, however, the sound is captured as a series of discrete samples. Because the sampling is done at such a high rate and speed, when the digital recording is played back, the human ear is unable to discern this. Thus, the sound appears to be continuous. This is in essence the same principle that applies to pictures in your newspaper or magazine (which are actually made up of a very large number of tiny pixels) or to old-fashioned movie reels or flicker books.

The series of illustrations in Figure 2.1 illustrates this.



**Figure 2.1** Recorded audio material zoomed in to show (left to right) 20 seconds, 1/5th second, 1/50th second, 1/1,000th second, and 1/10,000th second.

The first illustration (left) represents about 20 seconds of a digitally recorded piece of music. Each successive picture zooms in closer and closer until, by the time we get close to one ten-thousandth of a second, we can see a discrete series of dots (samples) that actually make up the sound. The line shown joining those dots is actually an imaginary one.

This brings us to the topic of bits and bytes. For now, all you need to know is that all information in the computer, whether it's the text of a document, a photograph, or a recorded piece of music, is stored as a series of bits and bytes, zeroes and ones. We'll examine things such as different bit rates when they become relevant.

## Audio and MIDI

You will be working with two kinds of media files—audio and MIDI. At this stage, you only need to be aware of one major difference between these two.

- **Audio files** are actual recordings of sound. You can think of them as being just like sound recorded to tape, except, of course, they are digital rather than analog. If you copy the file to a flash drive, you can play it back through any computer using any audio program, and it will sound essentially the same. For example, if you have a recording of a trumpet playing “Silent Night,” then no matter how many different computers you play it back through, it will always sound like a trumpet playing “Silent Night.”
- **MIDI files** do not contain any recorded sounds. What they do contain is simply a series of instructions to a computer (under software control) that enable the computer to generate sound every time the MIDI file is played (or, strictly speaking, processed). By editing a MIDI file, it is perfectly possible to change the instructions so that the sound created by the file is also changed. For example, if you have recorded a series of instructions telling the computer to create the sound of a trumpet playing “Silent Night” to a MIDI file (usually but not necessarily via a keyboard), you can edit that file to change the instructions so that the next time the file is processed, instead of a trumpet a flute is heard.

MIDI files are used in conjunction with synthesizers, which can also be used to shape the sound. A synthesizer, by the way, doesn't need to be a keyboard or any other piece of external hardware. It can also be simply a piece of software that resides in your computer and is used in conjunction with your recording software.

## The Five Main Project Stages

In a couple of days you'll be ready to embark on your first recording project! Before doing so, it will be useful if you have some roadmap to show you where you are going.

Having an overview of the bigger picture can be quite helpful, especially if there are times when you feel you might be losing your way.

Any recording project can be thought of as progressing through five main stages (see Figure 2.2 and Table 2.1).



Figure 2.2 The five stages of a recording project.

**Table 2.1** The Five Stages of a Recording Project

Stage	Explanation (Overview)
<b>Recording</b>	This is the process of getting your music into the computer in some usable form. It will usually involve recording a series of audio files, MIDI files, or both. Recording is sometimes called <i>tracking</i> .
<b>Editing</b>	This is the process of polishing up your recorded material to make it sound better. For example, there might be some sounds that you have accidentally recorded that you want to remove.
<b>Mixing</b>	Usually a recording will consist of a number of instruments and/or voices. For example, you might have a vocal track, a guitar track, a fiddle track, and some percussion. Mixing involves balancing the various tracks so that everything can be heard clearly.
<b>Rendering</b>	Rendering is the process of converting your mixed recording into a single audio file (usually stereo) almost ready for distribution via, for example, audio CD or the Internet.
<b>Mastering</b>	During the mastering process, the previously rendered file is given a little more polish, perhaps brightened up a little or given a touch more final punch to make it more appealing to its intended audience.

There's actually another stage that comes before all of these—preparation! I'll leave that out for now, because until you've had at least a taste of what is involved in recording, you won't know what you need to prepare.

During this course I'll take you through the first four of these stages. I'll say something more about what mastering is, but it would be too ambitious to attempt mastering at this stage. When you get to the point where you're ready to learn about mastering, you might want to check out *Mastering Music at Home* by Mitch Gallagher (Course Technology PTR, 2007)—but that won't be for some while yet!

## Assignment

Jump onto the Internet and do a search for digital recording. Browse through the results and visit any links that seem especially interesting to you. If you find there's nothing that interests you, give up now. You're in the wrong game.

## Day 2: Windows and Your Soundcard

Our first task will be to examine the ways in which your PC and the Windows operating system work together with your soundcard. This is because your recording program will need to interact with your soundcard just about all the time. It is useful (and perhaps even interesting) to have some understanding of what is going on.

The basic principles will be the same regardless of which soundcard you are using. However, many of the actual details (such as what you will see in your settings windows) will vary, depending on which card you are using. This should not matter because for the time being, we are exploring this issue only in overview.

### Understanding Your Soundcard: XP

Let's now take a tour of your soundcard. Don't worry if you don't understand everything that we're going to do right now. If you get the general drift, you're doing just fine.

The notes on the next couple of pages apply if you're using Windows XP. Read them anyway, even if you're using Vista, because many of the principles are the same. More about Vista later, in the "Understanding Your Soundcard: Vista" section!

Open your Windows Control Panel and double-click on Sounds and Audio Devices Properties (see Figure 2.3). The dialog box that then opens has five tabs, arranged in a rather strange order. The first things you need to check are your audio input and output devices.



Figure 2.3 Windows XP Sounds and Audio Devices Properties: Audio tab.

### **Audio Devices**

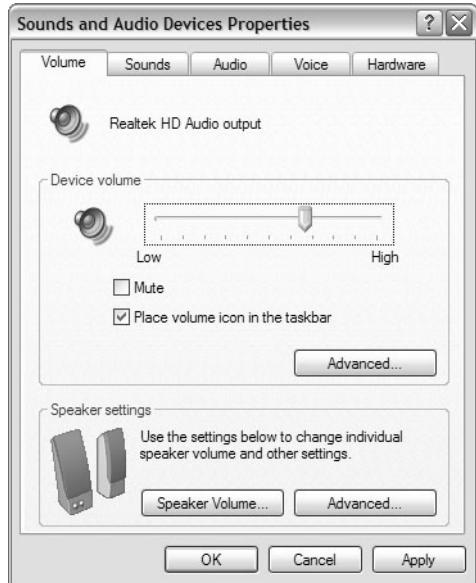
Click on the Audio tab, and you'll see a display similar to the one shown in Figure 2.3. It's possible that your soundcard might offer more than one option for any of the three default devices shown. For example, many standard cards these days have a stereo 3mm input (often colored pink) in the card itself at the rear of the computer and another one more conveniently located on top of or in the front of the computer. In that case, you could select either as the default sound recording device.

If Microsoft GS Wavetable is available for MIDI playback, select this as your default output. Otherwise, we will return to this subject later.

### **Volume**

If you click on the Volume tab, you will see a number of settings similar to those shown in Figure 2.4. In particular, notice that you can:

- Set the maximum output volume for your audio device. As you will see later, you can use the Windows mixer or your recording software's mixer to reduce the output volume below this level.
- Choose to display a volume icon on your Windows taskbar (toward the right-hand side). This can be useful for making adjustments to your settings on the fly, without needing to open the Control Panel each time.

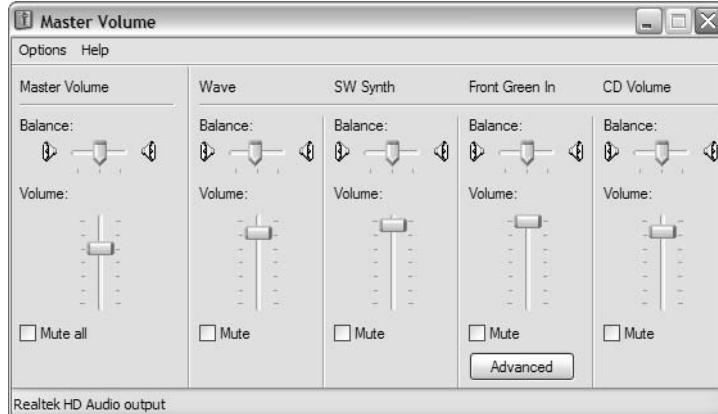


**Figure 2.4** Windows XP Sounds and Audio Devices Properties: Volume tab.

Notice also the Advanced button in the Device Volume section.

This causes your soundcard's Master Volume mixer to be displayed (see Figure 2.5). Of course, you might not have exactly the same selection of items as shown here.

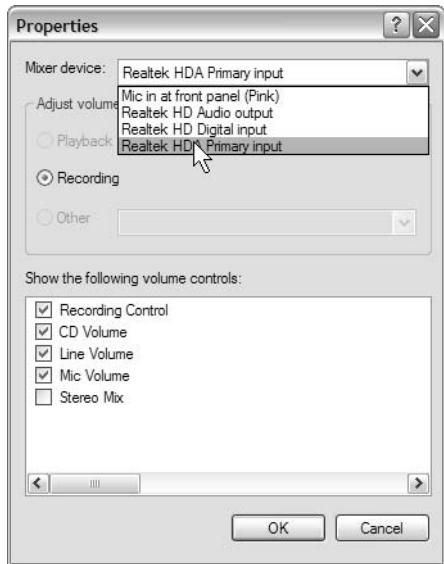
You can use this to control the level of output produced by different items. You can also adjust the balance between left and right speakers for any items (but you wouldn't



**Figure 2.5** Windows XP Master Volume controls.

normally need to do that here). And, you can temporarily mute any unwanted item. Notice that this mixer has a menu.

If you choose the Options command and then Properties, you will see a dialog box displayed similar to that shown in Figure 2.6.



**Figure 2.6** Windows XP Master Volume Properties.

You can use this dialog box to determine which items are shown in your soundcard's input and output mixers.

At this stage, you need not be too concerned about all these. Just be aware that they are there and that they control what inputs and outputs are available for use with your computer.

For example, if you find sometime that no sound is heard coming through a set of speakers or your headphones, it could be because that output has been disabled in your options.

## Sounds

Click now on the Sounds tab (see Figure 2.7). This is where you can specify your Windows sounds scheme.

It is strongly advisable to select No Sounds. The last thing you want is for Windows to sound off with the sound of a chirping bird (or whatever) as it starts up some background program while you're in the middle of recording a track.



**Figure 2.7** Windows XP Sounds and Audio Devices Properties: Sounds tab.

## **Hardware**

The Hardware tab gives you access to information about the status of your soundcard. In particular you can check to see which version of the card's drivers you are using.

You can consult the manufacturer's website to see whether there are more up-to-date drivers available for your particular card. If so, you should consider downloading and installing them.

If installing new drivers ever produces any problems or conflicts, you can always visit the Settings page shown in Figure 2.8 and roll your system back to its previously installed drivers. To display this page, first click on the Hardware tab, then select the device, then click on Properties.

## **Other Control Software**

It is likely (but not always the case) that there is additional software for controlling and managing your soundcard. This usually takes the form of some program that is supplied with the soundcard and installed on your PC when the soundcard itself is installed. If this exists, it is likely to be accessed from the Windows Control Panel.

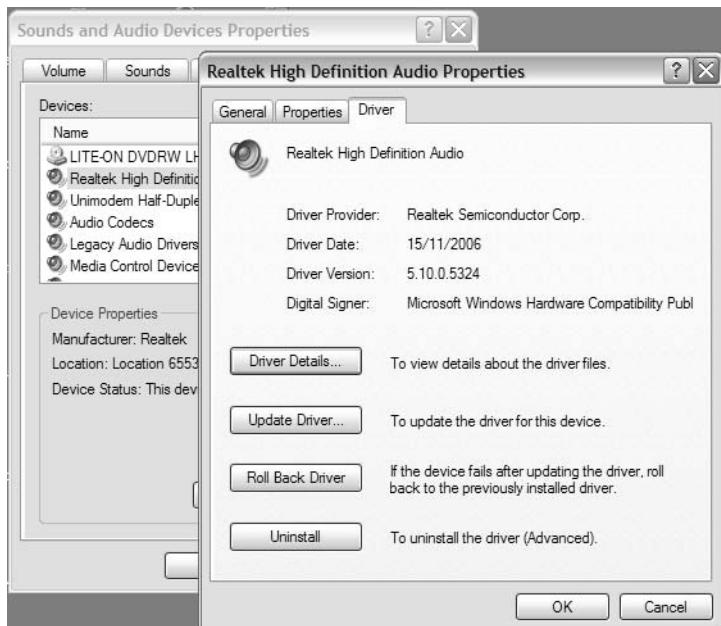


Figure 2.8 Windows XP Sounds and Audio Devices Properties: Hardware drivers.

In the case of inexpensive consumer-standard soundcards, this software often provides much the same functionality as the Windows Control Panel, but with perhaps some additional functionality.

As you move up the ladder to professional-standard soundcards, this additional functionality becomes both substantial and important. I'll say more about that later in this book.

Meanwhile, Figure 2.9 shows one of the settings pages of the control software that comes with one example of a Realtek consumer-standard soundcard.

### Understanding Your Soundcard: Vista

This section shows some of the equivalent screens if you are using Vista rather than XP (see Figures 2.10 through 2.14). They are accessed by selecting Sound from the Vista Control Panel. By and large, they provide a similar functionality to their XP counterparts and should need little, if any, further explanation.

In particular, you should familiarize yourself with the various Recording (Microphone and Line In) settings.



Figure 2.9 An example of control software that may be supplied with your soundcard.



Figure 2.10 Vista Sound settings: Playback.

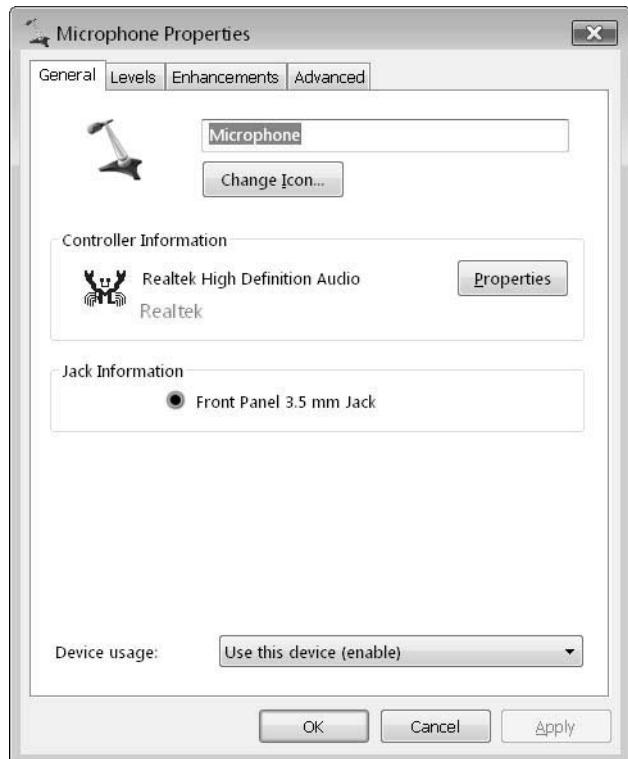


Figure 2.11 Vista Sound settings: Microphone Properties.

As with XP, you may find it preferable to use both the Windows mixer and the soundcard's control software (usually accessible from the Windows Control Panel) to get the settings you want.

Notice that in many cases you will be able to use the soundcard's controls to adjust, in particular, the volume of any incoming audio that is being recorded.

You may need to refer back to this when we start our lessons on recording.

## Assignment

1. Either insert an audio CD into your PC's CD drive or browse Windows Explorer and select an MP3 file.
2. Open either the CD or the MP3 file with your media player of choice (for example, Windows Media Player). The music should now play.

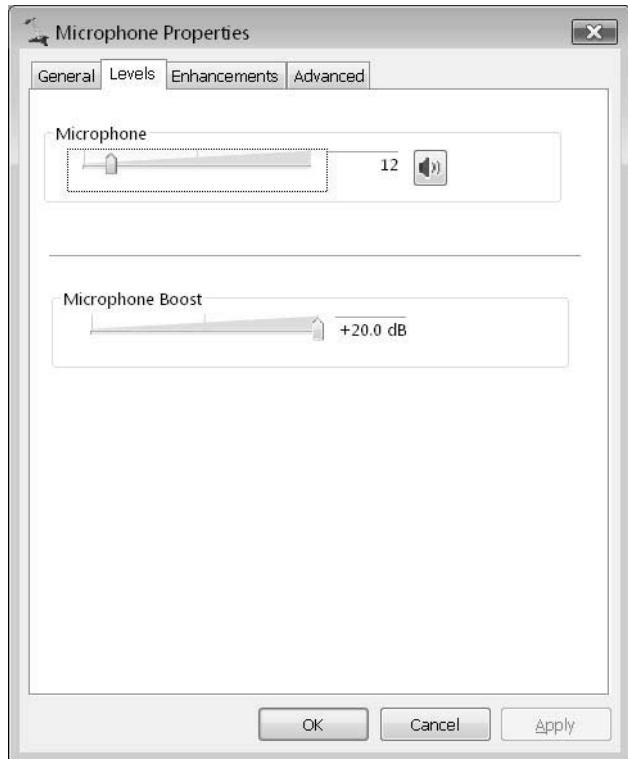


Figure 2.12 Vista Sound settings: Microphone Properties, Levels tab.

3. Open the volume control for your soundcard and experiment a little. For example:
  - Try muting and unmuting in succession each of the various sound sources. Which one causes Windows Media Player to be muted? That is clearly the one that controls the signal going from the soundcard to your speakers or headphones (or both).
  - Try adjusting the volume and balance controls for this item. What happens?
  - Try also adjusting the volume (and any other controls) in your media player. What happens?
  - Make sure when you are finished that before you close the Sound and Audio Devices Properties window, your settings are such that music can be heard when an MP3 file or audio CD is played.



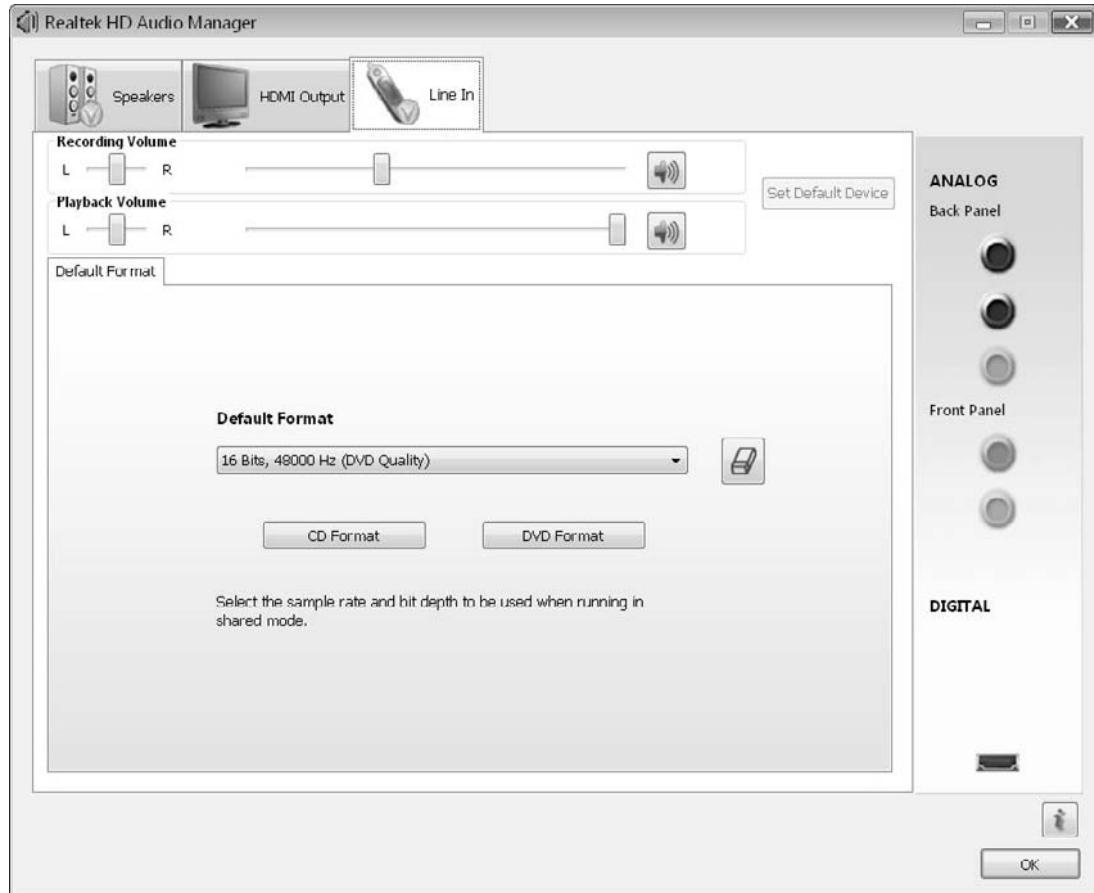
Figure 2.13 Vista Sound settings: Microphone Properties, Advanced tab.

You have just demonstrated the signal flow that takes place when any piece of software (in this case, your media player) is used to produce sound (see Figure 2.15).

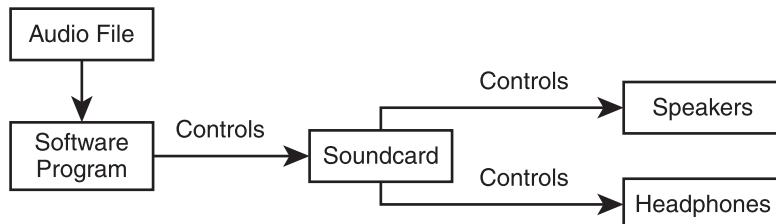
### Post Script

The main purpose of this lesson has been to help you develop some understanding of your soundcard, its characteristics, and its functions. You'll probably be pleased to know, however, that once you have installed and set up your recording program correctly, you should have very little need during this foundation course for direct interaction with your soundcard. The recording software will handle these tasks for you and in the background.

Nevertheless, it is important that you have at least this much understanding of your soundcard. If and when the time comes for you to upgrade to a professional soundcard, this becomes even more important. For example, you will need to tweak its controls and adjust its settings from time to time. For this reason, there is more information related to soundcards in Chapters 3 and 4 of this book.



**Figure 2.14** Example of a soundcard's control software for Vista.



**Figure 2.15** Audio flow of sound generated by your computer.

## Day 3: First Steps in REAPER

Before you begin, you will need to install the REAPER program. The CD in the back of this book includes the installer for the fully featured, uncrippled, and unexpiring REAPER version 2.5. Upon installing the program, you are entitled to leave it on

your computer and use it free of charge for up to 30 days. After that, you are required to purchase a license if you then wish to continue using it. Currently, a noncommercial license costs \$50, and a commercial license costs \$225. Instructions for how to obtain such a license are included with the software and also in the last chapter of this book.

It is very likely that by now a more recently updated version of REAPER may be available from the company's website. However, for the purposes of this course, it is strongly recommended that you use the version supplied with this book. It is the version around which this course has been designed. If you later decide to purchase a REAPER license, you will at that time also be able to upgrade to the latest version.

## Installing REAPER

To install REAPER on a PC, follow this sequence:

1. Insert the CD supplied with this book into your PC's CD drive and wait a few seconds.
2. Open Windows Explorer and navigate to display the contents of the CD.
3. Double-click on the REAPER folder to open it.
4. Double-click on the reaper25-install.exe file.
5. Follow the onscreen instructions to install the program, accepting all the default settings. At the end of the install process, you will be asked whether you wish to run REAPER now. Difficult as it is to resist the temptation, please answer No.

REAPER will be installed to a folder called C:\Program Files\REAPER\. The executable program file will be reaper.exe. Please note that nothing is written to your Windows registry as a result of installing REAPER.

At the time of writing this, distribution arrangements for the OS X version of REAPER have not yet been finalized. Go to [www.cockos.com/reaper](http://www.cockos.com/reaper) for the latest available information about installing REAPER on a Mac.

## Installing the Sample Files

On the CD supplied with this book, there is a folder called Sample Files that contains a number of subfolders, each of which has material used in this course. Copy the entire Sample Files folder to your hard drive, so that the material is stored in C:\Sample Files. During the exercises that follow, when instructed to open any of these files you should always open the file from your C:\Sample Files directory, not directly from the CD.

## Installing the Documentation

The REAPER folder of the CD supplied with this book includes a PDF file. This is an evaluation copy of the official REAPER User Guide. If you copy this file into the folder C:\Program Files\REAPER, then you will find that it can be opened directly from REAPER's Help menu.

You should not need to refer to this guide during this 30-day course. However, if you do decide to continue to use REAPER beyond this course, you will find this documentation invaluable. Please note that if you do this, there is a \$5 authorization fee. More information about this can be found on the first page of the document.

## Setting Up REAPER

Before you can use REAPER to play or record your music, you need to set it up to enable it to communicate with your soundcard and, in particular, with your soundcard's drivers (see Figure 2.16).

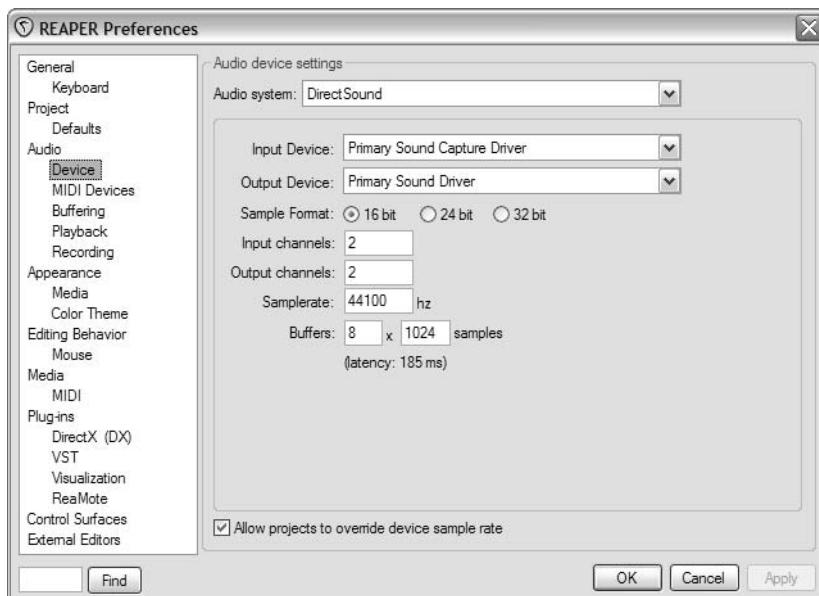


Figure 2.16 Specifying your Audio Device Preferences for REAPER.

If you are using a consumer-standard soundcard (such as the Realtek) that is supplied as customary with most computers, then you will need to select its DirectSound drivers. If, on the other hand, you have a professional-standard device (such as an M-Audio soundcard or USB or FireWire interface), then you will get better performance using its ASIO

drivers. If you are not sure which to do, for the purpose of this course, the safest option is to select DirectSound for the time being. I will have a lot more to say about drivers and soundcards in Chapters 3 and 4.

Follow this sequence:

1. Start the REAPER program by double-clicking on your desktop shortcut.
2. A sample project file might be opened and displayed. Please ignore it for now.
3. From the main menu, choose the Options command, then Preferences.
4. Navigate to the Audio Device page (refer to Figure 2.16).
5. You need to display the drop-down list of driver types in the Audio System field. For most consumer-standard onboard soundcards, the safest option at this stage is usually DirectSound.
6. You also need to specify your PC's input and output devices. Most likely, these will be similar to those shown in Figure 2.16, but if in doubt you should consult the soundcard's documentation.
7. Leave all other settings at their defaults. (These may not be the same as in the example shown, but that does not matter.)
8. Click OK to close the Audio Device Properties window.

## Your First REAPER Lesson

In this first REAPER lesson, we will focus only on understanding the very basics of navigating and controlling the REAPER interface. In fact, we could likely spend two or three lessons on navigation alone, although you'd probably find just navigating pretty boring after a while. For this reason, we will focus only on some very basic tasks and techniques in this lesson. And pretty soon, you're going to get sick of seeing this next paragraph.

*The subject of digital home recording is absolutely massive. There is almost no limit to the amount of things that you can learn within REAPER alone. That's the pitfall. Many people get frustrated and fail to progress because they go off in all sorts of directions, exploring all sorts of highways and byways, trying to learn and do too much at once. They then get lost and confused and don't know what to do to get back on track. You will get the most out of the course if you impose on yourself the discipline of staying within the confines of each lesson.*

In this lesson, you will learn how to:

- Open and close a project file.
- Play and stop a project file.
- Perform basic navigation through a project file.
- Zoom in and out.

Being able to navigate confidently and use the basic track controls are skills that are absolutely crucial to being able to work comfortably within REAPER. Stay focused and resist, if you can, the temptation to wander off exploring all over the place. For one lesson, this is quite a lot to learn.

## Introducing REAPER

If REAPER isn't already open, start the program by double-clicking on the REAPER icon on your desktop. Now follow this sequence:

1. It's possible a project file might be automatically opened. You can ignore that and instead open one of the course sample files that you copied from the CD. The method follows the standard Windows procedure.
2. Choose the File command from the main menu, then select Open Project. When the Choose Project to Open dialog box appears, navigate to C:\Sample Files \RosesBloom\, click on RosesBloom.RPP, then click on Open. If you are asked whether you wish to save anything first, answer No. Your file should now be opened.
3. To make sure the output is set to stereo, hover your mouse in the area labeled Master (near the bottom left of your screen) over the Output button. This is immediately below the S button and immediately above the IO button. If the tool tip displays Output:mono (L+R), click the button once. The tool tip will change to Output:stereo. Choose the File, Save command to save this change.
4. Choose the File, Save As command and save this file to a new name, such as RosesBloom Demo. Now you know you can't do any damage to the original!
5. Your screen should now be similar (but probably not identical) to that shown in Figure 2.17.

There's so much on the screen to take in that it's difficult to know where to begin. Stay focused now! First, you can see that the project is made up of a number of tracks. In fact, this project has six tracks in all. It does not matter if right now you cannot see all six. You should certainly be able to see at least the first two.



**Figure 2.17** The REAPER screen.

On the left is the Track Control Panel (see Figure 2.18). This stores information about the track (such as its number and name) and is also used to control the track's behavior in a number of ways. We'll return to this in later lessons.



**Figure 2.18** The Timeline, Media Items, and the Track Control Panel

To the right are the media items for each track. In this example, one media item is shown for each track. The media items contain the actual recorded material.

You can also see that above the first track there is a Timeline. This indicates the length of the media items (in this case, in minutes and seconds). Just before the 46-second mark you can see a dark vertical line. This is the edit cursor. It indicates the position at which the song was stopped when last played. It's possible that your Timeline might be displayed in units other than minutes and seconds (such as measures or beats). This does not matter for now.

**Playing a Project File.** Now cast your eye a little farther down, and you will see a Transport Bar (see Figure 2.19). You'll probably recognize this as being in some ways similar to controls that you have seen in Media Player or other software. Hover your mouse over any of the controls on the Transport Bar, and you will see a tooltip indicating its function.



Figure 2.19 The REAPER Transport Bar.

From left to right, the first five controls are Go to Start, Play, Pause, Stop, and Go to End. Each of these controls can be operated by clicking the mouse over the button or by pressing its corresponding keyboard shortcut when the REAPER window has focus. These shortcuts are also indicated in the tooltips.

Go to Start	W
Play	Space
Pause	Ctrl+Space
Stop	Space
Go to End	End

Click on the Play button or press the spacebar. The song should now start playing, and you should hear it. You should also see a play cursor moving across the media items from left to right. After a few seconds, press the spacebar again. The music will stop, and the play cursor will return to the same position as the edit cursor.

If you didn't get any sound, check to be sure your speakers are switched on and that your speakers and/or headphones are plugged in. Check any physical volume controls

on the speakers and/or headphones. Also check to make sure your Windows soundcard settings are okay.

See whether you get any sound using Windows Media Player with a CD or MP3 file. If not, the problem probably doesn't lie with REAPER. If you do, check your REAPER Audio Device settings. Open the Preferences, Audio, Device settings page and try selecting a different set of audio system drivers. Click OK to close the settings window, and try again.

### ***Navigating the Project***

As you progress with this course, it's important that you are able to navigate your way around a project. REAPER offers many smart ways of doing this, but for now let's come to grips with just the basics:

- Pressing W will move the play cursor to the start of the project.
- Pressing End will move the play cursor to the end of the project.
- Clicking your mouse at any position on the Timeline will move the edit cursor to that position. If the project is being played when you do this, the play cursor will jump to this position when you press the spacebar or click on Stop.
- If you accidentally click and drag your mouse when doing this instead of making a clean click, you will accidentally create something called a *loop*. This will cause the same section to play over and over again. Pressing Esc will clear this. (We'll look at loops more in a later lesson.)
- To scroll up and down a project, use the vertical scrollbar or use Ctrl+Alt with your mouse wheel.
- To scroll a project from left to right, use the horizontal scrollbar or use Alt with your mouse wheel.

Now let's try some examples.

1. Press W to return the cursor to the start of the song.
2. Press the spacebar to play the song.
3. After 30 seconds, press the spacebar again. The song will stop, and the cursor will return to the beginning.
4. Press the spacebar again. After a minute, press Ctrl+spacebar. Playback will pause, and the play cursor will remain at the minute mark.

5. Click on the Play button on the Transport Bar to recommence playback.
6. Click your mouse on the Timeline at about the 45-second mark.
7. Just before playback is due to end, press the spacebar. The play cursor will return to the 45-second mark.

### ***Zooming In and Out***

Often when you're working with REAPER, you'll need to zoom in closely on a track, a time segment, or both. Other times you will need to view the bigger picture. Again, there is a heap of clever tricks for doing this, and again, we are going to focus on just the basics.

- Press + to zoom in horizontally and – to zoom out horizontally. Press either key repeatedly to zoom quickly.
- Press Page Up to zoom in vertically and Page Down to zoom out vertically. Press either key repeatedly to zoom quickly. Alternatively, you can hold the Ctrl key while scrolling the mouse wheel up and down.
- Press Ctrl+Page Down to zoom to full project length.
- To toggle any track between three levels of vertical zoom, first select the track in the Track Control Panel, then double-click on the vertical scrollbar. This toggles between three states—minimized height, normal height, and full height.
- Press End to move the play cursor to the end of the project.

Now let's try some examples.

1. Press Page Up five times, then press Page Down five times.
2. Press + six times, then press Ctrl+Page Down.
3. Select Track 3 in the Track Control Panel, then double-click on the vertical scrollbar four times.

### ***The Mixer***

There has already been a lot to absorb in this lesson. Don't worry, the others will not all be this intense. Before closing, though, we'll just have a quick look at the mixer (see Figure 2.20). We're doing this because you must have seen it on your screen, and you might be wondering what's going on.

Your mixer should look similar to that shown in Figure 2.20, but it may not be identical. Any differences will be cosmetic and do not matter.



Figure 2.20 The REAPER mixer.

Each of the tracks in our project will be displayed in the mixer as well as in the tracks area. The mixer shows essentially the same information as the Track Control Panel, but in a different way.

When you play your project, the flashing bars of light in each track represent the volume for that track. There is also another item called the MASTER. In Figure 2.20, this is shown on the right. Yours may be on the left or on the right—at this stage, it doesn't matter which. The flashing bars of light in the MASTER represent the total volume of all the tracks combined.

## Assignment

Practice, practice, practice!

Practice playing and stopping, playing and pausing, and zooming and navigating until you are confident that you're getting on top of all these controls.

Take a break for a half hour, then go back and do it again. It's not the most exciting part of the course, but it is laying down solid foundations. You'll find the sections that follow a lot easier to master if you don't have to keep stopping and thinking about these very basic skills.

Now that you've had a taste of working with a REAPER project file, in our next lesson we'll take a look behind the scenes and discover the various files that go into it.

## Day 4: Project and File Basics

If you've had almost any experience at all working with Windows, you'll already know that application data is stored in files, and that files are grouped together and organized in folders (sometimes called *directories*). For example, Word documents are stored as .doc files. Excel spreadsheets are .xls files. The documentation that comes with many programs is often stored as a .pdf file. These are just three of countless examples.

Any application that you use for recording also stores its application data in files. One important difference between recording software and most other types of applications is that you never get just one file for a recording project. *A recording project always consists of not one file, but a series of files of different types that belong together.* This entire set of files makes up the project.

Different applications use different extensions for their file names. Moreover, some programs may also use certain file types that others do not. For example, REAPER provides the option to create for each project a .undo file to store the project's history. Most recording applications do not have this facility.

In this section, we will be focusing on the main file types that are common to most or all recording applications. Just to confuse you some more, however, there is no universal standard terminology used by the different competing programs. For example, a peak file in REAPER serves the same purpose as a picture file in SONAR. Once you understand the key concepts, this shouldn't cause you too much confusion.

### Examples of File Types Used by Projects

Here are some examples of the main file types that are used to store data for almost any recording software program.

#### ***Project Files***

This is the file that you open from the File, Open menu command. It holds the project structure, such as how many tracks the project contains and what each track is called. It also stores such information as the various track settings when you last saved the file. These settings include, for example, the volume level of each track.

Note that the project file does not itself contain any recorded material. However, it does include the information that your DAW software needs in order to know which audio files are to be associated with which tracks.

In REAPER, project files have a .RPP extension.

### ***Backup Files***

Most DAW software offers you a number of options for automatically creating backups of your project files. For example, you might want a backup created automatically when a file is saved or at regular time intervals.

In REAPER, backup files have a .RPP-bak extension.

### ***Media Files***

These are the files that store the actual audio or MIDI data. The extension used in this case is dependent not on which DAW software you are using, but on your choice of file format. I'll say more about different audio formats and their characteristics later, but for now you surely will have heard of at least some of these—for example, MP3 is a format widely used for distributing audio via the Internet. As a general rule, the more tracks you have, the more media files you will have. A typical project will include a number—perhaps a large number—of media files.

There are numerous types of media files, including .aiff, .flac, .mp3, .ogg, and .wav (all audio) and .mid (MIDI). These formats are shared by different DAW programs. For example, REAPER, Cubase, and SONAR can all use .wav media files.

### ***Peak Files***

Peak files are the files that visually represent the sound waves produced by an audio media file. Each audio media file will have a peak file associated with it. MIDI files do not have associated peak files because they do not contain actual recorded sounds.

In REAPER, peak files have a .reapeak extension.

Figures 2.21 through 2.33 may help you understand how these various files fit together. Figure 2.21 shows a REAPER media file that does not yet contain any media or peak files.

In Figure 2.22, our project file now has six media items. For each media item, there will be an associated media file containing the audio material. In this example they are .ogg files, but they equally could be .mp3, .wav, or some other format.

In Figure 2.23, REAPER has created a peak file for each media item. The peak file is a visual representation of the audio.

Let's now see what happens if we use Windows Explorer to look inside the project folder (see Figure 2.24).

You should be able to identify the different file types you can see here and the purpose that each type serves. These files really belong together as a set.



**Figure 2.21** A REAPER project file with no media items.

The peak files on their own are of virtually no use whatsoever. With most audio applications, these are automatically created for you when you record and modify your media files. The project file on its own will give you a lot of information about the contents of the project, but by itself it cannot reproduce any sounds or play your recordings. Each media file by itself is of very limited use. For example, you could open any one of them in another program and edit or play it. But they are only of any real benefit to you in combination with each other, the peak files, and the project file.

It's therefore important that you keep these files together in the same folder. If you start moving them around or deleting them inside Windows Explorer, you are likely to get a message warning you that some files are missing when you open the project file.

## Audio File Types

We have already said that there are several different audio file types. You would normally expect to work with .wav (PC) or .aiff (Mac) files if you were intending to distribute your music via audio CD, or .mp3 files if you were planning to distribute the output over the Internet. This is in part due to technical limitations, but it also makes practical sense. The .wav and .aiff formats give you a more accurate and better-quality recording but require a large file size. On the other hand, .mp3 (and various other formats) require a much



Figure 2.22 A REAPER project file with media items but no peak files.

smaller file size, but they do so at the cost of compromising on fidelity. One common practice is to record in .wav or .aiff format to ensure the best quality recording, but then to render the final output down to .mp3 for distribution over the Internet.

If your curiosity is easily satisfied, that's the bare minimum you need to know for now about different file formats. However, it's a pretty interesting subject, and as a spinoff you can learn a fair bit about audio by learning more about the characteristics of different file formats.

So, here's your assignment.

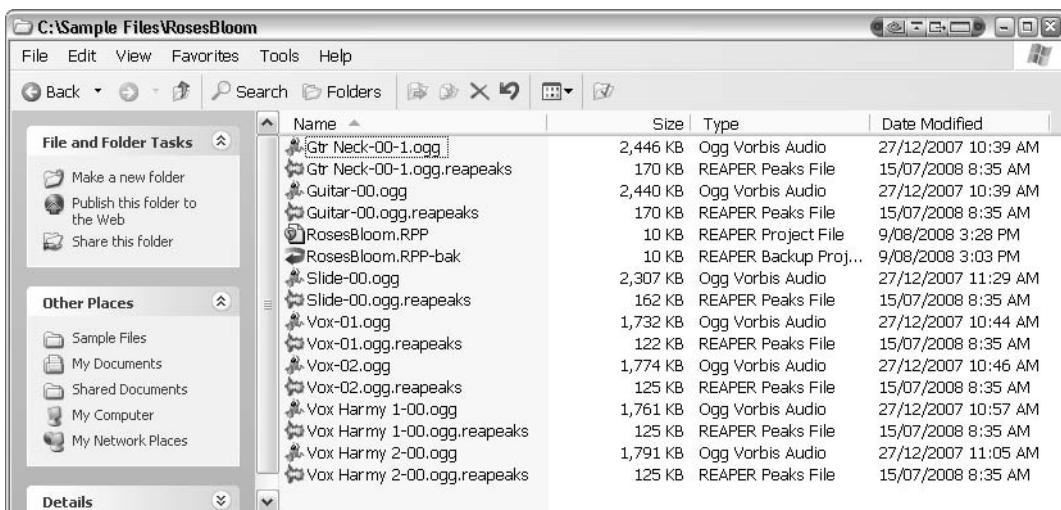
## Assignment

Jump onto the Internet. Without getting bogged down in the minutia of technobabble, find out:

1. Why is .wav or .aiff format used for recordings that are to be distributed by audio CD, whereas .mp3 is preferred for distribution over the Internet?
2. What is the basic difference between lossy and lossless audio compression?
3. What are the main characteristics of the .ogg and .flac formats?



**Figure 2.23** A REAPER project file with media items and peaks.



**Figure 2.24** Inside a typical REAPER project folder.

If you find yourself getting interested, dig deeper. If you're getting bored or confused, pull back. You don't need an in-depth knowledge of this subject, just a general understanding.

## Day 5: First Steps in Recording

At last you're ready to start some recording! Well, almost ready.

Hopefully, after completing the assignment at the end of yesterday's lesson, you now have some understanding of at least some of the different audio formats. It is suggested for this course that you record in .wav format. Then, at the end, you can render your recordings twice, in both .wav and .mp3 format. That way, you will be able to burn your music to audio CD and upload it to the Internet or copy it onto an MP3 player if you wish.

We just need to explore a little more theory in this area before we can start. Don't worry, it is only a little more.

### Audio Format Settings

No matter which audio format you select, you will also need to specify a number of settings. These settings determine primarily the technical quality and fidelity of the recording. Figure 2.25 shows the .wav format Audio Settings page for a REAPER project.

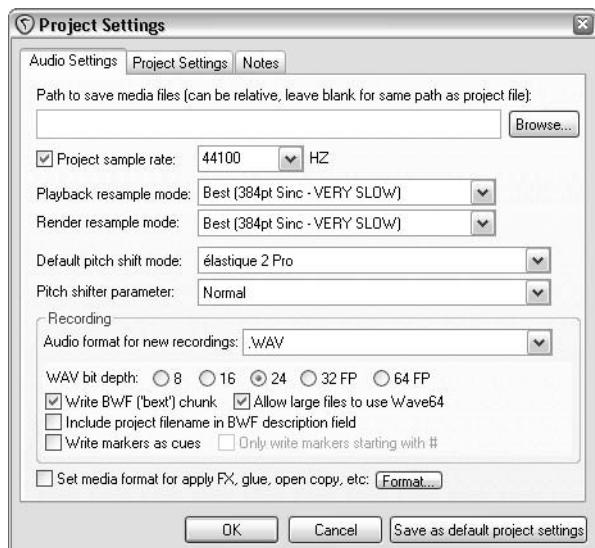


Figure 2.25 Typical .wav format audio settings for a REAPER project.

On this page there are two main items that are important—project sample rate and bit depth. For the time being, forget all the other options except these two.

Sample rate can, in theory, be anything between 8,000 and 192,000. The range of values you are likely to wish to use in practice, however, is considerably more restrictive than that.

As with most such things, the higher the number, the better the quality, and the lower the number, the poorer the quality—but less use is made of your computer's resources. Anything below 44,100 won't give you CD quality, and if you're using a consumer-standard soundcard on a normal desktop PC, chances are that anything much higher than that will impose too much of a burden on your PC. Indeed, the card might not be capable of handling the higher sample rates.

If you know something about sound frequencies, you might also be interested to know that the sample rate also determines the highest frequency you will be able to record. Simply halve the sample rate. This means, for example, that at a sample rate of 44,100, you will be able to record frequencies up to 22,050 Hz, well above the limits of human hearing.

The question of bit depth is more difficult to explain. The short explanation is that the higher the bit depth, the more accurate the representation of the sound that is being recorded. It's all to do with the way in which the computer converts your music into that string of 0s and 1s. For this reason, 24-bit recording is universally preferred to 16-bit—it spans a range of over 16 million bits to ensure greater precision as compared to just over 65 thousand. There are other advantages to 24-bit recording, too. It offers you more ease and flexibility in managing recording and output levels. However, if you have a consumer-standard soundcard, it might not be capable of recording at 24-bit. This is especially likely to be the case if your PC is a few years old.

When you commence the examples on the following pages, try selecting .wav format, a sample rate of 44,100, and a bit depth of 24-bit. If you find that your setup cannot handle this, change the bit-depth setting to 16-bit. For the purposes of this course, 16-bit audio offers perfectly acceptable quality.

## A Simple Recording Example

In this very first example, you will create a very simple recording of your voice, play it back, and save it. Then we'll go on to take the first steps in a rather more ambitious recording project.

We'll be using "Happy Birthday," a simple song that everybody knows and that you can sing unaccompanied for about 45 seconds to a minute. Before we begin, however, let's just pause to consider some of the issues that you will have to look out for.

- **Creating a project file and creating and arming tracks.** You will need to create a project file for your recording, together with as many tracks as you wish to record. These tracks will need to be armed and supplied with various items of information, such as which input is being used for the recording.
- **Setting a recording level.** Before you record, you need to make sure that the strength (volume) of the incoming audio signal is adjusted to an appropriate level.

Let's see how it's done. Follow this example step by step.

1. Make sure that your microphone is plugged into your soundcard.
2. Start REAPER. We are going to create a new file for the song. Choose the File, New Project command from the menu.
3. If the Project Settings window isn't automatically displayed, choose the File, Project Settings command from the main menu.
4. Select the audio format WAV, a sample rate of 44,100, and a bit depth of 24, as shown in Figure 2.25. Click on OK. If you later find that 24-bit is too ambitious, you can at that time change it to 16.
5. Save the project file straight away. Choose the File, Save Project command (or press Ctrl+S). Navigate to the area of your hard disk where you want to store the song, then type a name (as shown in Figure 2.26). Make sure you select the Create Subdirectory for Project and Move All Media into Project Directory options. This will make it easier for you to keep track of your work.

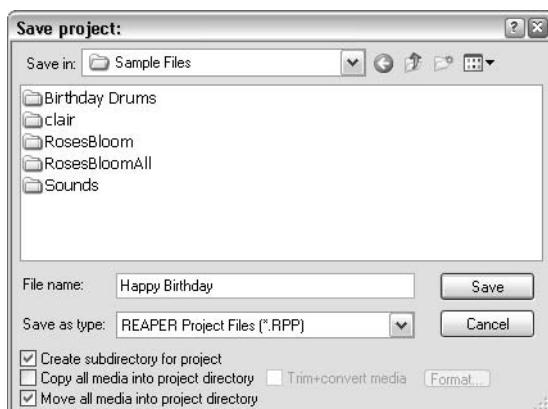


Figure 2.26 Saving a project file with its media items.

6. The next task is to create a track for your project. REAPER offers several ways of doing this; use whichever method you prefer. You can press Ctrl+T or double-click anywhere on the empty part of the Track Control Panel. If you forget these shortcuts, choose Insert, Track from the main menu.
7. You should name the track for recording. To do this, double-click in the blank area to the immediate right of the track number, type the name (such as Vocal), and press Enter.
8. Now press Ctrl+S to save the file. Get used to doing this often—I won't keep reminding you!

The next step is to arm the track and select the input you wish to use for recording. This is where your microphone is connected to the soundcard. Before doing this, read the next few paragraphs. This will make it easier for you to understand what you are doing.

For most onboard consumer-standard soundcards, this will be a stereo input, receiving the audio signal on two channels (left and right). It will only be necessary to record on one of these channels. The procedure is to display the drop-down list of available inputs and select a single mono input. In Figure 2.27, the Left input has been selected. Note that recording with a single mono input will not in any way preclude you from later creating a stereo mix.

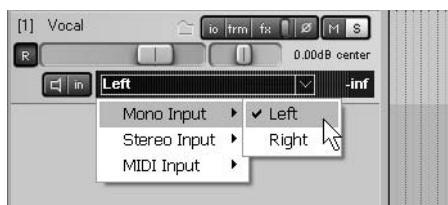


Figure 2.27 Assigning an input for recording.

When you come to record, the level of the incoming signal will be affected primarily by three factors:

- The record input volume level that you previously set with the soundcard or Windows mixer
- How loudly you sing
- How close your mouth is to the microphone

When you record, the signal level will be displayed on the track's VU meter. You will be aiming to peak at about -10 dB. Under no circumstances should you allow a recorded signal to go as high as 0.0 dB. Your aim is to capture a loud enough signal without clipping. Figure 2.28 shows an example.



Figure 2.28 Recording without clipping.

Now it's time to return to the exercise and carry on where we left off. Don't be disappointed if you don't get it right the first time. After recording a track, if at any time you find that you don't like it, you can select the media item (by clicking on it), then use the Delete key to remove it. This will enable you to start all over again. That said, if this is your first attempt at recording, you do not need to be too fussy about the quality of the outcome. You'll be getting plenty more practice!

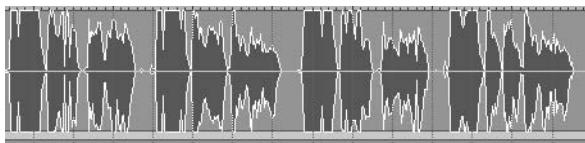
9. To arm the track, click on the button labeled R (just below the track number) in the Track Control Panel. This button will become red. Like many such buttons, it is a *toggle*. After recording, you will need to click on this button again to disarm the track.
10. Now assign an audio input for this track. Display the drop-down list of available inputs (as shown in Figure 2.27) and select a single mono input, preferably Left.
11. Now test your sound levels by singing into the microphone. Start with the microphone about 5 or 6 inches from your mouth. It might take a little while to get it right. Perhaps counter-intuitively, the signal level will not in any way be affected by the track's own volume fader. This is because the track's volume fader only controls the output of an audio signal, not the input.
12. When it seems that you are ready, have a go at recording. To commence recording, press Ctrl+R. (You could instead click on the Record button on the Transport Bar. This is the red button to the right of the Go to End button.)

However, keyboard shortcuts are often quicker and easier.) As the track is being recorded, REAPER will build and display the media peaks for this track.

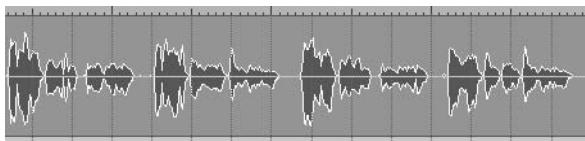
13. To stop the recording process when you have finished, either press the spacebar or click on the Stop button. If you are asked whether you wish to save or delete your work, choose Save All. Click on the track's R button to disarm it.
14. Zoom out so you can see the entire recorded track. Play back your recording. Of course, it will not be anything like a professional-quality recording, as much as anything because of the limitations of the equipment you are using.

## Waveform Patterns

You can learn quite a lot about a recording simply by studying waveform patterns. Take a look at the three examples in Figures 2.29 through 2.31.



**Figure 2.29** A wave file recorded too hot.



**Figure 2.30** A wave file recorded about right.



**Figure 2.31** A wave file recorded too low.

Notice in Figure 2.29 that the waveforms go all the way to the top, almost as if they had been trimmed like a hedge. They look like they are trying to climb out of the peak file altogether! In this case, it is almost certain that *clipping* has occurred. This happens when a signal is recorded too loud and goes up to and over 0.0 dB. The resulting sound is quite unpleasant. Almost certainly, this media item is unusable. In a case like this, it is often said that the signal was too hot.

Figure 2.30 looks a lot more promising. The recording is a fairly strong one with clear dynamics (peaks and troughs). Definitely no clipping occurs. It is quite likely to be usable.

The example in Figure 2.31 has been recorded much more softly. You can see that it has a very severely limited dynamic range. On the face of it, it appears to have probably been recorded too quietly, but without listening to it you can't be sure. Perhaps it is an instrument that is meant to be played softly. It might be usable, or it might be advisable to record it again.

As a general rule, it is better to risk recording at too low a volume than too high. A low volume can often be lifted afterward, but a recording that has been damaged by clipping cannot be repaired.

## Assignment: A Recording Experiment

Even though you're using budget equipment, there's still a lot you can learn here. Start asking yourself questions like these:

- What difference does it make to the shape, clarity, and timbre of the sound if I move the microphone closer to my mouth or farther away?
- What difference does it make if I approach the microphone from a different angle or perspective—for example, singing directly into the microphone or slightly over the top?

Try this experiment:

Record this song several more times, experimenting with these and any other ideas that may occur to you. In each case, create a new track before recording. Don't forget to disarm each track when you have finished recording it.

When you are playing back, you can use the S button on any track (in the Track Control Panel) to solo that track, so only that track is audible. Clicking on this button toggles this feature on and off.

Alternatively, you can click on the M button on any tracks to mute them, so they are not heard during playback. Again, clicking on this button toggles this feature on and off.

You can use the volume fader on the track and/or on the master to adjust the final volume.

When you have finished, don't forget to save and to back up all your work to an external hard drive, a flash drive, or a CD.

## Day 6: Second Steps in Recording

In this lesson, you will build on what you learned yesterday by beginning work on your first real recording project. Again, we'll use "Happy Birthday," but this time we'll be starting a new project from scratch. Before recording anything, plan out your project. Today we will be recording just one track. This will be a rhythm instrument, such as a guitar. However, there is no reason why you cannot map out your project.

### The Recording Project

The finished project should eventually require three or four tracks. If you like, you will be able to add as many extra tracks as you wish later, but let's not get overly ambitious with this first project. Decide on a simple arrangement. Keep the following in mind during your planning:

- You will need between one and three instrument tracks. These might include, for example, a guitar and a bass guitar with perhaps some simple percussion. You might find that the tabs shown in Figure 2.32 are helpful.
- You will need a main (lead) vocal track.

**Happy B-Day Tab**

C	G	G      F
Happy birthday to you		Happy birthday dear name
e -----		e -----3---0-----
B -----	1---0---	B -----1---0---
G 0--0--2---0-----		G 0--0-----2-

C	F      C	G    C
Happy birthday to you		Happy birthday to you
e -----	e ---1---0-----	
B -----3---1---	B -----1---3---1---	
G 0--0--2---0-----	G -----	

**BASS:**

G -----	52---332---
D ----32----	53---32---353
A 3353--	3353--33---5-----

**Figure 2.32** Tab charts for "Happy Birthday."

- Consider using a vocal harmony track.
- Remember that if you have only the basic setup described in this book, you will only be able to record one track at a time. You will be shown how to do this.
- The song should not be too long. Aim for three minutes, maximum.
- Plan to eventually record at least two verses and a chorus, and the arrangement should include an instrumental lead break of perhaps 20 or 30 seconds.

Remember! Use Ctrl+R each time when you are ready to start recording, and press the spacebar to stop. If you make mistakes, you can select any recorded item and press Delete to remove it.

In your assignment you will be recording just the first track for now. The others will come later.

## Assignment

In this assignment you will have the opportunity to consolidate what you have just learned by creating another project of your own.

1. Create a new project. Select .wav as the required audio format, then select your sample rate and bit depth settings.
2. Save the file in its own directory along with all media items. Make sure it is kept in its own folder, not the folder that you created yesterday. Name the file Happy Birthday 1.
3. Create a track for your first rhythm instrument. This should be an instrument that is played for the whole length of the song.
4. Give the track a suitable name and arm it for recording.
5. Experiment with microphone placement and your settings to get a sound you like.
6. If this is an acoustic guitar, there are a number of placements you can try. In each case, try a distance of 6 to 9 inches from the instrument. One possibility is directly in front of the sound hole. Another is a few inches away from the sound hole, toward the bottom and down a little. A third is a little way above the sound hole, toward the neck. There is no single best position that suits every possible guitar/microphone/musical genre combination.

7. If this is an electric guitar, try sending the output to an amp and recording through a microphone pointed at the amp. Start with a distance of about 3 inches from the amp and experiment with the position and angle.
8. It is usually a good idea to record a count-in before you start playing. Later, you will be shown how to remove this when you have finished recording.
9. When you have found the placement that you like best, record your tune. If you make a mistake, at this stage you will need to delete the media item and record the whole thing again.
10. Don't forget to disarm each track when you have finished recording it.
11. Later, you will be shown how to record over just a part of a track. To delete any unwanted takes after you have heard them, simply select the media item (left single mouse click) and press the Delete key on the keyboard.
12. When you have finished, save the file.
13. Create your other tracks, but do not yet attempt to record them.
14. Save the file. Your project should look *similar* (but not identical) to that shown in Figure 2.33.



Figure 2.33 A project with one track recorded and several others set up.

15. Close REAPER and back up all your work to at least one flash drive or external hard drive, preferably two.

## Day 7: Recording Additional Material

Today is a really big day, with a lot of work for you to do. I'm keeping Day 8 free for a consolidation exercise. If you need to, instead of doing the consolidation exercise, you can spread today's work over two days. This is probably the most difficult day of the course in terms at least of understanding what you're doing. If you find yourself struggling, stick with it. After this lesson, it really does get easier!

### Introduction to Layering

You're going to start really getting down to doing some serious recording today, using a technique known as *layering*. This essentially consists of listening to your previously recorded track through your headphones while you lay down a new track alongside it. In the first example that follows, you will listen to the guitar track you recorded earlier in the Happy Birthday 1 file while you record a vocal track.

Sounds straightforward enough? In many ways it is, but before you can begin, there are a few hugely important technical concepts that you will need to come to grips with. These include *input monitoring* and *latency*.

### Input Monitoring and Latency

*Please do not be tempted to jump ahead or undertake any of the Day 7 examples until you have read through this subsection carefully!* This subsection is absolutely critical to your understanding of home recording and digital audio. It will save you so much angst later.

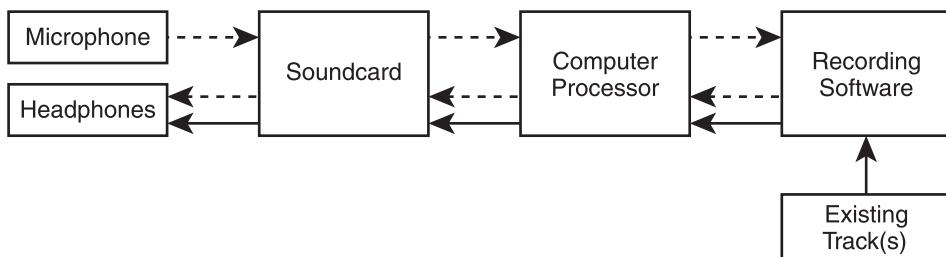
The term *input monitoring* refers to the practice of listening to yourself through headphones when you are recording. You set it up so that in your headphones, you can hear a mix of the previously recorded track (or tracks) along with the material you are currently recording—in this case, your vocals. This can be handled in one of two ways:

- By the software that you are using for recording (in this case REAPER).
- Directly by the soundcard's own control software. This is called *direct monitoring*.

The diagram in Figure 2.34 illustrates the signal flow when you are using your recording software to manage your input monitoring.

If you're really smart, you might be able to identify a problem here. Think about it for a moment. The recording of the guitar is played back and fed to your headphones. In the diagram, these are represented by the solid lines and arrows. You sing along with what you hear. The signal of your voice is fed all the way in through the soundcard and the computer's processor to your recording program. This then directs it back through the

### Input Monitoring with DAW Software Program



**Figure 2.34** Input monitoring using the DAW software's input monitoring facility.

computer and the soundcard and out to your headphones. This signal flow is represented by the dotted lines. *This process takes time*. It is unavoidable that there will be a delay between your singing and hearing your own voice. That delay is what we call *latency*. The critical factor is whether the length of that delay is acceptable.

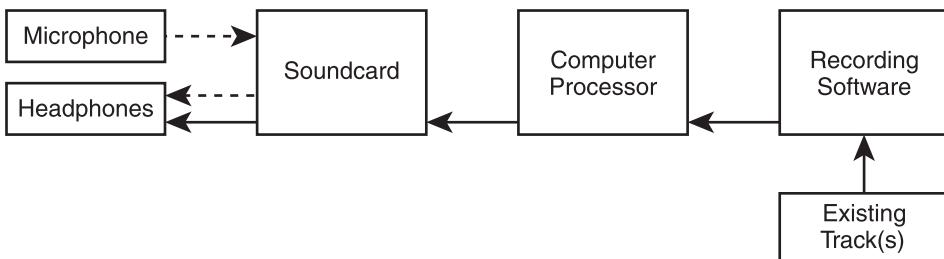
As a rule, if you notice it, it's not acceptable. And if it's much more than about 5 milliseconds (five one-thousandths of a second), you *will* notice it. We'll talk later in today's lesson about what you can do to minimize the delay (that's where buffers come in), but for the time being, let's just get our heads around the principal issue.

Now for the bad news: *It is almost guaranteed that you will not be able to successfully reduce latency to an acceptable amount if you are using an ordinary home computer and an inexpensive consumer soundcard*. You really need a professional-standard soundcard (costing probably several hundred dollars) and a PC with a more powerful CPU to have any real chance. Soon we'll give it a try, but don't be surprised if you find that for the purposes of this 30-day course, you'll have to get by without input monitoring. It's not ideal, but it can be done.

Let's then take a look at the alternative method—direct monitoring. Figure 2.35 shows the signal flow for direct monitoring. The diagram is an oversimplification, but it serves to illustrate the concept.

With direct monitoring, the signal coming into the soundcard from the microphone is fed directly back out from the soundcard (under the control of the computer) without needing to be processed first by the recording software. Of course, the signal is also fed to the recording program in order for it to be recorded, but that is a separate flow from the monitoring process.

### Direct Input Monitoring



**Figure 2.35** Input monitoring using the soundcard's direct monitoring facility,

It should be apparent that this method has obvious advantages over the other method. By cutting out some of the processes, you should be able to achieve better levels of latency. Why, then, do we not use this method all the time? There are two main reasons:

- Not all soundcards are capable of supporting direct monitoring. And yes, you guessed it: It is a feature you would expect to find on professional-standard soundcards, but not budget-priced consumer ones.
- Direct monitoring can only be used with audio, not MIDI recording. That is because if you are recording a stream of MIDI data from a keyboard, the data stream doesn't consist of any sound, only a series of instructions. (However, if your keyboard has its own speakers capable of generating sound from the MIDI data, you can, of course, use this.)

### Example

Thank you for your patience! Finally, we're ready to progress. Before we start recording, though, we are going to do a latency test. These instructions assume that you are using a budget consumer-standard soundcard with DirectSound drivers. If you are using a professional-standard soundcard with ASIO drivers, you will need to modify these instructions at the steps where this is indicated.

1. Open your project file Happy Birthday 1 and immediately save it as Happy Birthday 101 (which will enable you to keep a separate record of each lesson's work). Make sure your headphones and microphone are plugged into your soundcard correctly.
2. If for any reason the R (arm for recording) buttons are not visible on your tracks, press the Page Up key. This should expand the track height. If you expand it too much, Page Down reverses this.

3. Click on the R button for Track 2, your Lead Vocal track. This arms the track for recording. Do not record anything just yet.
4. Select the required input, then click on the In button and select the Monitor Input option (see Figure 2.36).



**Figure 2.36** Enabling the DAW's input monitoring facility.

5. Now speak slowly into the microphone, counting from one to ten. You should hear yourself in your headphones. The critical question is this: How long a delay is there between you speaking and you hearing yourself? For input monitoring to be used when recording, there really needs to be no *apparent* delay.
6. You can test this a little more by playing the song and singing along with the music for a while. The delay (latency) will probably be a real problem. (Incidentally, you can adjust the volume of the recorded music by adjusting its track volume fader. It might also be possible to adjust the level of the vocal signal going to your headphones if your soundcard's control software mixer allows for this.)
7. Note that *adjusting the level of the volume going to your headphones is not the same as adjusting the level of the volume being recorded*. You might wish to pause for a second to get your head around this. Remember that you have both an *input* audio signal stream and a separate *output* one.
8. For the time being, turn input monitoring off. This is done in the same way as you turned it on. To check what latency you are getting, choose the Options, Preferences command and select the Audio Device page. If you're using DirectSound, you should see something like what is shown in Figure 2.37.

Without getting too technical, the latency is determined by the number and size of the buffers that are used to pass the audio signal flow through the soundcard. We'll look at this in more detail in Chapter 3. For now, think of it as being like a conveyor belt moving an endless number of buckets, each containing a certain

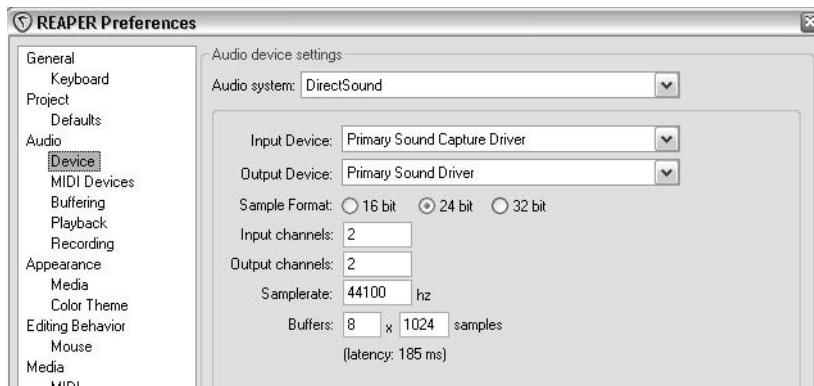


Figure 2.37 Checking your latency settings.

quantity of audio data. When it gets to the end of the conveyor belt, each bucket is tipped, and its contents wait in an orderly line to be processed.

The greater the number of buckets used (buffers) and the greater the size of each bucket (number of samples), the higher (worse) the latency. Conversely, using fewer and smaller buckets gives better latency, but requires a more sophisticated soundcard and a faster computer processor. Otherwise, you get a traffic jam that results in nasty pops and crackles.

In the example shown in Figure 2.37, the default latency setting is 185 ms—far too high for input monitoring. If you reduce the number and size of buffers (see Figure 2.38), you can reduce latency. In the example shown, we are trying to reduce latency to 11 ms. This is still a touch on the high side, but it might just be acceptable.

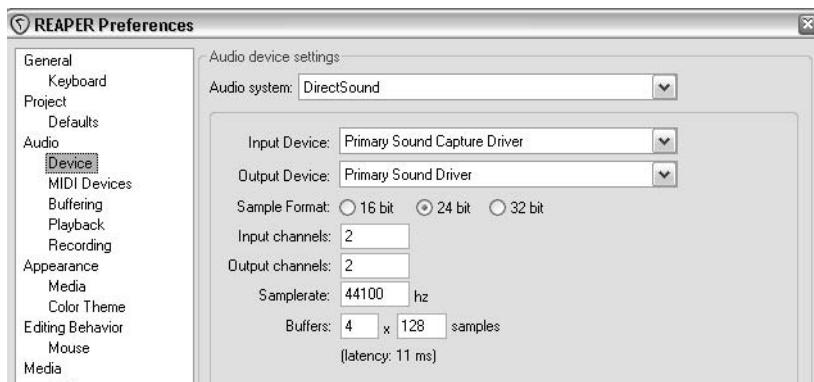


Figure 2.38 Changing your latency settings.

If you are using a professional-standard soundcard with ASIO drivers, the display will be very different from that shown here. REAPER's Audio Device Preferences page will include a button labeled ASIO Configuration. Clicking on this will open a window for the soundcard's own native control software. This is what you will need to use in order to change latency. You will need to consult the documentation that comes with your soundcard for exact instructions on how to do this.

The exact steps involved will be product-specific but will nevertheless be likely to follow the same principle as that used for modifying DirectSound settings. You will still need to make changes to the buffers and sample values to achieve the required effect.

Right now, let's get back to the instructions for working with DirectSound drivers.

9. Change the number of buffers to 4 and the number of samples to 128 (as shown in Figure 2.38). This will reduce latency to 11 ms. Click on OK to close the Preferences window. Turn input monitoring back on, and again count slowly from one to ten. If you are using an inexpensive soundcard with a basic home computer, you will almost certainly hear your voice along with lots of pops and crackles. This indicates that your system cannot handle this low level of latency.
10. If this is the case, turn input monitoring off again and experiment with adjusting your buffer settings to get latency as low as you can without experiencing crackles and pops when you turn the input monitoring back on. For this experiment, stick to exact multiples of 2 (for example, 2, 4, 6, 8) for buffers and powers of 2 (such as 16, 32, 64, 128, 256, and so on) for samples. For most domestic standard equipment, you are unlikely to achieve much better than a Buffers setting of 6 and a Samples setting of 512. This would give a latency value of 69 ms, which is still rather high.
11. You will now have to judge for yourself whether your latency level is acceptable for input monitoring. If it isn't, you can still record—you might just have to try a little harder and, above all, resist the temptation to shout rather than sing (because you won't be able to hear yourself too well when you're recording). *Remember that this course is all about understanding recording, not yet about getting top results.* If you're unable to use input monitoring, therefore, it should not put you at too much of a disadvantage.
12. At last we're ready to record. Well, almost. There's just one more thing we need to check first. Some consumer soundcards include an option called What You

Hear (or similar), which can be set on or off as one of its recording settings. If your card includes this feature, it should normally be set to Off, especially when you are using layered recording. Otherwise, you may find that the output of previously recorded tracks is recorded again, along with any new material that you are intending to record.

13. You should now be ready to record your lead vocal. Use the method that you used on Day 6, keeping in mind that with your headphones suitably adjusted you should be able to follow the existing guitar part. Don't forget to disarm the track when you have finished recording it.
14. Don't be too concerned if there is a passage in which you are not too happy with your singing. In the Day 9 lesson, you'll see how to fix this.
15. When you have finished, play the song. You should hear both guitar and vocal. Do not be concerned if the balance between vocal and guitar doesn't sound right (for example, one may dominate the other). We'll fix this later when we start mixing.
16. Don't forget to save your work frequently!

## Assignment

Record your two additional tracks in a similar way. When you have finished, save the song. Don't forget to back up all your work. This is the last time I will remind you!

## Day 8: Consolidation Exercise

After yesterday's heroics, you probably deserve a day at the beach or something. If you feel like you need a break for a day, take one. Otherwise, spend some time reviewing what you have learned in the last two lessons.

## Assignment

In this assignment you will have the opportunity to draw a breath and consolidate everything that you have learned up to now. Take it slowly and steadily. Be prepared if necessary to look back over the earlier lessons.

1. Decide on a simple, short, well-known song, such as a nursery rhyme, a pop standard, or a Christmas song.
2. Create a brand-new project file.
3. Name this Project 2 and ensure that it is saved in its own folder, together with all its media items.

4. Record, one at a time, between three and five tracks for this file. The arrangement can be any combination of instruments and vocals that you wish. If you like, it can be purely instrumental or *a cappella*.
5. Save your work and back it up!

## Day 9: Overdubbing

Today and tomorrow, we'll be looking at the different techniques you can use when you need to replace part of a recorded track—perhaps a verse, perhaps a line, perhaps just a phrase—with a new recording. You will be shown two ways of doing this. The first way is fairly straightforward but also rather inelegant. The second way (which you will discover tomorrow) is a lot smarter, but it takes a little longer to understand. First, however, a word about undo.

### The Undo Safety Net

Now that we're starting to dig a bit deeper into REAPER's capabilities, it's worth noting the fact that, like other Windows applications, REAPER lets you undo your work. Choosing the Edit, Undo command (Ctrl+Z) will undo your previous action, and you can repeat this as many times as you wish. For example, pressing Ctrl+Z four times will undo your last four actions. You can go all the way back to the start of the current session (that being when you last opened the project file) if you wish. This offers you some opportunity to recover from mistakes.

All recording software includes a basic undo capability of this nature. However, it's worth noting that one reason for using REAPER is that its Preferences settings offer you additional capabilities beyond this (see Figure 2.39).

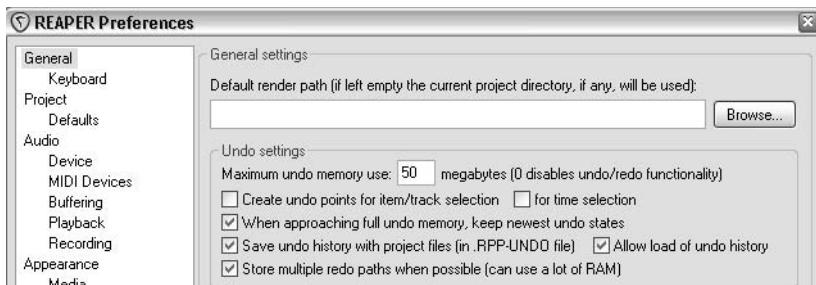
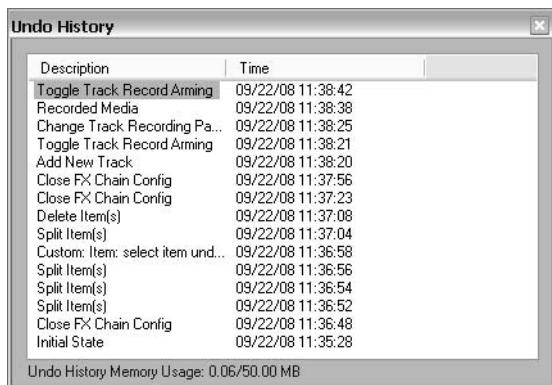


Figure 2.39 Setting Undo Preferences.

You can specify a number of Undo settings (under Options, Preferences, General), including the option of saving your Undo History with the project file. Provided this

option remains enabled, you will be able to go back and undo any change that you have made to the project file since its creation, even if it was yesterday or last week or last month.

You can view your project's Undo History (available from the View menu) and restore your project to any state just by double-clicking on that item (see Figure 2.40).



**Figure 2.40** Browsing the Undo History window.

What's more, you can even store multiple Redo paths. This option may seem quite mind-blowing at first, so don't worry too much about it for now. Basically, it allows you to restore your project to any earlier state, try out a whole different set of commands and actions, and then decide which of these sets you prefer to use.

Especially for a beginner or an inexperienced user, it can be quite reassuring to know that you have such a comprehensive safety net at your disposal.

## Simple Overdubbing

In this first example of simple overdubbing, you will be following the procedure outlined in the following section and recording your new material on a track by itself, just below the original recording. This method has the advantage of being relatively straightforward. Don't worry if some of the terminology seems a little strange—I'll explain it as we go along.

### Procedure

Here is an overview of the basic procedure for simple overdubbing:

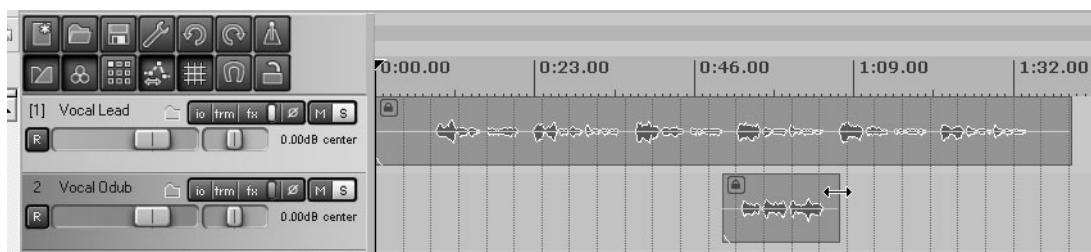
1. Insert a new track immediately below the existing track.
2. On the new track, record the new version of the material that you wish to replace.

3. Use splitting and slip editing to create the required blend of the old recording and the new recording.

### ***Example***

Let's now see exactly how this is done:

1. Open the file Happy Birthday 101 and immediately save it as Happy Birthday 102.
2. Play the song. With your headphones on, identify a short passage on the lead vocal track that you would like to replace. Even if you're fully happy with what you recorded before, pretend you're not. This exercise is about learning, remember, not about making the perfect recording.
3. In the Track Control Panel, select the Lead Vocal track and press Ctrl+T to insert a new track immediately beneath it.
4. Make sure your microphone is set up and ready to go. Practice singing along with the existing track to get your microphone placement and sound levels right.
5. Name the new track Vocal Odub and arm it for recording. Select the required input. If you are using input monitoring, turn it on.
6. After auditioning the part you wish to record, position the play cursor a few seconds before that point, press Ctrl+R to start recording, record the required passage, then press the spacebar to stop. Accept the option to save your item. Disarm the track.
7. Your screen should now be similar (but of course not identical) to that shown in Figure 2.41, with a new vocal clip just below the main vocal track.



**Figure 2.41** The original and overdubbed vocal tracks.

8. Notice there is an unwanted extra-short passage at both the end and the start of this item. You can remove this by slip editing these passages out. Make sure the newly recorded passage is selected. Hover the mouse over the end of the newly recorded passage so it changes to a double-headed black arrow, as shown in Figure 2.41.
9. Click and drag the mouse to the left, then release it.
10. Similarly, hover the mouse over the start of this passage to see the same double-headed arrow, then click and drag to the right, then release it.
11. If you drag the mouse too far it does not matter. *You are not really deleting the material, only hiding it.* This process, by the way, is known as *nondestructive editing*. You can click and drag in the opposite direction to restore it.
12. Now select the original Vocal Lead media item. Position the play cursor approximately in the middle of the area that you wish to replace with the new recording. Press S to split the item at that point (see Figure 2.42). If you forget to select the required media item before pressing S, then REAPER will split the items in all tracks at that point. If you make this mistake, you can press Ctrl+Z to undo.

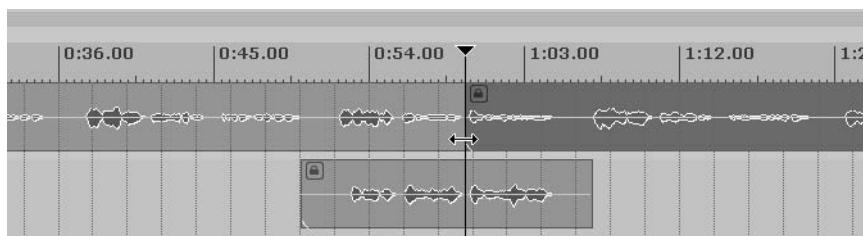
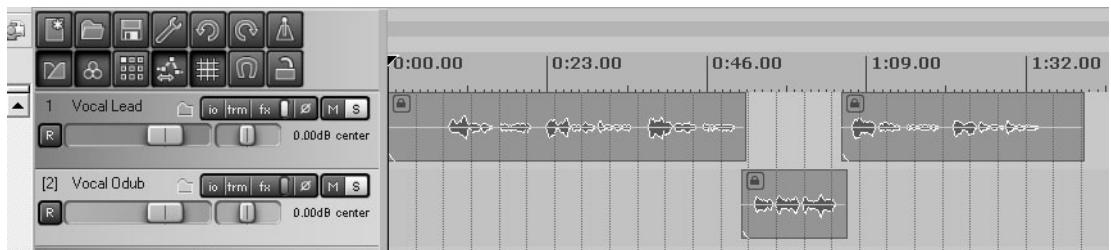


Figure 2.42 Splitting the original track.

13. You can now use the same slip editing technique as before to remove the unwanted portion of this media item.
14. When you have finished, your screen should resemble that shown in Figure 2.43. Don't forget to save the file.

## Assignment

Your assignment today is to work on your project files, practicing overdubbing, splitting, and slip editing until you are at ease with these techniques. Practice them over again.



**Figure 2.43** Both tracks after splitting.

Tomorrow we will look at punch recording. This is another method that you can use to overdub material. It is a little more complicated to set up, but once it is set up, it is faster and in some ways more elegant to use.

If you created the Project 2 example earlier, practice on this file also.

## Day 10: Auto Punch Recording

Auto punch recording is a method of overdubbing that is somewhat more sophisticated than the simple overdubbing that we have used up to now. In essence, with punch recording, you identify the portion of any track (or tracks) that you want to replace with a new recording. You can then start the recording process confident that the record mechanism will only be engaged during the portion of the song you specified.

Punch recording carries the advantage of not requiring any additional tracks to be created. You might wonder why this should matter. The answer is that if you have an arrangement that already includes perhaps 20 or 30 tracks, and you create another track for each item where you wish to overdub, you could easily end up with 40 or 50 tracks total. REAPER won't mind, but the greater the number of unnecessary tracks you create, the harder it gets for you to manage your project, and the greater the potential strain on your PC's resources when it comes to mixing.

### Overdubbing in Auto Punch Mode

In overview, the procedure for overdubbing in Auto Punch mode is this:

1. Turn on the auto punch recording feature.
2. Along the Timeline, select the area that you wish to rerecord.
3. Select and arm the track(s) to be recorded.
4. Record the necessary material.
5. Turn punch recording off and disarm the track.

This last point is important. If you forget to do this, you might run into problems the next time you wish to resume normal recording.

1. Open the file Happy Birthday 101 and immediately save it as Happy Birthday 103. *Make sure you open the right file—we don't want to make any changes just yet to the file we created for yesterday's lesson.*
2. From the menu, choose the Options command, then Record Mode: Time Selection Auto Punch. Notice that the appearance of the Record button on the Transport Bar has changed so that it now displays the text *loop punch*.
3. Identify on your Vocal Lead track the area you wish to replace. Click and drag along the Timeline to select this area (see Figure 2.44).

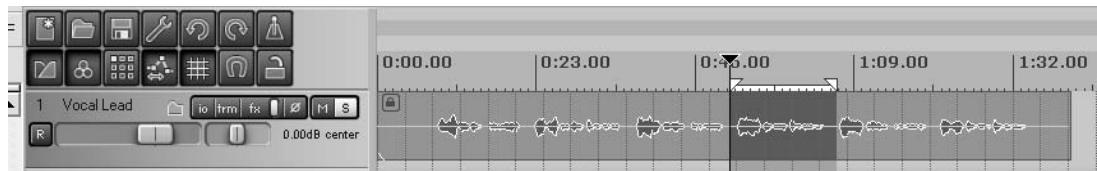
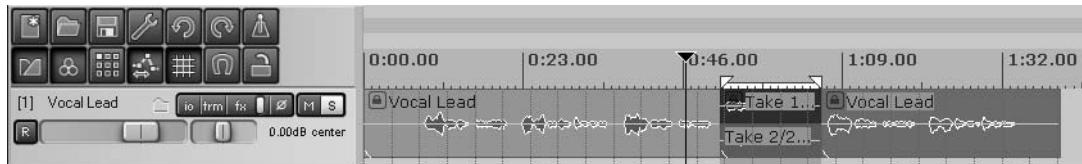


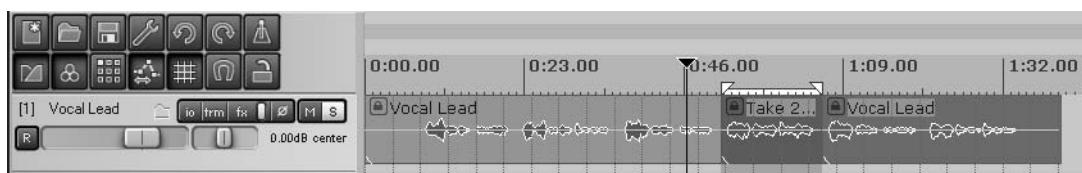
Figure 2.44 Making a time selection for auto punch overdubbing.

4. Make sure your microphone is set up correctly. Arm the track for recording and assign the correct input. Click your mouse either on the media item or just below it, a fair way before the area over which you wish to record.
5. Make sure the status of the Toggle Repeat button (to the right of the Record button on the Transport Bar) is set to Off.
6. Click on the Record button or press Ctrl+R to start the recording process.
7. Sing along with the song. Nothing will be recorded until you reach the start of the selected area. At the end of the selected area, recording will stop, but the transport will continue to roll.
8. Press the spacebar to stop and then choose the option to save your recording.
9. Return to the Options menu and choose Record Mode: Normal to restore normal record mode.
10. Save the file.

- Depending on your settings, you should now see one or the other of the screens shown in Figures 2.45 and 2.46. Pressing Ctrl+L will toggle you between these two ways of viewing your project.



**Figure 2.45** Takes displayed in multiple lanes.



**Figure 2.46** Takes displayed in a single lane.

- Press Ctrl+L to ensure that your display resembles that shown in Figure 2.45, so you can see both the original recording and the new one. These are known as Take 1 and Take 2.
- Play the song. By clicking on either of the two takes, you can select which one you would like to be selected and heard in the mix.

This exercise serves not only as a lesson in punch recording, but also as an introduction to working with multiple takes. We will explore the topic of multiple takes in more depth and more detail tomorrow.

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**Note:** When working in REAPER, you can choose whether to display vertical grid lines on your screen. This option is toggled on and off by clicking on the grid icon on the REAPER toolbar or by using the keyboard shortcut Alt+G.

---

## Assignment

Your assignment today is to work on your project files, practicing overdubbing using the auto punch technique until you are at ease with it. Practice it over again.

If you created the Project 2 example earlier, practice on this file also.

Hey! You're a third of the way through already. And it gets even better from here on in!

## Day 11: Recording and Working with Multiple Takes

Yesterday, you were introduced to the idea of takes when you experimented with auto punch recording. The essential quality of a take is that when second and subsequent takes are recorded, they are stored in the same track as the original take (media item). Then, as you will soon see, you can patch together your preferred track using the best aspects of all takes.

### Multiple Takes: The Basics

Notice that when you are using REAPER, no media item is ever deleted from your project unless you specifically ask it to be. Consider this example:

- You record a new track and save the recorded item.
- You return to the start of the track and record it again.

In this case, the second item does not replace the original (as would be the case, for example, if you were using magnetic tape to record). Instead it complements it—it sits there right alongside it.

You can choose to display either all takes (in parallel lanes) or only the selected (preferred) take. These two display states are toggled by choosing the Options, Show All Takes in Lanes command, or simply by pressing the keyboard shortcut Ctrl+L.

Before we continue, I should just note that not all DAW software handles multiple takes in the same way as REAPER. Some programs do not have this facility, and with others it may be an option that you have to turn on before recording. This carries the obvious risk that you might accidentally erase earlier takes without meaning to.

Figure 2.47 shows a recording with two tracks, each of which has two takes. In this example, I have chosen to use Take 1 of the Vocal track and Take 2 of the Guitar track.

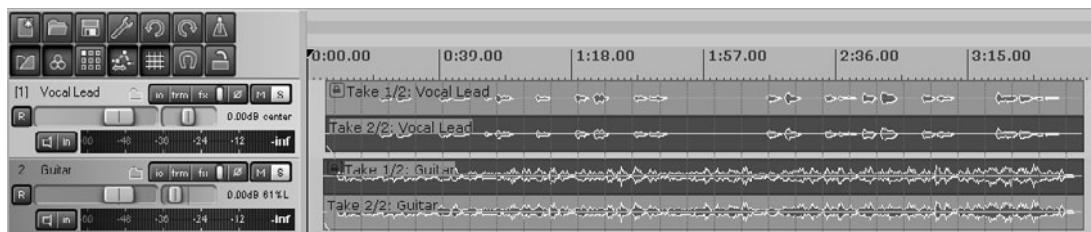
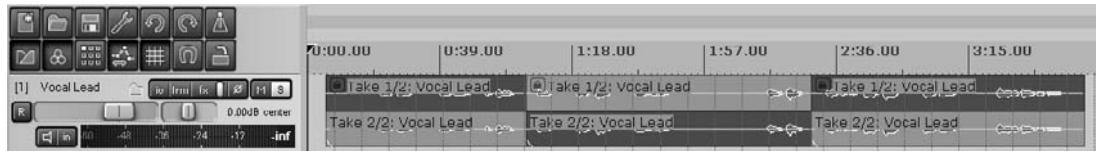


Figure 2.47 Two tracks with two takes each, displayed in lanes.

In Figure 2.48, there are two takes of a vocal track. These have been split in two places, so that each take is now made up of three media items. I have then patched the track together using the best from each take. In this case, I start with Take 2, then switch to Take 1, then back to Take 2 again.



**Figure 2.48** One track with two takes, sliced to select the best parts of each take.

Remember, the procedure for splitting is:

1. Select the media item.
2. Position the play cursor at the point where you wish to split.
3. Press S—or if you forget the keyboard shortcut, you can right-click over the item and choose Split Item at Cursor from the context menu.

### Exercise

1. Open the file Happy Birthday 101 and immediately save it as Happy Birthday 104.
2. Identify a track that is suitable for use with additional takes. A backing vocal track might be a good choice, but you may use an instrumental track if you wish.
3. Record two more takes for this track and make sure your takes are displayed in lanes. You should leave the track armed until you have finished your last take.
4. Using the method explained previously, split up this track into several segments, then patch your track back together by selecting the best items from each take.
5. When you're finished, press Ctrl+L to display only your selected items rather than all of the lanes.
6. Save the file.

## More Advanced Takes Management

We won't be doing much of this now, but it's worth noting that we have only scratched the surface of what you can do with takes in REAPER.

If you right-click over any take and then choose Take from the context menu, you will see a fly-out menu that shows many more things you can do with takes (see Figure 2.49).

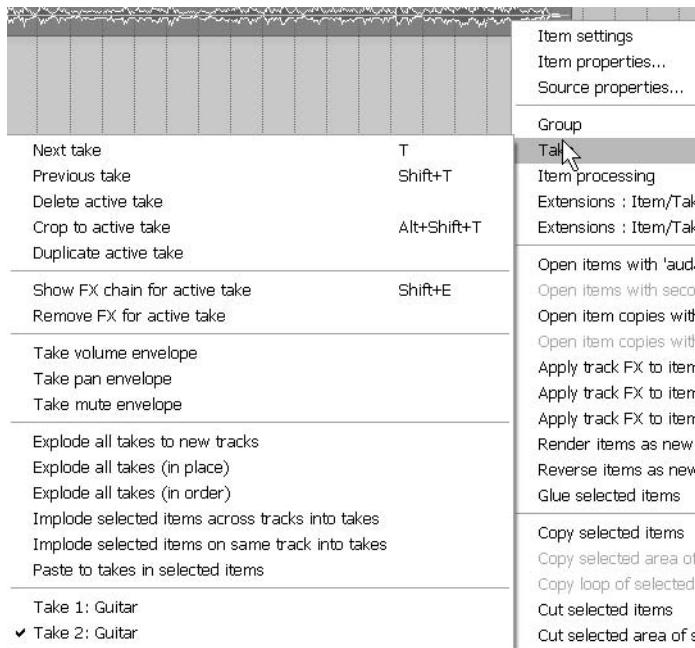


Figure 2.49 REAPER's Take menu.

Of course, most of these won't make too much sense to you right now, though by the end of these 30 days, many of them will.

For now, just take note of the fact that you can:

- Delete any take(s) that you are certain you do not want.
- Explode all takes to new tracks. This option can be useful if you decide you want to use more than one take in your mix.

If you wish, save your file to a new name, such as Project 104a, and experiment with this. Don't worry if you make mistakes!

## Assignment

Open your Project 2 (or some other) file and experiment with creating and working with multiple takes.

## Day 12: Basic MIDI Recording and Editing

From a teaching point of view, this is one of the most challenging sections to include in this foundation course. That is because the MIDI needs of individuals will vary so much. Some people will eventually become power MIDI users, while others will have absolutely no use for it whatsoever. What's more, even among those people who do use MIDI intensively, the way in which it is applied will vary enormously from person to person.

For that reason, this section will give you only a very general introduction to the topic. Its aims are:

- To enable you to develop a general understanding of what MIDI is and the essential ways in which it differs from audio
- To give you a simple glimpse at some of its potential applications and uses
- To give you a modest hands-on introduction to working within a MIDI environment

You can then begin to decide for yourself how important MIDI is likely to be to you and in which direction(s) you wish to take it (if any). After that...well, there are literally hundreds of books and websites devoted to the topic.

### MIDI Files and Synthesizers

Let's start by having a look at a very simple example of what a MIDI file is and how it interacts with a synthesizer (see Figure 2.50).



Figure 2.50 First look at a MIDI track.

1. Open the file Night.RPP. This is included in its own folder within the package of sample files that you copied from the CD earlier. Right away, save it as Night 1.RPP.
2. Play this file. You will notice that the play cursor moves from left to right as normal, but there are no sounds and no activity on the VU meters. That is because unlike an audio item, a MIDI item does not contain any actual sound. It only contains instructions that, when processed by a synthesizer, will be used to generate some sound. Stop playing and rewind the play cursor.
3. Let's add a synthesizer to this track. Click on the track's FX button (about midway between the track name and the Mute button) to display the Add FX window. Don't be overwhelmed by what you see! In the left-hand column, click on Instruments.
4. Double-click on VSTi: ReaSynth (Cockos) to insert this synthesizer into the track. Make sure you choose the correct instrument, *not* ReaSynDr (Cockos).
5. You should now see a display similar to that shown in Figure 2.51.

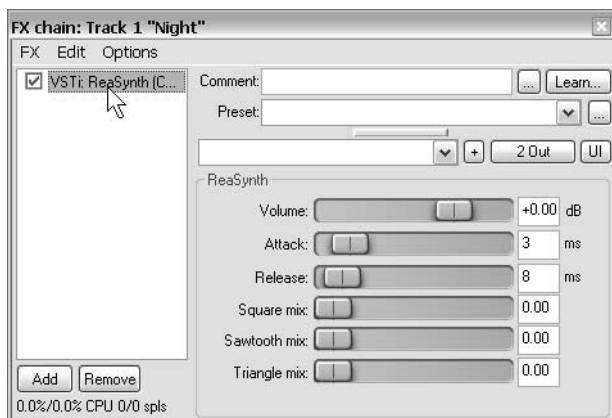


Figure 2.51 ReaSynth: An example of a simple, basic software synthesizer.

6. I'll say more about the ReaSynth interface shortly. For now, just note that synthesizers come in all sorts of shape and sizes and vary from being fairly simple to being quite complex devices. The example that we are using here is one of the less complicated.
7. Play the song again. This time you should hear music and see plenty of VU meter activity. Understand that the music is being created by the synthesizer

from the instructions in the MIDI file. No music as such has been recorded, only the instructions. We can demonstrate this now.

8. Move the three faders labeled Square Mix, Sawtooth Mix, and Triangle Mix to the right, so their values are shown as approximately 0.33, 0.54, and 0.79, respectively. The result is that the instruments that appear to be being played will change!
9. Experiment to your heart's content with these controls and also with the Attack and Release controls.
10. When you've had enough, stop the music and save the file.

## Simple MIDI Recording

There are two basic ways in which you can create your own MIDI files. You can record them, using a device such as an external keyboard or even a virtual keyboard, or you can hand-craft them using a MIDI editor. Before we finish this section, we will have looked at both methods. We'll start with the virtual keyboard.

If you have an actual keyboard (such as a Casio, Korg, Roland, or Yamaha), you can use that. However, for this next example, you'll use the virtual MIDI keyboard that comes supplied with REAPER. It's a little basic, but fine for our purposes.

### **Example**

1. Use the File, New Project command to create a new project. Press Ctrl+T to create a single track. Save the file into its own subdirectory with the name MIDI 1.
2. Name the track Keyboard. Arm your track for recording. Set the MIDI Input to Virtual MIDI Keyboard, All Channels (see Figure 2.52). Click on the In button and select the Input Monitoring option.

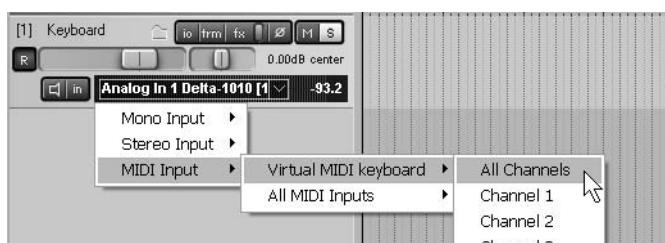


Figure 2.52 Getting ready for MIDI recording.

3. Fade the volume fader down to about -14 dB. This is a sensible precaution against making your signal too loud. You have already seen that when you are recording audio, the track volume fader does not affect the volume of the audio being recorded. When you are using the Virtual Keyboard with a MIDI track and a synthesizer, however, you have a different situation altogether. The track's volume fader will help determine the strength of the output signal that you hear.
4. Choose the View, Show Virtual MIDI Keyboard command. This displays the keyboard, as shown in Figure 2.53. If you wish, you can increase the area displayed in the same way you would resize any other window.

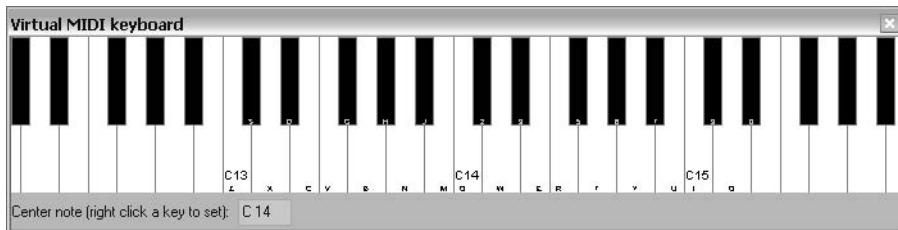


Figure 2.53 The Virtual MIDI Keyboard.

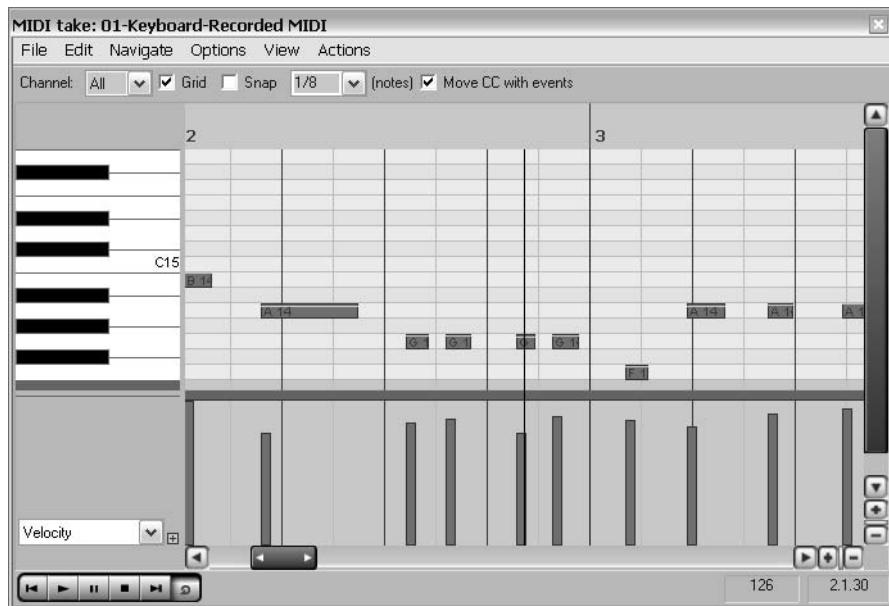
5. You can right-click over any note to make it the center note. You can play the keyboard itself by left-clicking the keys.
6. If you try playing now, you will hear nothing. That's because the instructions that are generated when you play are not being directed to any synthesizer.
7. Insert ReaSynth into the track's FX chain (as before) and start playing. Adjust the track's volume fader to suit.
8. Work out a simple tune of about 20 or 30 seconds' duration. When you are ready, press Ctrl+R to start recording, then play your tune on the keyboard. Stop the recording and save your MIDI item. Don't worry about how good or bad the composition is. Disarm the track when you are finished.
9. Your screen will now resemble the one shown in Figure 2.54.
10. Save the file.
11. Play back your MIDI track.
12. If you wish, adjust the various ReaSynth faders to ensure a different sound.



**Figure 2.54** A simple recorded MIDI item.

## Simple MIDI Editing

REAPER's MIDI Editor (see Figure 2.55) lets you edit your MIDI items in many ways. For example, you can move them around, lengthen them, shorten them, change their pitch, make changes to their velocity and expression, and do much, much more besides. In this foundation course, we won't go beyond the most fundamental aspects of MIDI editing, just to introduce you to the concept.



**Figure 2.55** REAPER's MIDI Editor.

Here is an example that you can work through.

1. Double-click on your recorded MIDI item to take it into the MIDI Editor. Here you can see, among other things, that the notes you have recorded are displayed.

2. In this simple example, we will just look at four basic tasks, namely deleting notes, making changes to notes, adding notes, and moving notes.
3. As with most computer programs, there is more than one way to accomplish the most common tasks. If in doubt, your best bet is usually to right-click in the appropriate place and look on the menu for what you want to do.
4. Let's delete a note. Find a note that you wish to delete, right-click over it, and choose Delete Notes from the menu.
5. Now let's make the note before the one that you just deleted longer. Position the mouse over the end of the note that you wish to make longer. Your mouse will turn to a double-headed arrow, just as before when you were slip editing. Click and drag to the right, then release the mouse.
6. To add extra notes, just hover the mouse at the required place, then click and drag to draw the note.
7. To move a note, just drag and drop with the mouse.
8. To work with a number of notes, you can right-click and drag around them to select them. You can then, for example, click and drag to move all selected notes or press the Delete key to delete them.
9. That's it for now! Close the MIDI Editor and save the file.

There's one more thing that I should mention here, and it's important. Don't think of MIDI as being a self-contained island on its own. You can use both MIDI tracks and audio tracks in the same projects, and even a combination of MIDI items and audio items on the same tracks.

Figure 2.56 shows a project with six audio tracks and a seventh MIDI track (Track 4), which has been added to create more atmosphere during the instrumental passage near the end of the song.

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**Note 1** Today we have barely scratched the surface of the MIDI Editor and its capabilities. The variety of applications for synthesizers is enormous. These include virtual instruments (pianos, trumpets, flutes, or whatever you want), drum kits, and electronic music. I'll have more to say about this in a later chapter.

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**Note 2** With some DAW software, the MIDI Editor can be opened directly from the program's main menu. With REAPER, however, the only way to open the MIDI Editor is with an existing MIDI item.

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**Figure 2.56** A project with both audio and MIDI items.

If you wish to use the MIDI Editor to create a MIDI item (rather than record one first), you can do this by first using the Insert, New MIDI Item command to create an empty MIDI item. Before doing so, you should create a track for this item (or select an existing one) and select an area along the Timeline to define the required position and length of this item. You can then open the MIDI Editor with your new empty MIDI item.

## Assignment

1. Open the file Happy Birthday 104 and save it as Happy Birthday 105.
2. Identify an opportunity where you perhaps can add a short MIDI section to enhance this arrangement. This might be during an instrumental break or a fadeout near the end.
3. Create a MIDI item for this purpose. Don't be overly ambitious. As yet, you have been taught very little about MIDI, and you have been introduced to only one very basic synthesizer. You won't be able to achieve miracles! Treat this exercise as one of fun, learning, and creativity.
4. If you wish, try using this technique on one of your other projects, too.

## Day 13: Creating and Using Loops

Loops are passages of music that are repeated over and over throughout a song. A loop can be constructed equally easily from a MIDI item or an audio item. Let's begin this section by looking at some examples.

When you installed REAPER, if you accepted the default options, a project file called BradSucks\_MakingMeNervous.RPP was installed along with it into a folder called C:\Program Files\REAPER\BradSucks\_MakingMeNervous. Open this file in REAPER and save it to a new name. Play it. As you listen to it, examine the tracks carefully.

You'll see that some of the tracks appear to have regular notches all the way along them. In Figure 2.57, you can take a closer look at the first three tracks.

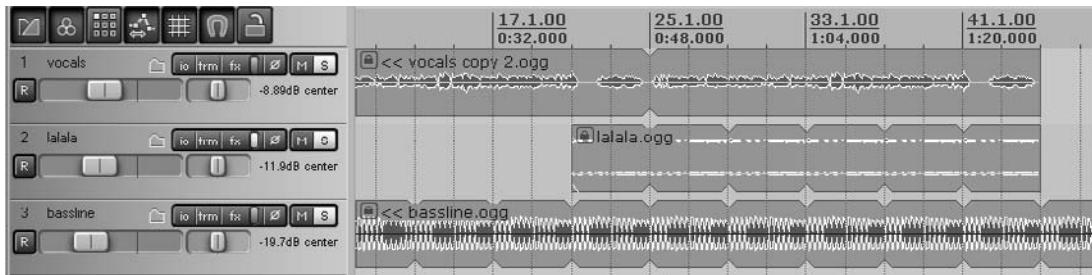


Figure 2.57 Three examples of tracks created using loops.

Notice that Tracks 2 and 3 display regular notches at quite short intervals. In Track 1 these notches are farther apart, but where they occur, they fall on a gridline, too. What's going on? The answer is looping.

In each case, the original media item begins at the start of each item and ends where you can now see the first notch. By first hovering the mouse over the end of the item, rather like you did when you were slip editing, you are able to click-hold and drag the item out so that it will be repeated as many times as may be required.

There is an item on REAPER's Options menu to Enable Snapping (shortcut: Alt+S). Turning on this option can help you make sure your various items in different tracks line up exactly when you are working with loops. If you try dragging one of the looped items in this file to the right, for example, you will find that with snapping enabled, it is impossible to nudge it slightly out of position. It will snap to the next beat. If you try this out, don't forget to drag the item back again.

Snap settings can be set on a per-project basis by displaying the Snap/Grid Settings dialog box (also from the Options menu). At this point in time, we don't need to get into too much detail—you just need to understand the general idea of what's going on here.

Looping can be applied to both MIDI items and audio items. Some people use loops extensively, especially for bass lines and percussion. Others rarely use them, if ever. Their advantages include precision in timing and performance. However, some would say that the use of loops dehumanizes the music and detracts from spontaneity and creativity in performance. Whether the use of loops is appropriate depends as much as anything else on the style of music and your personal preferences.

## Working with Loops

In the example that follows, you will record a simple percussion instrument, such as a shaker. If you don't have anything suitable available, just clap your hands instead. You will then make this item into a loop, after which you can overdub a vocal alongside it on a new track.

The object of this exercise is to give you some hands-on experience in working with loops. Don't worry too much about the quality of either your handclaps or your singing!

1. Create a new REAPER project file. Check the Project Settings—use Alt+Enter if they are not displayed automatically when you create the file—to ensure that you have a Project BPM setting of 120 and a Time Signature setting of 4/4.
2. Set the Timebase to Beats (Position, Length, Rate), as shown in Figure 2.58. For the moment, don't be too concerned about the reasons for this—it just makes life easier when we are looping and snapping.

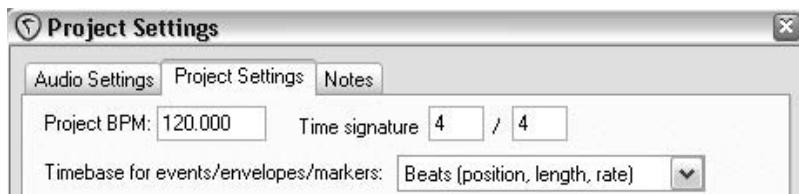


Figure 2.58 Setting the project timebase in REAPER.

3. Press Ctrl+T to insert a new track. Name this track—for example, Shaker or Handclaps.
4. Save this project into its own directory along with its media files. Call the file MyLoops 1. Make sure the grid lines are displayed.

5. Right-click over the Timeline and choose Measures – Beats (M:S Secondary) as the units to be used for display. This ensures that your gridlines will align with your beats.
6. Arm your track for recording in the usual way and record a measure (four beats) of your percussive instrument of choice. Most likely, you will have some silence recorded before and after the recorded material.
7. Play it to make sure you are happy with it. If you’re not, delete it and try again. When you have finished, disarm the track.
8. Slip edit the item (if necessary, from both ends) to the required length. As with the loops in the BradSucks file, your media item should start and finish exactly at grid settings.
9. Right-click over the item and choose Glue Selected Items from the menu.
10. If you now slip edit the item outwards from the end, it will be looped. Extend it out to about 20 or 30 seconds.



Figure 2.59 Creating a looped item.

11. Now suppose you think your beat sounds too slow. Open the Project Settings window and change the Project BPM from 120 to 140. Close this window. Notice what’s happened! The item has been sped up and is shorter than before. Of course, if you wished, you could extend the looped item farther to the right to make it longer again. For this example, though, we won’t bother.
12. Add another track, name it Vox, and record yourself singing along with the handclaps. If you like, just sing something like “Doo Dah” for the purpose of completing this example.
13. Now play back the percussion and your singing together.
14. Save the file.

## Assignment

You've got a pretty free hand here. Have some fun with loops. Don't expect to become an expert on every facet, but do aim to learn enough to at least have a handle on what you're doing.

Design a modest project for yourself that uses at least one loop (similar perhaps to the example you have just completed) and implement that project.

## Day 14: Sampling: Where Audio Meets MIDI

We've worked with MIDI, and we've worked with audio. Now it's time to take a brief look at sampling. *Sampling* is a process by which you can use MIDI data to shape or alter the sound produced by previously recorded audio data. Especially if you have never come across this before, you are really going to enjoy this section.

The package of sample files that you downloaded earlier includes a number of small .wav files with names like fiddle.wav and harmonica.wav. I am going to use these to demonstrate what I mean by sampling and to give you a very basic introduction to what sampling software does.

### Working with Audio Samples

In the example that follows, we will use REAPER's Virtual Keyboard to create a MIDI track in conjunction with a sampled .wav file of a fiddle. Because of the limitations that this imposes, don't expect the results to be too spectacular! This is primarily a learning exercise, remember.

The example that follows will use REAPER's plug-in ReaSamplOmatic5000. It is an example of a very simple and basic sampler (see Figure 2.60).

Before you begin, use Windows Explorer to find the folder containing the sample .wav files. The folder's name will be Sounds. Open it and, using your customary software (such as Windows Media Player), play the file fiddle.wav. You can see and hear that it contains a short sample of a violin being played. Close the media player.

1. Create a new REAPER project file. Save it as Samples 1.RPP, making sure that it has its own subdirectory and that any media items associated with this project are copied there.
2. Display the Virtual Keyboard.
3. Insert a new track into your project file. Arm it for recording and select as your MIDI input the Virtual Keyboard (all channels).

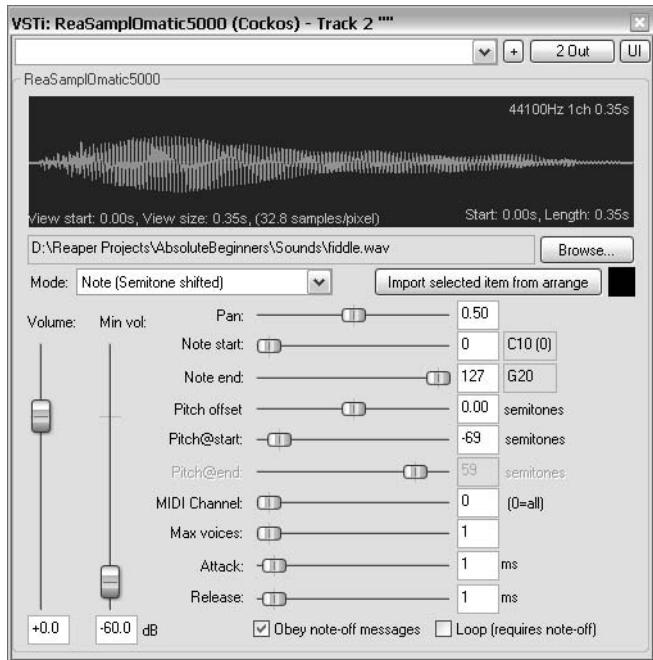


Figure 2.60 The ReaSamplOmatic5000 interface.

4. Set input monitoring to On for this track.
5. Click on the FX button for this track to open the Add FX window. Click on the category Cockos (in the first column), then in the second column, click on ReaSamplOmatic5000. Click on OK to insert this effect into your track.
6. Now you'll import the file fiddle.wav into this sampling software. Click on the Browse button and navigate to your Sounds folder. Select the file fiddle.wav and click Open.
7. Your ReaSamplOmatic5000 window should now be similar to the one shown in Figure 2.60.
8. You can play the Virtual Keyboard either with your mouse or using the keys on your QWERTY keyboard. If you wish, you can right-click over any note to make it the Virtual Keyboard's center note.
9. After some practice, record about 20 or 30 seconds of music, then stop the recording, save the recorded media, and disarm the track.

10. Play the track back. Notice that you can adjust the parameters within ReaSamplOmatic to achieve some interesting results. For example, you can change Max Voices to 2 or 3 and/or set the Release time to around 100 ms.
11. Let's just recap what's going on here. You have used an audio sample of a fiddle to create a MIDI track. You have not recorded a fiddle! We can easily prove that.
12. Click on the Browse button again, then select slide.wav and click on Open. This file contains a sampled recording of a slide guitar.
13. Play the track again. Can you see what's happening here? The sampled .wav file is played back by the synthesizer.
14. Save the file.

## Assignment

1. Create a new REAPER project file. Name it My Samples.RPP and ensure it is stored along with its media in its own folder.
2. Create three tracks, named Aaaah, Ooooh, and Mmmm.
3. Record on the first track a very short sample of yourself singing “Aaaah”—just a few seconds.
4. Mute it while you record on the second track a similar sample of yourself singing “Ooooh.”
5. Repeat this process to record yourself singing “Mmmm” on the third track.
6. Save the file and close it. Create a new project file. Save it to its own folder with all its media items being copied there.
7. Slip edit your media items to hide any unwanted silence at the beginning and end. In each case, right-click over the item and choose Glue Selected Items to actually remove the silence.
8. Create a new track and arm it for recording. Insert ReaSamplOmatic5000 into this track. Insert the file Aaaah.wav into ReaSamplOmatic5000.
9. Use the Virtual Keyboard to record a MIDI track using this file.
10. Play back your recording. In the ReaSamplOmatic5000 window, experiment with substituting your two other voice sample files, one at a time.
11. Save the file when you are finished.

## Bonus Example: The Drum Kit

In this 30-day foundation course, it is of course only possible to skim over the surface of most topics. Nowhere is this truer than with sampling. We could quite easily spend around half the course on this topic alone. Before moving on, I'd like to show you just one more example that uses sampling. Again, it's a fairly simple one, this time involving a drum kit.

1. Open the file Birthday Drums.RPP and play it. You should be able to hear the sound of a snare and a kick. Save the file as Birthday Drums Plus.RPP.
2. Open the FX chain for the track. You can see that eight separate instances of ReaSamplOmatic5000 have been inserted into this FX chain, each with a sampled .wav file of a different percussion instrument. This should be similar to that shown in Figure 2.61.

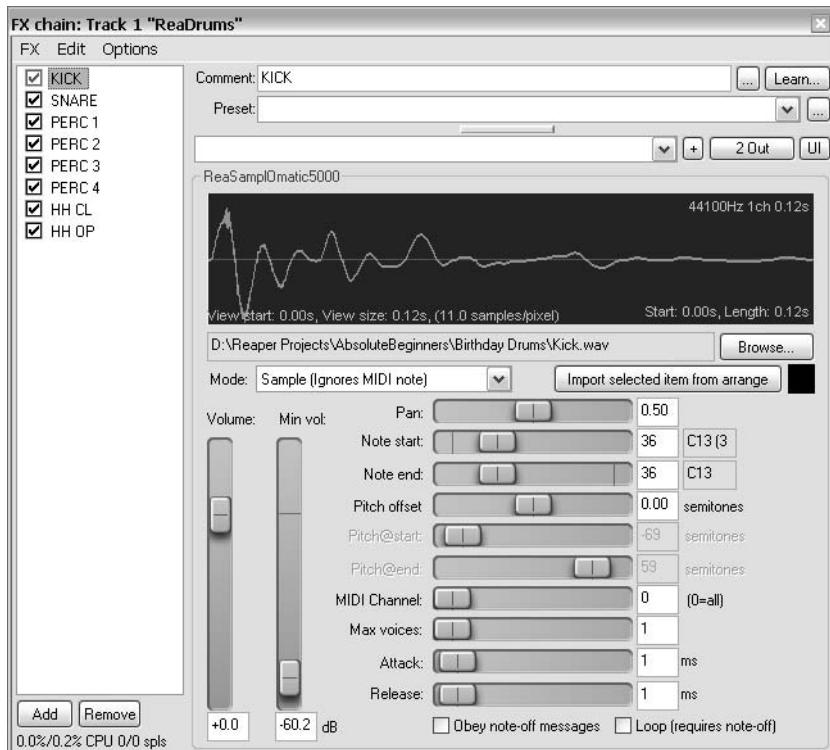


Figure 2.61 Using multiple instances of ReaSamplOmatic5000 to create a drum kit.

3. Close the FX chain and double-click on the second MIDI item to open it in the MIDI Editor. There you can see where the notes have been inserted for the kick and the snare.
4. Click anywhere in the editing grid in the row labeled Hat Open.wav to insert an Open Hat note at that point. This is just a learning exercise, remember. If you wish, you can drag and drop the note along its row to reposition it.
5. Play the file and save it. If you wish, continue experimenting with editing this file.

## Day 15: Consolidation Exercise

Congratulations, you have reached the halfway point of this foundation course!

We've gone as far into recording as we're going to go. For the remainder of this course, we will be concentrating on the post-production aspects of recording. These are the various tasks that you will need to understand to transform your raw (or *dry*) recordings into a quality product.

This is a good time to take stock of what we've covered so far.

If you'd like, choose a well-known simple song, such as a nursery rhyme, an old standard, or a Christmas song. Work out a modest arrangement and record it.

Save this project in its own folder along with its media items as Assignment 15.RPP.

## Day 16: First Steps in Mixing—Pan and Volume Faders

Mixing your project requires you to get on top of an astonishing number of techniques. Over the next few days, I'll be introducing you to some of the most important of these, starting with the pan and volume faders.

You're going to enjoy this lesson. It will probably take you no more than about 10 or 15 minutes (if that) to grasp the concepts and the basic techniques. You will then be able to spend the rest of your life refining and improving how you implement them!

### The Importance of Space

One very important consideration to keep in mind when you are mixing is the importance of space. Put at its simplest, your mix will consist of a number of tracks. This could be as few as perhaps three or four or as many as fifty or sixty (or more). If you had a separate speaker for each track, then it wouldn't be too difficult for you to play back your production so that each instrument could be clearly and distinctly heard. Unfortunately, you do not have that luxury. Instead, all of these tracks will need to

be mixed down to be played back over just two speakers—left and right. Somehow you have to make sure that each part—every instrument and voice—finds enough space in these two speakers to make itself heard.

Two of the techniques available to help you achieve this are the pan and volume faders.

Pan faders are used to determine how a track's output will be directed between the two speakers. This can be anything from left speaker only (100% left) to right speaker only (100% right), or any combination in between.

Volume faders are used to determine for each track the strength of the signal that is sent to the two speakers.

Figure 2.62 shows one track in REAPER's Track Control Panel with the two faders, volume (left), and pan (right).



Figure 2.62 REAPER's Track Control Panel volume and pan faders.

Notice the information 0.00dB center is displayed on the panel. This indicates the volume level and panning. For example, moving the volume fader to the left will cause the display to change to a negative number as you lower the volume. Moving the pan fader, say, to the right will cause the display to change from center to a number such as 40% R.

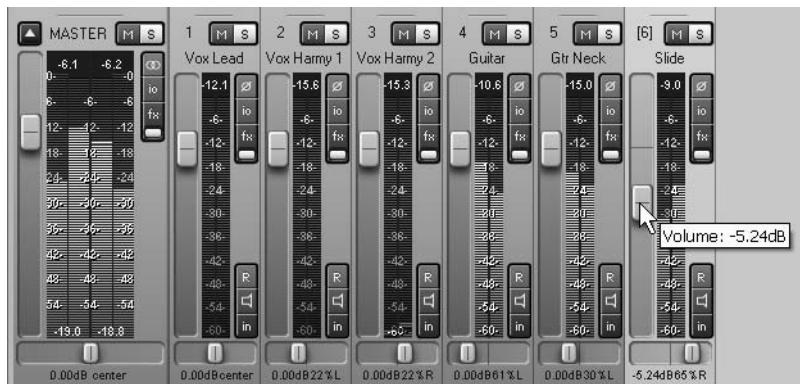
In this next example, you will see how by adjusting volume and pan, you can change and improve the sound of a mix.

1. Open the file RosesBloom.RPP and play it. You should notice that although you can hear the various parts that make up this song, you cannot really hear all of the items too clearly.
2. Now adjust the pan faders for Tracks 4, 5, and 6 approximately as shown in Figure 2.63. To do this, click and drag with your mouse. Don't worry if you do not get the exact same settings. In this example, near enough is good enough. For example, set Track 6 anywhere between 60% right and 70% right.
3. As you play the song, you should now be able to hear the acoustic guitar and the slide guitar more distinctly. Notice, too, that you can also hear the main vocal more clearly, even though you have not adjusted its settings. This is because by moving the three tracks away from it, you have made more space for the vocals.



**Figure 2.63** A project file with pan faders adjusted to give each track its own space.

4. Now adjust the panning for Tracks 2 and 3 approximately as shown. Each individual harmony should now be capable of being heard more distinctly.
5. Save the file as RosesBloom 16.RPP.
6. Play the file again. Notice that you can toggle the solo buttons—marked S—and/or the mute buttons—marked M—if you wish to focus on an individual track or selection of tracks.
7. Observe the VU meters in the mixer while you play the song. Notice that the relative height of the two lines (left and right) for each track reflects how you have panned that track. The REAPER mixer is shown in Figure 2.64.



**Figure 2.64** The REAPER mixer.

8. It might seem to you that the slide guitar is somewhat dominant compared to the acoustic guitar tracks. To correct this, experiment with adjusting the volume faders on these three tracks, possibly settling on levels similar to those shown in Figure 2.64.
9. Save the file.
10. Try experimenting with other permutations of pan and volume settings for the various tracks. You will probably be surprised by how much difference seemingly small changes can make.
11. If you wish to save your changes, do so to a new name.

## Assignment

1. Open your project file Happy Birthday 105 and save it as Happy Birthday 106.
2. Experiment with pan and volume fader settings for the individual tracks in this project. When you like what you have, save it.
3. If you made any more project files in the earlier assignments, experiment with pan and fader settings for these files also and save them.

## Day 17: Introduction to Audio Editing

A very important aspect of post-production is *editing* your recorded material. Editing encompasses a number of tasks. Today we will look at two examples, just enough to give you a general understanding of the kind of activity that editing involves.

One of the great strengths of REAPER is that whenever you identify a task or a sequence of tasks that you need to use, often you are able to create a *custom action* that enables these tasks to be bound together whenever you need them to be. Although you won't be doing that today, it's useful for you to at least be aware that this can be done. In a later lesson, we will look at some relatively simple examples of making and using custom actions.

Today we will look at three fairly common editing tasks:

- Selecting and muting parts of any track or tracks
- Applying a fadeout at the end of a track
- Editing multiple takes

## Muting a Passage

Often you will wish to remove parts of a recorded media item from your mix. To mute part of a track, you need to follow this sequence:

1. Select the media item that includes the area to be muted.
2. Define the time selection that is to be muted.
3. Split the item at the time selection.
4. Mute the new split item.

This might seem a little cumbersome. Remember that although you will not be doing so today, later you will learn how you can combine these steps into one custom action when you become a little more experienced.

1. Open the file RosesBloom 16.RPP and save it as RosesBloom 17.RPP.
2. Suppose you wish to consider muting the section from the 21-second mark to the 26-second mark on Track 3. You wish to do it in such a way that you can easily change your mind later if you wish.
3. Click on the media item in Track 3 to select it. In Figure 2.65, you can see that the background of the selected item is darker. On your screen it will be shown in a different color.



**Figure 2.65** Selecting a portion of a media item for editing.

4. Click and drag along the Timeline from approximately the 21-second mark to the 26-second mark. You can see that this time selection is shaded. The actual time selection is also indicated on the Transport Bar (to the right).
5. Right-click on the selected area of the selected item and, from the menu, choose Split Items at Time Selection (near the bottom of the menu).

6. Right-click over the same area again, then choose Item Settings, then Mute. The selected passage of Track 3 will now be muted and shown in a darker color.
7. At any time you wish, you can right-click over this item and choose the Item Settings then Mute command to toggle on and off the mute status of this passage.
8. This exercise may be regarded as a demonstration only. When you have finished, unmute any passages that are still muted.

## Creating a Fadeout

A fadeout at the end of a track enables you to control the rate at which the sound of an instrument decays. It is a very commonly used technique.

The procedure for doing this is to first slip edit the track to remove any excess silence at the end, then add a fadeout curve at the end of the track.

1. Using the technique you learned earlier, slip edit the Slide Guitar track to remove the excess silence from the end (see the image furthest left in Figure 2.66).

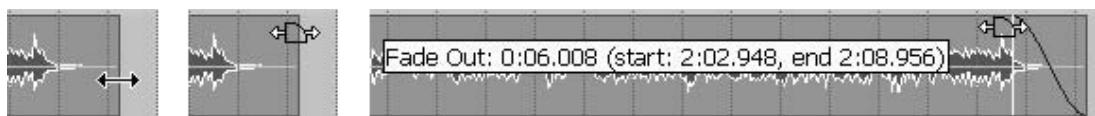


Figure 2.66 Slip editing (left) and creating a fadeout.

2. Position the mouse over the right edge of this media item, but a little higher up, so as to see the fader displayed (see the second image in Figure 2.66).
3. Click and drag to the left to define the length of the fade, then release the mouse (see the third image in Figure 2.66).

If you wish to experiment a little more, you can right-click on the fader curve and change the shape and intensity of the fade. You can also adjust its starting point by clicking and dragging it along the media item.

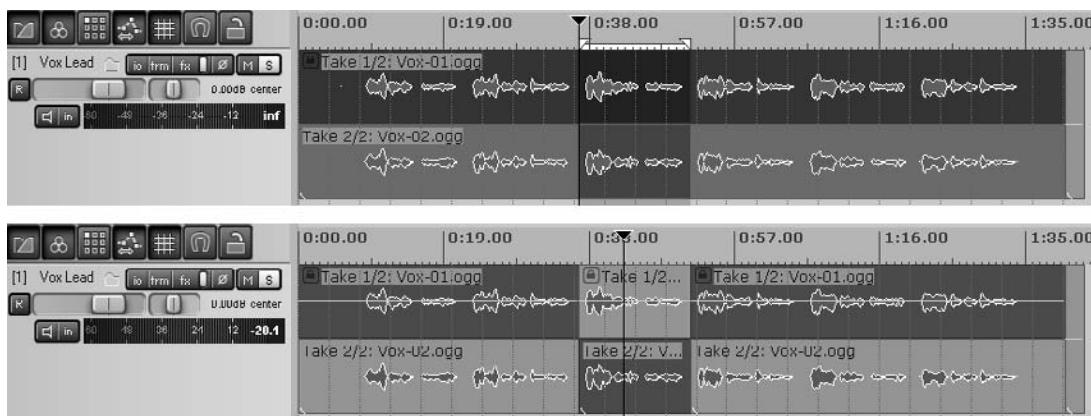
## Editing Multiple Takes

At first this will be the trickiest of the three techniques that you are using today. It is probably also the most interesting, and one that opens up a new area of creative opportunities for you.

You might recall that a few days ago, you learned how to record more than one take of a track. You did this with your lead vocals. Up until now, you have contented yourself

with simply choosing which take you prefer. Now you are going to learn how to slice up a track that consists of more than one take, so you can use the best parts of each take.

1. With the file RosesBloom 17.RPP open, select the Vox Lead track.
2. Expand the height of this track, then press Ctrl+L. The Track view will expand to show you that there are two takes for this track. Both layers are now shown.
3. The currently selected take is shown in a lighter color than the other takes.
4. Solo this track and play the song. As you do so, switch from take to take.
5. Suppose you have decided that you wish to use all of Take 2, except for the passage from 36 seconds to 51 seconds. For this passage, you'll select Take 1. The first step is to split the item.
6. With this media item selected, select the required passage (see the top half of Figure 2.67).



**Figure 2.67** Working with multiple takes.

7. Right-click over the Timeline and choose Split Items at Time Selection from the menu.
8. Press Escape to clear the time selection, then simply click on Take 2 for the selection that you have just split. Your selection is now complete, as shown in the lower part of Figure 2.67.
9. Of course, you could make more splits if you wished to do so. When you have finished, press Ctrl+L again to collapse your track back to a single layer. At each point where you have created a split, the selected take will be identified.
10. Save the file.

REAPER's take-management features go quite a bit deeper than just this. For this foundation course, however, what is important is that you understand the main concept of working with and editing multiple takes.

## Assignment

1. Add fadeouts to the end of the two guitar tracks in the file RosesBloom 17.
2. In Track 3, mute the passage from 1 minute 6 seconds to 1 minute 20 seconds.
3. Save the file.
4. Experiment with applying these techniques to your other projects. In particular, make sure you come to grips with working with and editing tracks that contain multiple takes.
5. Save your work when you are finished.

## Day 18: Plug-Ins and FX: EQ

Over the next few days, we'll be taking an introductory look at some of the effects plug-ins (often known as *FX*) that are widely used to polish and improve the sound of your productions. A plug-in can be thought of as being like a piece of software that by itself serves little or no purpose, but which can be used in conjunction with your DAW software.

These plug-ins generally fall within three broad categories, and we will be looking at some relatively simple examples in all three categories:

- **Frequency manipulation.** These plug-ins work by modifying the frequencies of the sound waves.
- **Dynamic manipulation.** These plug-ins work by changing the dynamics of your material.
- **Time manipulation.** These plug-ins work by altering the way in which your recordings interact with time.

## Manipulating Frequencies

The sound spectrum is measured in Hz and KHz. It ranges from very low (starting below the range of human hearing) to very high (way beyond the range of human hearing). The diagram in Figure 2.68 shows how this can be roughly divided.

Bass (Low)	Low Mids	Mids	Highs	Ultra Highs
0	200	800	5,000	8,000 20,000+

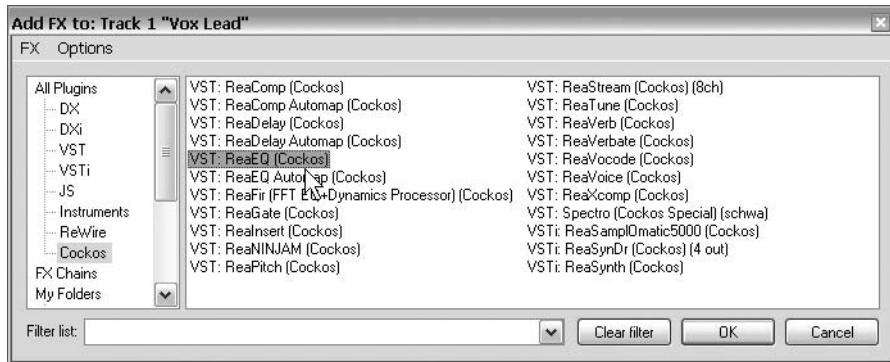
Figure 2.68 The sound frequency spectrum.

For example, this shows that the range from about 200 Hz to about 800 Hz covers the part of the spectrum we sometimes call the *low mids*.

By emphasizing or deemphasizing certain frequencies, we are able to modify how the sound is heard by the listener. For example, if we increase the volume at around 3,000 to 5,000 Hz, we can make a voice or instrument appear to be more present. If we increase it around the 200-Hz mark, we may make it sound somewhat warmer.

It takes a considerable amount of time and experience to really get on top of how you can make the best use of this phenomenon. Today we will explore some simple examples, using a type of plug-in known as *equalization*, or often just *EQ*.

1. Open the file RosesBloom 17.RPP and save it as RosesBloom 18.RPP.
2. Select Track 1, the Vox Lead track.
3. Click on the FX button (just a little way to the right of the track name) to open the Add FX window. Look in the left-hand pane. A little way down from the heading All Plugins, there is a subheading Cockos. Click on this. The right-hand pane will now display a list of Cockos plug-ins, as shown in Figure 2.69. It does not matter if your list is not exactly the same as the one shown here.



**Figure 2.69** Inserting a plug-in into an FX chain.

4. Double-click on ReaEQ (Cockos) to add this plug-in to the selected track. Notice that the ReaEQ window includes the following controls:
  - A series of tabs numbered from 1 to 4. These are where you can specify your requirements for any of four bands to which you want to apply EQ.
  - A fader labeled Frequency. This is used to set the required frequency for the currently selected band.
  - A fader labeled Gain. This is used to set the required volume gain for the currently selected band.

- A fader labeled Bandwidth. This is used to determine the range of the frequency for the currently selected band.

When you first open the ReaEQ window, it should resemble the example shown in the first (top) example of Figure 2.70. By the time you have finished this exercise, its appearance should be similar to that shown in the lower example.

5. Observe the ReaEQ window. In particular, notice how by clicking on any of the four numbered tabs, you select a particular band. Play the song.
6. Select Band 2. Set the Frequency to about 180 Hz and the Gain for this band to about 6 dB. Lower the *overall* Gain fader (on the right-hand edge of the box) to about -1.0. You should notice a difference in the quality of the voice.
7. Notice in the FX Chain window there is a small check box called Enabled next to the name of the FX. By clicking on this, you are able to toggle bypass on and off for this effect. This is a useful tool for evaluating the effect of your changes.
8. Now adjust the Bandwidth fader to about 1.20. This reduces the frequency range over which your EQ changes are applied.
9. Select Band 3.
10. Set the Frequency to about 3250 Hz, Gain to about 4.0, and Octave Bandwidth to about 1.3. These settings do not need to be exact. As you play the song, toggle bypass on and off. You should notice a significant difference in how the Vox Lead track sounds with your EQ changes enabled and disabled. In particular, it should stand out above the overall mix a lot more.
11. Save the file.

## Assignment

1. Solo and listen to the track called Guitar.
2. See whether you can use ReaEQ to make this track sound a little brighter. You will need to listen to it with the rest of the mix as well as soloed.
3. Save the file when finished.
4. Experiment with the application of EQ to any of your other project files. Be cautious at first and try not to be overly ambitious. Remember that the object of this exercise is to help you understand how frequency-manipulation plug-ins such as EQ work, not to pursue the Holy Grail of the perfect sound.

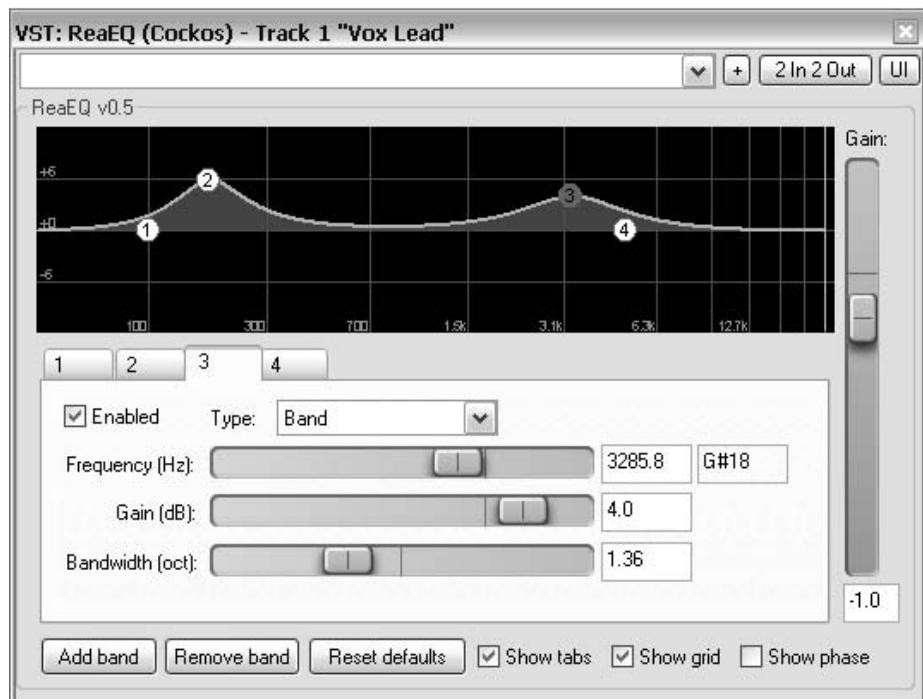
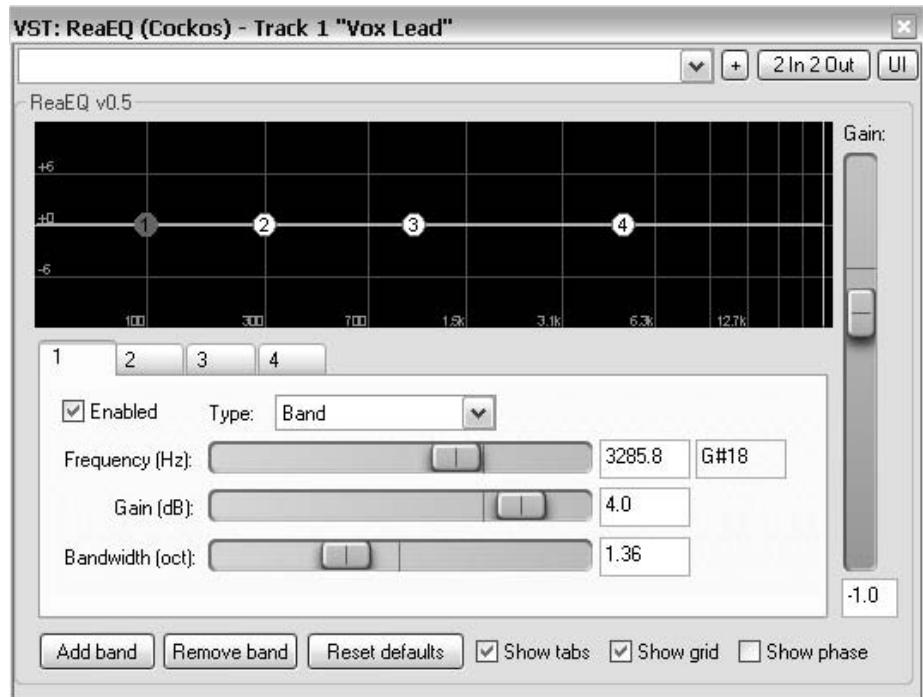


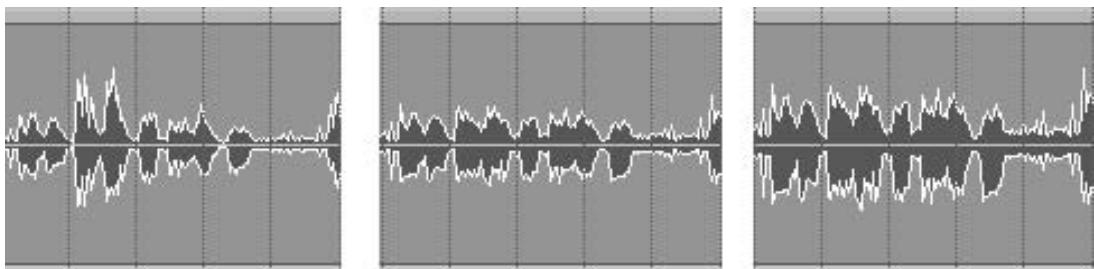
Figure 2.70 An example of using EQ to shape sound.

## Day 19: Plug-Ins and FX—Compression

As you might have guessed, the whole topic of using plug-ins to manipulate the dynamics of your recordings is an enormous one. And again, for today we will need to be contented with a simple introduction that will help us to understand at least the first principles.

### Manipulating Dynamics

Every recorded item has its unique pattern of dynamics, representing the variation of, among other things, the volume range of that item. Dynamic manipulation tools, such as a compressor, are used to make adjustments to that range. Typical examples can be smoothing out the variations between peaks and troughs in the volume, as well as possibly raising or lowering the overall volume of the compressed item. The three parts of Figure 2.71 illustrate how compression affects the dynamics of a recorded item.



**Figure 2.71** Applying compression to a recorded audio item.

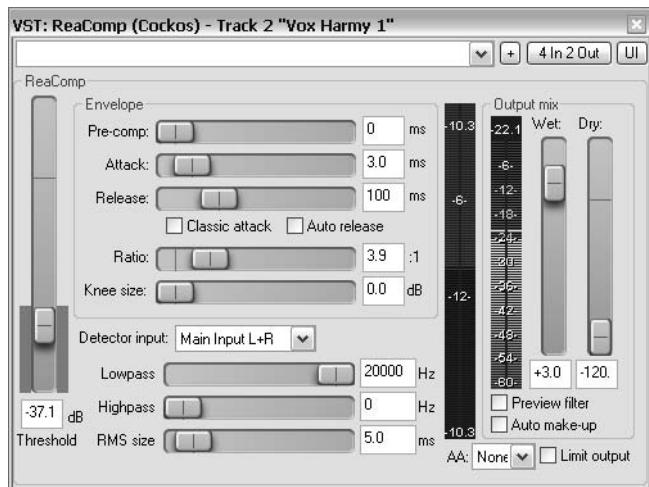
The illustration on the left represents a portion of a vocal recording. Notice that there is substantial variation between the louder parts and the quieter parts.

In the second illustration, the recording has been compressed so that the overall volume is much more even. However, you can see that the louder passages are now a lot quieter.

In the third illustration, not only has the item been compressed, but the volume has also been raised so that now the entire passage is set approximately at the level of the loudest part of the original recording.

In this example, the compression has been applied quite heavily, such as you might do for a TV or radio commercial. In other cases, you frequently will find that compression is often better used with a defter touch to preserve much of the original dynamics.

Like many plug-ins, a compressor includes a large number of parameter controls. Figure 2.72 shows ReaComp, a compressor that is supplied with REAPER.



**Figure 2.72** The ReaComp interface.

For this introductory lesson, you need to concentrate only on three of the parameters and their controls.

- **Threshold.** This is the vertical fader on the left. This setting determines the volume at which the compressor will become engaged. For example, a threshold setting of -15 dB would ensure that compression will only be applied when the signal exceeds -15 dB.
- **Ratio.** This determines the severity with which the compression will be engaged. For example, a threshold of -15 dB and a ratio of 4:1 means that for every 4 dB above -15 dB that is present in the dry signal, an increase of only 1 dB will be allowed. Thus, a -15 dB signal would remain at -15 dB, a signal of -11 dB would be reduced to -14 dB, and so on.
- **Wet.** This allows you to adjust the overall volume of the processed (wet) signal. You can raise or lower it by whatever constant number of decibels you specify.

Let's now work through an example to show you how this is done.

1. Open the file RosesBloom 18.RPP and save it as RosesBloom 19.RPP.
2. We are going to apply some compression to the vocal harmonies. Select Track 2, Vox Harmy 1 and click on the FX button for that track. This will open an FX window.

3. Select the FX category Cockos, then double-click on ReaComp (Cockos)—*not* ReaXComp (Cockos)—to add this plug-in to this track.
4. Don't be overwhelmed by the number of faders and parameters shown in this window. We will focus on just three of them. For now, ignore all the others.
5. Set the Threshold to about -37 dB, the ratio to about 4:1, and the Wet gain to about +3 dB (as shown in Figure 2.72). This is intended to have the effect of first pushing down the vocal peaks quite substantially and then lifting up the overall level of the compressed signal so that it can be distinctly heard sitting behind the lead vocal.
6. Also insert ReaComp into the Vox Harmy 2 track (Track 3) and experiment with getting the levels for these three parameters right to achieve a similar effect with this track.
7. Save the file.

---

**Note:** You might be wondering why your compression is affecting the *sound* of the media item but not its *appearance*. This is because you are not destructively changing the recorded item itself. What you are doing is changing the sound that is sent to the speakers. If you wish to create a media item with the compression actually embedded in it, you can do so by right-clicking over the item and choosing Apply Track FX to Items as New Take (Mono) from the context menu.

---

## Assignment

Experiment with the application of dynamic compression on any of the tracks in your other projects, especially on those tracks that you wish to sit more in the background of your mix.

At this stage don't be too ambitious, but don't be too worried about making mistakes, either. That's how you'll learn!

The golden rule when using dynamic manipulation plug-ins is that when in doubt, too little is usually better than too much!

## Day 20: Plug-Ins and FX—Delay

This category of plug-ins is in many ways the most interesting to experiment with. Time manipulation plug-ins typically include delay, chorus, and reverb. They work generally on the principle that any manipulation of the way in which your recorded material

behaves in relation to time can change the way it sounds, particularly to give it more atmosphere or a greater feeling of space.

## Manipulating Time

Time manipulation plug-ins can create quite dramatic effects and for that reason should be used sparingly and cautiously, at least until you have become more used to them and you have grown in confidence.

Later in this foundation course, we will take a look at reverb. Today, we will be working with a simpler example of such a plug-in—delay. You will see how, by adding a little delay to a lead vocal, you can appear to fatten it up as well as give it more space. This application is, of course, only one example of many that I could have chosen. We will be using the Cockos plug-in ReaDelay, which is supplied with REAPER (see Figure 2.73).

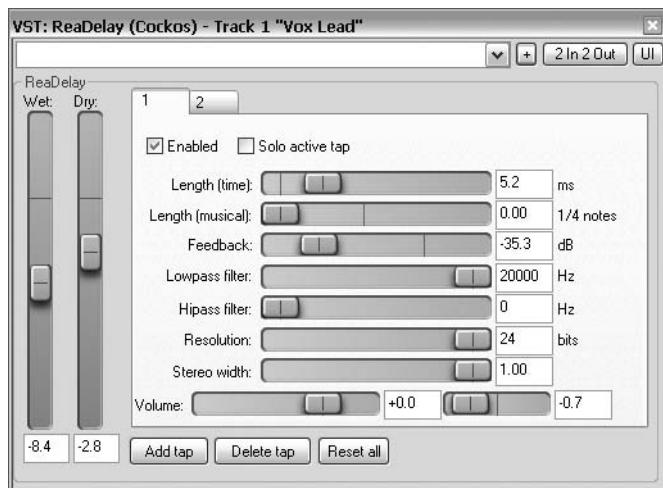


Figure 2.73 The ReaDelay interface.

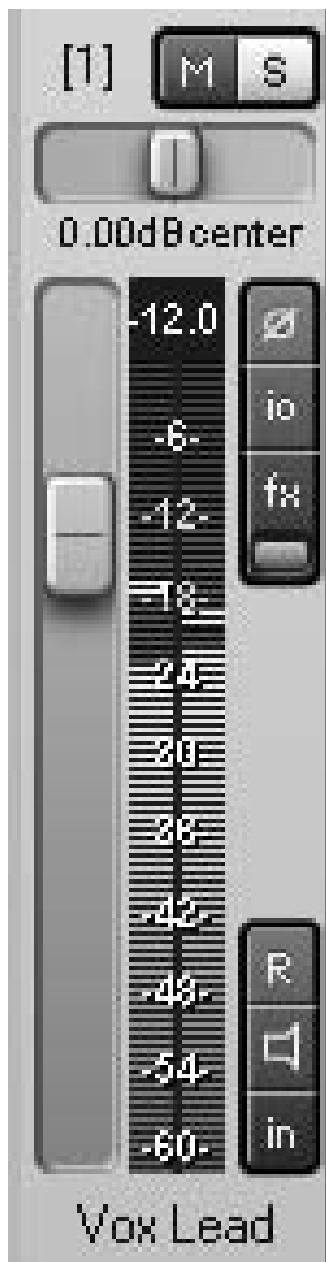
As with the other plug-ins you have encountered, ReaDelay incorporates a number of parameters that can be used to help create the effect you want. Also as with the other plug-ins, we'll be sticking to the basics, focusing on only a few of the most basic controls.

- **Wet and Dry.** These two settings between them determine how much of the delayed (wet) signal will be mixed with the original (dry) signal.
- **Length (Time) and Length (Musical).** You can use either of these to set the amount of time by which the wet signal is delayed.

- **Feedback.** By feeding a portion of the delayed signal back to itself, you can increase the effect. Go easy on this one!
- **Pan.** This control is actually not labeled. It is the fader in the bottom-right corner, next to the volume fader. It is used to send the signal more to the left or the right, as you wish.

Let's explore some of the ways in which we can use delay with a vocal track.

1. Open the file RosesBloom 19.RPP and save it as RosesBloom 20.RPP.
2. Open the FX chain for the Vox Lead track. Display the selection of Cockos plug-ins and double-click on ReaDelay to add this plug-in to this FX chain. If you completed the earlier example, it will appear immediately after ReaEQ.
3. A delay of less than about 7 ms cannot be recognized by the human ear as a separate signal. Instead, it will make the original signal appear fuller. Set the Length (Time) to 3.5 ms, the Length (Musical) to 0.00. Usually, you would want to use one or the other of these controls rather than both.
4. Add a little feedback to your signal, but no more than about -30 dB. Set the Wet fader to about -8 dB and the Dry fader to about -2 dB or -3 dB, as shown in Figure 2.73. Move the Pan fader left to about -0.7.
5. Now click on the Add Tap button. This creates a second page of settings, identical to the first.
6. Change the Length (Time) for this second page to 7.0 ms and the pan setting to +0.7 (toward the right).
7. Play the song with your ReaDelay plug-in enabled. If the vocal seems a little too loud or too quiet, adjust the Wet and Dry faders until it sounds about right. Play it with the plug-in bypassed. Compare the two. The differences might be quite subtle, but the delay-enabled track should sound a little fuller.
8. The signal coming from one speaker is being fractionally delayed more than the other. This can be confirmed by looking at the VU meter for this track in the mixer. This effect is shown in Figure 2.74. With ReaDelay enabled, the levels on the left and right channels will behave quite differently, even though the track itself is panned dead center.
9. If you wish, experiment with modest adjustments to your ReaDelay settings. For example (just for the fun of it), if you gradually increase the length of the



**Figure 2.74** In this example we have used delay to send a different audio stream to each speaker, even though the track itself is panned dead center.

Delay setting on either page (or both pages) sufficiently, eventually you will be able to hear a distinct second vocal signal trailing behind the first.

10. Save the file.

## Assignment

1. Experiment with adding a little delay to the Slide Guitar track in this project to see whether you can improve its sound. Save and close the file when you are finished.
2. Experiment with the gentle application of delay to any track or tracks in any of your other project files where you think it might be appropriate.

## Day 21: Submixes and Folders

In today's lesson you will learn how you can use submixes within your projects. Submixes are very commonly used to enable you to work with, and make changes to, a whole group of tracks at once. Within REAPER, you can use folders to do this.

### What Is a Submix?

*Submixing* is a common practice that you are likely to find useful during the post-production stages of your projects. A submix is a collection of individual tracks that are arranged together in such a way that you can manage them in some ways as if they were a single track.

Suppose, for example, that you have several individual tracks, each containing a vocal harmony. By grouping them into a submix, you can ensure, among other things, that:

- Once you have the balance of volume and panning right between the different tracks, you can make adjustments to the overall level of the submix using just one control.
- If there is any plug-in (for example, compression or delay) that you want applied to all the tracks in the submix, you can apply a single instance of that plug-in to all of the tracks rather than having to use a separate instance for each individual track.

This will become much easier to understand if you work through an example.

1. Open the file RosesBloom 20.RPP and save it as RosesBloom 21.RPP. We are going to create a submix for the two vocal harmony tracks. In REAPER, the

easiest way to do this is to use a folder. Unfortunately, not all DAW programs have this facility.

2. Select Track 1, Vox Lead.
3. Press Ctrl+T to insert a new track immediately after the current track and, of course, immediately before the first of the Vox Harmy tracks.
4. Name the new track Harmonies. We are now going to make this into a folder.
5. Identify the Folder button for this track (see Figure 2.75). This is the first button to the immediate right of the track name.



**Figure 2.75** A project organized into folders. Notice that tracks contained within folders are nested.

6. Click once on this button. Notice that the track number is now replaced with a three-bar symbol, indicating that this is now a folder. Notice also that all the tracks below this folder are now indented. At present, all five of these tracks are included in the folder.
7. We want Track 4, Vox Harmy 2 to be the last track in the folder. Click on the Folder button for this track.
8. The folder icon for this track will turn red to indicate that this is the last track in the folder. Notice that the tracks below this one are no longer nested.

9. Play the song. As you do, experiment with adjusting the volume fader for the new Harmonies folder. You should notice that the volume of both of the individual harmony tracks is affected.
10. Experiment with adjusting the pan setting for this folder. Notice that both the harmonies will move together to the left or right when you do this.
11. Add the plug-in ReaDelay to the FX chain for the Harmonies folder. Set the delay time to about 15 ms, the musical length to 0.0, and the Wet level to about –5.0.
12. As you play the song, the delay will be applied to both the vocal harmony tracks.
13. Save the file.

## Assignment

1. Create a folder called Instruments to hold the two guitar tracks and the slide track for this project. Make the Slide Guitar track the last track in the folder.
2. Apply a little gentle compression to your Instruments folder (adjusting the volume if necessary) to help the instruments sit behind the vocals just a little in the mix. Note that the volume track fader will be applied after any plug-ins in the track’s FX chain, not before it. In this case, this means that the track’s volume fader controls the volume of the signal after it has been compressed, not before. There are ways to adjust a track’s volume pre-FX chain, but we won’t be looking at that right now.
3. Now open one of your other project files, perhaps your Happy Birthday project. Investigate and explore how you might be able to make use of a folder or two in this project.

## Day 22: Basic Routing—Sends, Receives, and Busses

Routing is essentially the ability to send an output signal from one track to another. This is one area in which REAPER excels. Most DAWs allow you to create and use busses for this purpose. We’ll look at an example of this shortly. With REAPER, as well as being able to do this, you can also use sends and receives to mix an output signal from any track with the recorded items on any other track. That probably doesn’t make too much sense right now. Don’t worry; we’ll work through a couple of fairly basic examples.

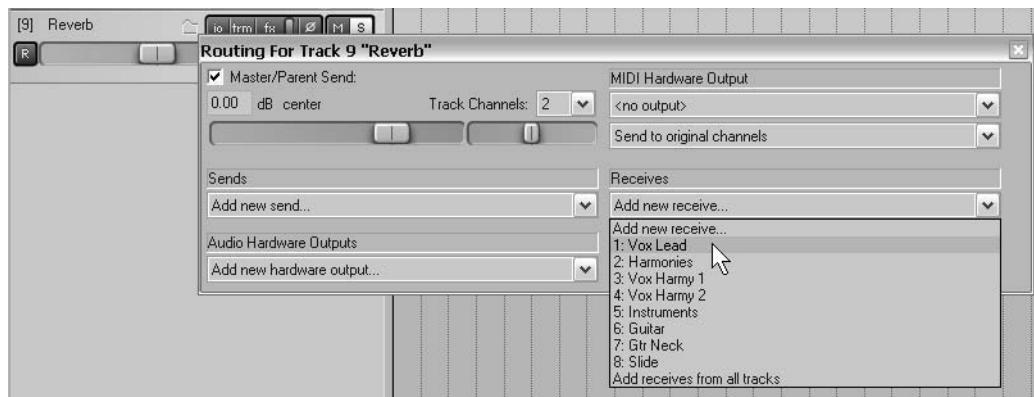
### Creating and Using Busses

In this example you will create a track called Reverb and insert a reverb plug-in into its FX chain. You will then create sends from your Vox Lead track and your two folders to

allow you to add a controlled amount of reverb to these tracks. Before you do, just a word about some of the terms that we will be using. A send and a receive are the two opposite ends of the signal flow. For example, creating a *send* from Track 1 to (say) Track 9 is exactly the same as creating a *receive* at Track 9 from Track 1.

Let's now work through the example.

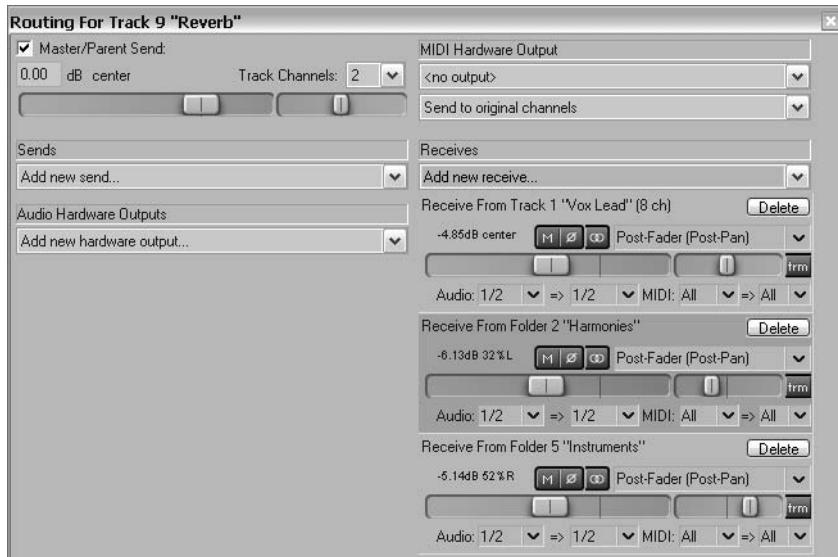
1. Open the file RosesBloom 21.RPP and save it as RosesBloom 22.RPP.
2. In the Track Control Panel, select the last track. If you completed the assignment at the end of yesterday's lesson, this will be the Slide, Track 8.
3. Press Ctrl+T to insert a new track. Name this track Reverb.
4. Into the FX chain of this track, insert the plug-in Cockos ReaVerbate. Display the drop-down list of presets and select Discrete Room.
5. You are now going to create the necessary sends/receives to create a signal flow to enable you to use this track as a reverb bus. Click on the I/O button (labeled *io*) for the Reverb track. This is located just to the right of the track's Folder button. The Routing window for this track will now be displayed.
6. Display the Add New Receive drop-down list (as shown in Figure 2.76) and select 1: Vox Lead to add a receive from this track.



**Figure 2.76** Creating sends and receives in REAPER.7. Repeat this procedure to add two more receives, one from each of the two track folders, Harmonies and Instruments. By now, Track 9 will have three receives attached to it.

7. Play the song. You should be able to notice the difference with the reverb added. Perhaps it sounds a little too mushy.

8. Adjust the levels of the faders in the receives until the amount of reverb added sounds about right. If you wish, you can also experiment with the panning. One possible outcome is shown in Figure 2.77.



**Figure 2.77** Adjusting volume and pan settings for sends/receives.

9. The purpose of reverb is to help create the feel of a live venue, where the sound of the different instruments and voices bounces around the room, adding a feeling of depth and space. It is a fairly complex topic. In this course we are just touching the surface, but feel free to experiment if you wish.
10. Save the file when you have finished.

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**Note:** You can assess the effect that your reverb is having on your overall sound by toggling the Mute button for your Reverb track on and off.

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Similarly, if you want to check on just how much is being added to each track, you can solo the Reverb track while the song is played.

### Using Direct Sends

The ability to use direct sends from one track to another is available in very few DAW programs. This is a shame, because it can be a very useful feature. One example of when you might wish to use it is to give a boost to a track that seems to be getting a little

buried in the mix. Another example might be to add more impact to an otherwise rather weak track. Happily, you can do this with REAPER.

1. Suppose you want to make the Vox Harmy 1 track a little more evident in the mix, but without actually making it louder. Simply increasing the volume would not be appropriate—that could just lead to it drowning out some other track. Instead, you can create a send to add part of its signal to another track, say Vox Harmy 2.
2. Display the Routing window for the Vox Harmy 1 track.
3. Add a send from here to the Vox Harmy 2 track, with volume at about -6 dB and pan set at about 50% left.
4. Now play the song with the Vox Harmy 2 track soloed. Even though you have soloed the track, you will hear the Vox Harmy 1 track behind it. This is the signal from the send you have just created.
5. Unsolo the track and play the song again. If you wish, adjust the volume and pan settings of the send to suit your preference.
6. Save the file.

## Assignment

Experiment with creating a reverb bus for your Happy Birthday project. If you wish, create a direct send as well, from one track to another.

Believe it or not, today's lesson has barely scratched the surface of REAPER's routing capabilities. Understanding and unlocking these capabilities can be one of the most important keys to being able to create stunning mixes.

## Day 23: Consolidation Exercise

Soon we will be heading for the home stretch! Before we do, it's worth taking a day out of our schedule to contemplate and consolidate.

We won't be introducing any new topics today. You've got a free hand.

It might be a good idea to spend today going back over some or all of the many topics we have covered in the last three weeks or so. As you do, ask yourself a number of questions, such as:

- Is this topic important to me? For example, almost everybody who is going to get involved in home recording will need to understand about soundcards and how they interact with software. However, not everybody will necessarily use sampling or MIDI editing.

- How well do I understand this? Identify those areas where, after this course, you feel you would definitely like to dig deeper. These might include topics such as routing or EQ, for example.

Or then again, you might prefer to simply take the day off and go to the beach!

## Day 24: Fun with Pitch and Play Rates

Today you are going to learn what for many people are two of the most interesting aspects of working with digital audio—the ability to make changes to pitch and play rates. Let's take a look.

### Pitch Modification

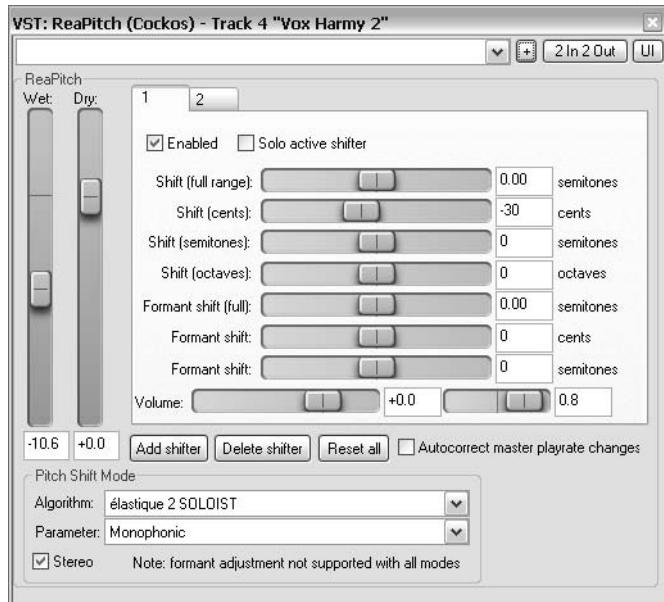
Here are two examples of circumstances in which you may wish to modify the pitch of a recorded item:

- To create one or more copies of an item, each with a degree of pitch shift. This can be used for double-tracking and other similar devices.
- To make a correction to the odd note here and there that might be out of pitch.

REAPER provides two pitch-modification plug-ins for use with your tracks. ReaPitch is ideal for creating double-tracked or fattened vocal harmonies, while ReaTune can be used for pitch correction.

The exercise that follows offers a simple example of how you can use ReaPitch to fatten up a vocal.

1. Open the file RosesBloom 21.RPP and immediately save it as RosesBloom 24.RPP.
2. Select the track Vox Harmy 2 and solo it.
3. Display the FX window for this track and add the Cockos plug-in ReaPitch.
4. First you need to select a pitch-shift algorithm. Different algorithms give better results for different instruments. The algorithm elastique 2 SOLOIST is especially good for use with vocals. Select this, as shown in Figure 2.78.
5. Set the Wet fader to about -10.0 and the Dry fader to 0.0. Use a Shift (Cents) setting of -30 and pan to the right at about 0.8. These settings are displayed in Figure 2.78.



**Figure 2.78** ReaPitch, REAPER's pitch-shifting plug-in.

6. Click on the Add Shifter button to add a second pitch-shift page. Change the Shift (Cents) setting to +30 and the pan setting to -0.8 (left).
7. Now play the song. You will probably wish to lower the volume fader a little on the Vox Harmy 2 track. Experiment with variations in the various settings.
8. Save the file.

---

**Note:** This example has been constructed to demonstrate a very simple example of how this pitch modification feature can be used. It introduces you to the concept of pitch shifting, but falls far short of illustrating REAPER's full capabilities in this area.

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## Changing the Play Rate

Here's a fun thing to do!

REAPER allows you to change the play rate of your project even after recording it, either to speed it up or to slow it down. It allows you to preserve pitch while doing so.

Before doing so, make sure that the Preserve Pitch option is enabled. You do this by right-clicking on the Transport Bar, then choosing Play Rate from the menu, then

Preserve Pitch in Audio Items when Changing Master Playrate (as shown in Figure 2.79). The algorithm used will be whichever is specified in the project settings. We do not really need to worry about this for now.

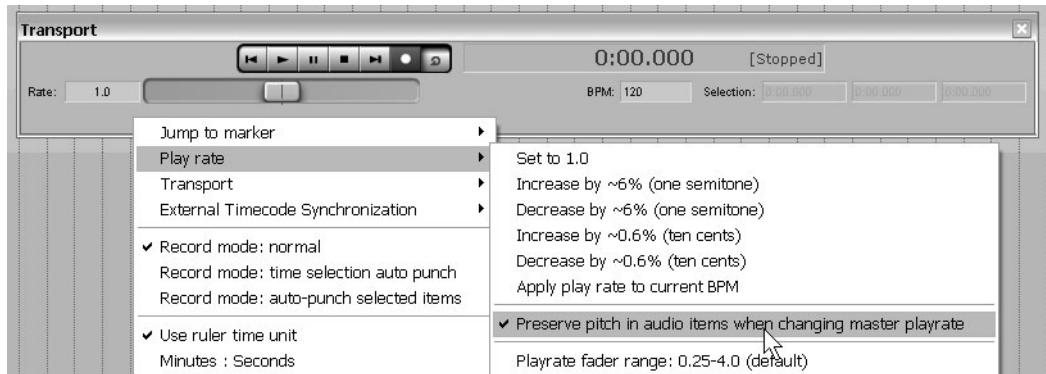


Figure 2.79 Preserving pitch while adjusting the play rate.

In this next example, we will experiment with changing a project's play rate.

1. Make sure that you have enabled the option to Preserve Pitch in Audio Items when Changing Master Playrate (refer to Figure 2.79).
2. Play the song.
3. Slide the play rate fader on the Transport Bar a little to the left. In Figure 2.79 this is labeled simply Rate and is located on the far left. Notice that as the value falls below 1.0, the song slows down.
4. Slide the play rate fader on the Transport Bar a little to the right so that the value is set at about 1.05. Notice that the song speeds up.
5. Double-click on the play rate fader. The play rate is restored to 1.0.
6. Save the file.

## Assignment

Time to relax and enjoy yourself! Using your Happy Birthday project, explore some of the possibilities open to you using pitch shifting and changed play rates.

---

**PITCH SHIFTING MEDIA ITEMS** I'll just mention here that REAPER makes it easy for you to change the pitch and/or play rate of any individual media item or group of media items.

If you select any item, then right-click and choose Item Properties from the menu (or just press F2), the Item Properties dialog box is shown. There are a dozen or more properties that you can modify, and we certainly will not be examining all of them here. Just notice for now that Playback Rate and Pitch Adjust are two such properties.

I should also mention REAPER's dynamic splitting capabilities. This is a very powerful tool, especially when you are creating tempo-based music. You can use dynamic splitting to automatically split any recorded passage into a large number of samples, each of which can be used individually and time- or pitch-shifted at will.

If this is important to you, you should check the feature out when evaluating alternative software solutions. For example, Cakewalk's SONAR includes Audio Snap, and Digidesign's Pro Tools includes Beat Detective, both of which have a similar capability.

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## Day 25: Making Life Easier—Markers and Macros

Good software should be capable of being adapted to your precise needs so as to make it faster and easier for you to use. Nowhere is this truer than when we are speaking of DAW software. This is because DAW programs, as you are probably coming to realize, tend to be vast and complex in their architecture and nature. It can be a big help to you to find ways to get around quickly and to make and use shortcuts for those features you use the most.

Your DAW should be capable of being customized in any number of ways, including (but certainly not confined to) the following:

- **Track templates.** These enable you to store any number of standard sets of track settings, including, for example, any favorite FX chains. These make setting up and creating new project files easier and quicker.
- **Screen layouts.** After a while, you'll develop your own individual preferences as to how you like the various items and views laid out on your screen at different times in a project's lifecycle. By saving and recalling your favorite screen layouts, you can switch quickly from one to another.
- **Markers and regions.** These make navigation within a project much easier and quicker.
- **Macros and custom actions.** Being able to string together into a single keystroke those particular sequences of commands and actions you use the most often is one of the most important customization features you will need from your DAW.

In this section I'll introduce you to just two of these—markers for fast navigation and macros (or custom actions) for fast command execution. The main reason for this is to give you the flavor of how useful customization can be.

## Markers

Markers act like bookmarks in a project file. They are especially useful for indicating the beginning and end of key passages in your songs. To create a marker, you can select any point along the Timeline, then use the Insert, Marker command or simply press the shortcut key M. You can also create markers by doing this while your project file is playing.

Once you have created a series of markers, you can easily jump from marker to marker with a single keystroke. Figure 2.80 shows a project file with two markers.



**Figure 2.80** A project file with markers.

In this next example, you will learn the most basic facts about making and working with markers.

1. Open the file RosesBloom 22.RPP and save it as RosesBloom 25.RPP.
2. Play the song. Suppose there is a passage of about 30 seconds commencing at about the 36-second mark in which you are especially interested.
3. When the song reaches 36 seconds, press M to create a marker at that point. Press M again at the 1 minute 5 seconds mark. This creates a second marker, as shown in Figure 2.80.
4. Stop playback. Press the number 1 key, and the play cursor will jump straight to your first marker.
5. Recomence playback. After a few seconds, press 1 again. Notice that the play cursor immediately jumps back to that marker and recommences playback from that point.
6. Double-click on the Timeline between the two markers. Be sure to do this along the area that actually shows the times, just below the area where the marker flags

are displayed. This creates a loop. The looped section will now play continuously until you stop it. One way to stop the looping is to press the Escape key.

7. Double-click on the red numbered indicator for either marker. This causes a box to be displayed. You can now give this marker a name if you wish.
8. Save the file.

## Macros and Custom Actions

The more adept you become at using REAPER, the more you will wish to use custom actions (or *macros*, as they are also called). The idea is quite simple. If there is a sequence of actions and/or commands that you find yourself regularly using, you can bind that entire sequence to a single key or keyboard combination.

You have already noticed that REAPER comes with many key commands built in. For example, these include Ctrl+S (File, Save command), M (Insert, Marker command) and Ctrl+L (Options, Show Takes in Lanes). In this section you will be introduced to using the Action List editor to create your own custom actions.

Custom actions can vary from the very simple to the very complex and everything in between. Today we'll take just one simple example.

We have already made a fair amount of use of the command Split Items at Time Selection. For example, we have used it when muting passages of a media item, when deleting part of a media item, and when splitting up a track that consists of multiple takes. Let's now assign a keyboard shortcut for this command.

But let's be a little bit more clever than this. Let's also get the macro to automatically select the item under the mouse cursor (without our needing to click) before then splitting the item and finally deselecting the item. We can then assign a single letter (for example, K) to execute all three commands in this macro.

In this example, I'm assuming that your file RosesBloom 25.RPP is still open.

1. From the REAPER menu, choose the Actions, Show Action List command. This causes the Actions window to open.
2. Click on the New button within this window to define a new custom action.
3. In the Custom Action Name box, type a name, such as Split Item.
4. You can see that the list of possible actions is very long. You need to filter it. In the Filter field, type **mouse cursor** (as shown in Figure 2.81). This filters the list to show only those actions that contain the words specified.

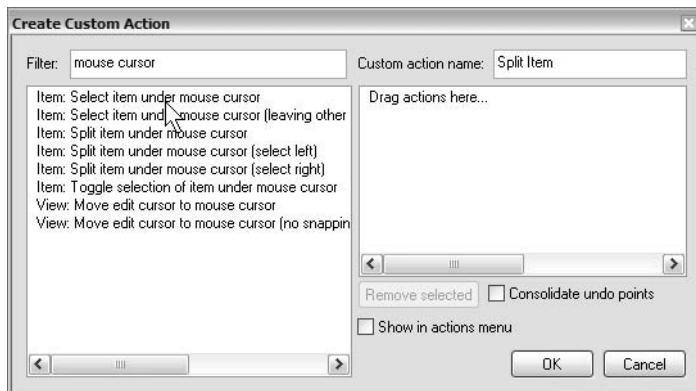


Figure 2.81 Selecting actions for a custom action macro.

5. Double-click on the action Item: Select Item Under Mouse Cursor to add this to your macro.
6. Remove the text from the Filter box, and in its place, type **split time**. This should cause the action Item: Split Item(s) at Time Selection to be shown. Double-click on this to add it to your macro.
7. Remove the text from the Filter box, and in its place, type **unselect**. The filtered list should include the action Item: Unselect All Items. Double-click on this.
8. Your list of actions should now contain three items, as shown in Figure 2.82.

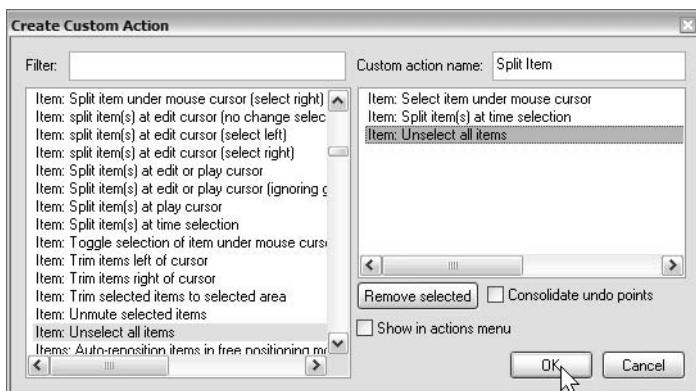


Figure 2.82 A custom action macro with three actions.

9. Click on OK to finish defining your actions.

10. Click on the Add button to define your shortcut.
11. When the Keyboard or MIDI Input window is displayed, press the K key.
12. Click on OK, then Close to close the Actions window.
13. Now make any time selection that you wish along the Timeline. Position your mouse over any media item (no need to click) and press K.
14. The selected item will be split automatically for you.
15. Save the file.

## Assignment

Open your Happy Birthday project file. Identify those places where you might wish to create some markers, and then create them. Save the file.

At this stage, you will probably find the topic of creating custom actions and macros a little too advanced and difficult. However, if you are feeling brave, you can give it a go.

## Day 26: Automation Envelopes

You're nearly ready to mix down your projects for burning to CD or for distribution over the Internet. Just before you do, though, there's one more topic that we need to look at—automation. When you see what it does, you might think that I have left the best until last.

Automation allows changes that you wish to make to any of the parameters of your tracks and FX plug-ins to be remembered and played back with the song. An example of when you might wish to do this could be during an instrument break. You might want the lead instrument to be moved to the center and increased in volume for that passage.

As with most things, automation in REAPER can vary from the quite simple to the rather complex. For example, you can set it up so that as the volume on one track is increased, another track is automatically faded down. Today, however, we will keep to some simple examples.

REAPER, like most DAWs, makes available several methods of going about automating your track and parameter behavior. One commonly used method is to tell REAPER to remember your fader movements and to use them to create an automation envelope. Another common method is to draw this envelope by hand. Today, we'll use the latter method.

## Setting Up Envelopes

If you click on the Envelopes button (immediately to the right of the io button) of any track, the Envelopes window for that track will be displayed (see Figure 2.83).

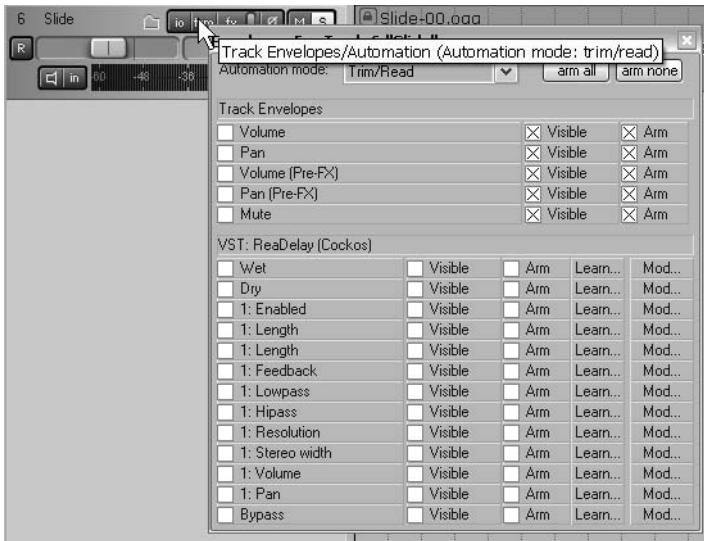


Figure 2.83 The Envelopes window can be used to create automation envelopes for any track.

This window will display all the parameters that can be automated for that track. The first section lists the track envelopes. They will always be present and include track characteristics such as volume, pan, and mute. In the case of volume and pan, these can be applied pre-FX or post-FX (or both).

The second section lists automatable parameters for any FX plug-ins that have been inserted into that track's FX chain.

In this example we are going to first create a pan envelope for a Slide Guitar track so that it will appear to move its position relative to the two speakers toward the end of the song. This is illustrated in Figure 2.84.

1. Open the file *RosesBloom 25.RPP* and save it as *RosesBloom 26.RPP*.
2. Display the Envelopes window for the Slide Guitar track (as shown in Figure 2.83).
3. Select the item Pan so that the check box for the item is marked with an X. Notice that a colored horizontal line is now drawn across the track. This is the pan envelope.
4. Close the Envelopes window.

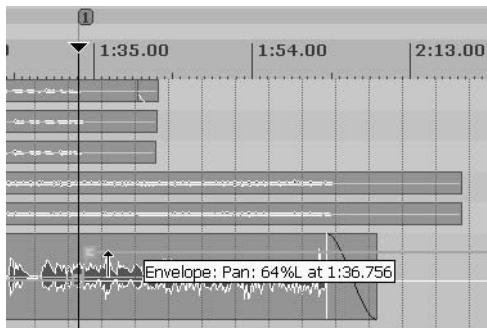


Figure 2.84 A track with a pan envelope.

5. Notice that this track is currently panned something like 65% to the right. We are going to move it toward the center from the 1 minute 33 second mark onward.
6. Position the play cursor at approximately 1 minute 33 seconds and press M to put down a marker at this point. This makes it easier to return to this point when you need to.
7. At the point where the play cursor crosses the pan envelope, click once. This inserts a node (as shown in Figure 2.84).
8. Click on the envelope just to the right of this to insert a second node.
9. Holding down the Shift key, click and hold the mouse on the envelope just to the right of the second node, then drag upward until the display indicates that you are approximately 60% to 65% left (see Figure 2.84), then release the mouse. This message may be confusing at first. Because you started 65% out to the right, by moving back 65% to the left, you will be panning this instrument at or close to the center.
10. Using the same method, add a pan envelope to the Guitar track that is currently panned about 60% left. Change the envelope's pan setting to about 70% right.
11. Play the song from about 1 minute 20 seconds onward. The slide should be clearly heard during this passage coming out from between the two guitar tracks. You can visually check this by observing the behavior of the VU meters for these tracks.
12. Let's now increase the volume slightly on the slide for this passage. Select this track.
13. Position your cursor on the marker that you created earlier. (Remember that just typing the marker number will take you there.)

14. You could open the Envelopes window and select the volume marker if you wished, but as a shortcut you can press V. This causes a volume envelope to be displayed.
15. Using the same method as before, add two nodes to this envelope and raise it by about 2.50 dB.
16. Play the project from just before the marker.
17. Save the file.

## Assignment

Experiment with the use of automation envelopes on one or two of your tracks in your Happy Birthday project.

## Day 27: Bringing It Together in the Mix

The last step that needs to be taken before you can prepare your recorded material for distribution by audio CD or over the Internet is to mix your project down to a single stereo file. You'll be doing that tomorrow. Before doing this, though, recall some of the main post-production aspects that we have been through to get to this point:

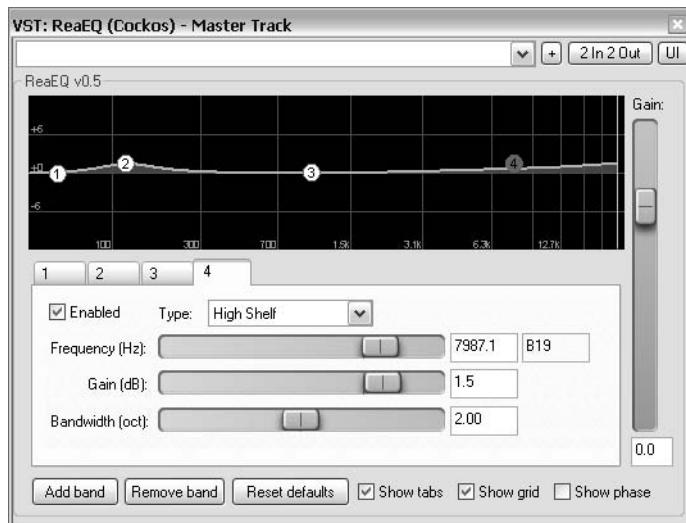
- Using volume and pan faders to achieve a balanced sound
- Editing audio tracks, including take selections and fadeouts
- Adding plug-ins and FX, including EQ, compression, delay, and reverb
- Creating submixes and folders
- Using sends, receives, and busses
- Experimenting with changes to pitch and play rate
- Laying down markers and making macros
- Adding automation envelopes

## The Master Track

There's one more item that needs your attention before you mix down—the Master Track. This is where you pay attention to how your project will sound overall. This step is best left until after you are happy with what you have done with all of your individual tracks.

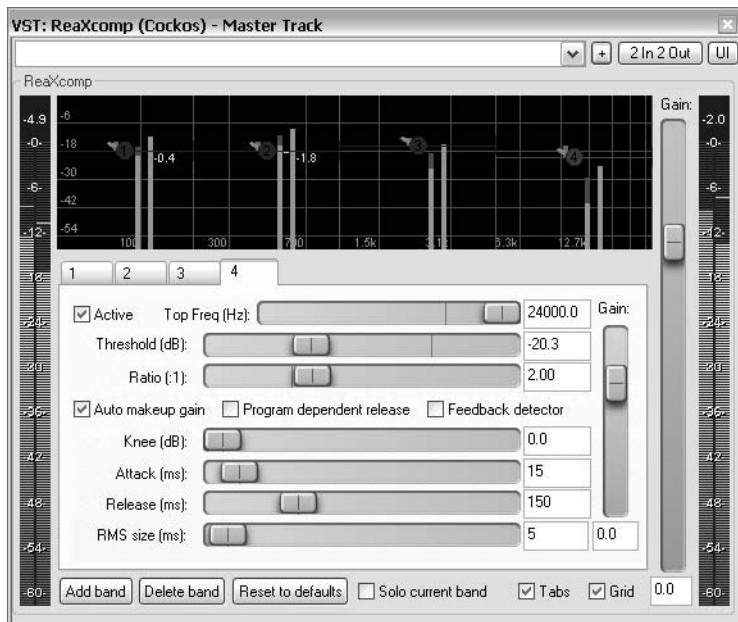
The science (and art) of mastering is a massive topic and, as such, is way too advanced for this foundation course. Today we will content ourselves with looking at how we might be able to beef up our overall mix a little. We will do this using some gentle EQ and some no-less-gentle compression. After these two effects, we will add a limiter to prevent clipping. This topic is definitely an area where it is advisable to tread very lightly.

1. Open the file RosesBloom 26.RPP and save it as RosesBloom 27.RPP.
2. Select the Master Track and open the FX window for that track. If you prefer to work in Track view rather than Mixer view, the key command Ctrl+Alt+M will toggle the display of the Master in Track view.
3. Into the FX chain for the Master, insert an instance of Cockos ReaEQ.
4. Select Band 2. Set the Frequency to about 120 Hz, the Bandwidth to about 1.4 octaves, and the Gain to about 1.5. This should add a little warmth or fullness to the overall mix.
5. Select Band 4. Notice that this is a high-shelf filter. This can be used to add a little air to the mix. Set the Frequency to about 8,000 Hz, the Gain to about 1.5, and the Bandwidth to about 2.00. Also, fade the overall gain control on the right up a little, as in Figure 2.85. Your mix should now sound just a little more lively.



**Figure 2.85** Boosting key frequencies a little on the Master Track.

- Now add to your FX chain an instance of Cockos ReaXcomp. This is a multi-band compressor and a very dangerous weapon if used injudiciously. It works in the same way as an ordinary compressor, except that it lets you apply different settings to different parts of the frequency spectrum. You are about to be shown how to use this cautiously and safely to beef up your mix a little without going over the top. For this purpose, we will accept the default frequency and band settings, as shown in Figure 2.86.



**Figure 2.86** Adding a little polish with some gentle multiband compression.

The secret is to be gentle. The four red dots that you see in the ReaXcomp window represent the threshold for each of four frequency bands.

- Play the song. As you do, pull down the red dot for each of the first three bands just enough to engage the compressor. You'll notice this on screen by a negative yellow number next to the bars. Figure 2.86 shows some examples of this. For example, for Band 1 this is  $-0.4$ , and for Band 2 it is  $-1.8$ . If these never get below about  $-2$  or  $-3$ , chances are that you're in the safe zone. If you wish, experiment also with the fourth band, but you'll likely find it best to leave this alone.
- This process should have added a little overall gain and given the mix a slightly fuller sound. In Figure 2.86, you can see that the incoming signal is peaking at

–4.9, whereas the signal leaving the compressor is peaking at about –2.0. Do not adjust the volume of the fader on the Master Track—that should be left at 0.0 dB.

Hopefully, this example will not have given you the confidence to become too gung ho with the use of multiband compression, but it should have given you an idea of its purpose.

Finally, we are going to add a limiter at the end of the chain. This will serve one important purpose—to prevent the volume of the output from the Master from going above 0.0 dB.

9. Within the FX window, click on the Add button. Select the category JS and in the Filter list (near the bottom), type **limit**. You should see a small number of items displayed.
10. Double-click on LOSER/MGA\_JS Limiter to insert this. By setting the Ceiling to –0.5, you will ensure that the possibility of clipping is avoided. As an aside, you will find that by moving the Threshold fader to the left, you can raise the overall volume of your mix without clipping. If you wish to do this, use the gentlest of touches. If you see a lot of red lights flashing in the plug-in window in areas below the faders, you have probably overdone it.
11. Play the song now, with your Master FX alternately enabled and bypassed. You should notice that with them enabled, the song is much more pumped and out front. By all means, experiment with different settings, but do not lose sight of the fact that one of the objectives of this course is for you to understand what goes into creating a good product—not for you to be able to work miracles at this stage.
12. Save your work when you're finished.

## Assignment

Open your Happy Birthday project and explore the possibilities of improving its output by gently applying some EQ, multiband compression, and limiting to your project's Master Track.

Please remember that the operative word is *gentle*.

## Day 28: Rendering for CD or MP3

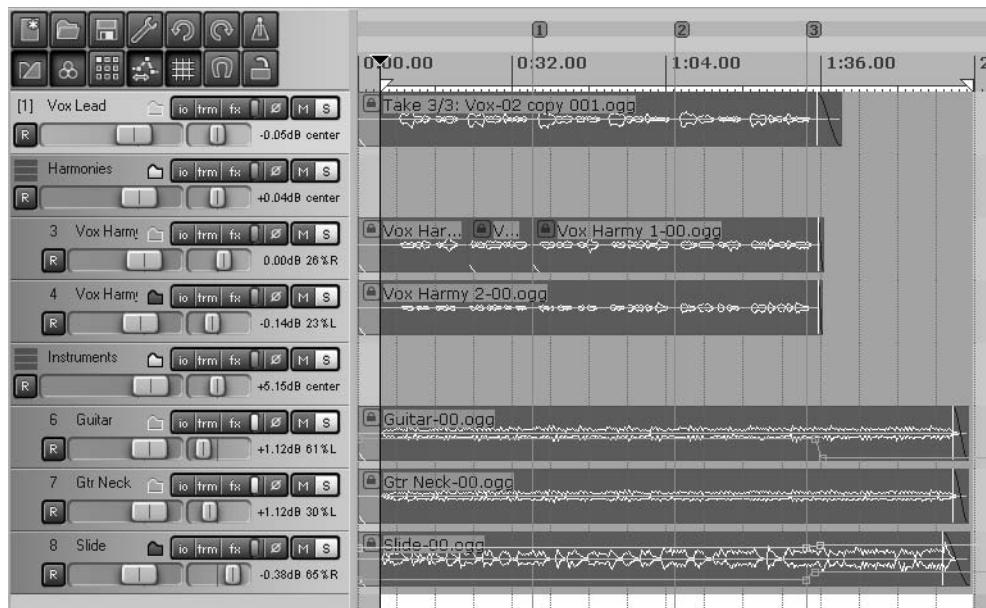
You are now ready to mix your music down either to audio CD or to an MP3 file. Let's do both!

## Rendering to Audio CD

Typically, you would want to render to audio CD when you have a collection of songs, perhaps four, five, or more. REAPER can handle this task with no problem, but in keeping with the spirit of the rest of this foundation course, we'll keep it really simple today. We'll just burn one song to one audio CD.

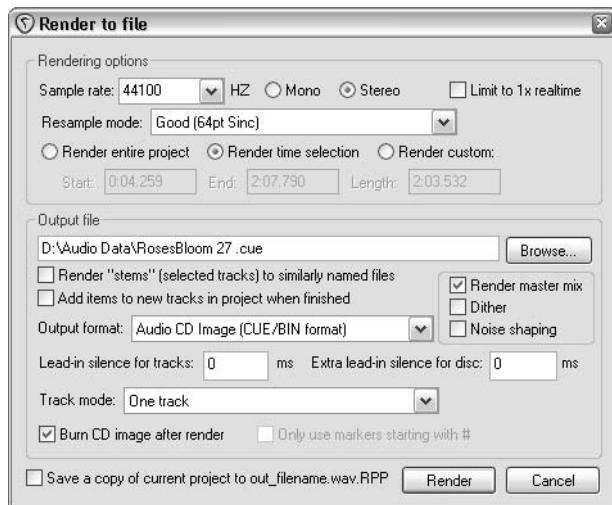
The method required basically consists of two steps. First, you select along the project Timeline the exact area that you want rendered. Second, you give REAPER the command and specifications required to render the project. Of course, before you do this, you should make sure that you have a supply of blank CDs on hand.

1. Have a blank, unused CD ready. Open the file *RosesBloom 27.RPP* and save it as *RosesBloom 28.RPP*.
2. First, you are going to select the area that you want to mix down. This should exclude the periods of silence at the start and finish of the song. Click and drag along the Timeline to select the required area (see Figure 2.87).



**Figure 2.87** Preparing to render a mix. The area to be rendered is selected on the Timeline.

3. From the menu, choose the File, Render command. This brings up the Render to File dialog box (see Figure 2.88).
4. Make sure the Sample Rate is set at 44,100 and that Stereo is selected.



**Figure 2.88** Rendering a project to CD. The typical settings are shown here.

5. Select the option to Render Time Selection.
6. Choose the output format Audio CD Image (CUE/BIN Format).
7. Make sure that Render Master Mix is selected. This ensures that the output of the Master Track (rather than individual tracks) will be mixed.
8. Set the Track Mode to One Track. This is because we have only one song to mix down. If there were several tracks, we would need to use markers in a special way to define the start of each track.
9. Select the option to Burn CD Image after Render.
10. Set the two Lead-In Silence options to 0.
11. Click on the Render button. You will see a graphic as the file is rendered in stereo format. This will be followed by the Burn Image dialog box.
12. Insert your unused CD into your PC's CD drive and wait a few seconds.
13. Click the Burn button. A Burning in Progress message will be displayed. When the burning is finished, the CD should be ejected, and you will be returned to the display of the rendered stereo file.
14. Remove the CD and close the CD drive.

15. Click on the Close button just below the file image on the screen. This will close the dialog box.
16. Close REAPER, saving the changes to your file.
17. Take your CD to a CD player and play it!

## Rendering to MP3

Before you can render to MP3, you need to ensure that the file lame\_enc.dll is present in your C:\Program Files\REAPER folder. If you do not have this file, use Google to find it on the Internet. It can be downloaded free of charge. After you have downloaded the file, move or copy it to the required directory.

The procedure for rendering to MP3 is similar to that used for burning to CD. The main difference is that you will choose a different output format in the Render to File dialog box.

1. Open the file RosesBloom 28.RPP. If after the previous example you saved it with the area to be rendered selected, then this selection will still be active. Otherwise, you will need to reselect the required area along the Timeline, as shown in Figure 2.87.
2. Choose the File, Render command.
3. The Rendering Options group of settings should be the same as before. Select the Output Format MP3 (Lame) as shown in Figure 2.89.

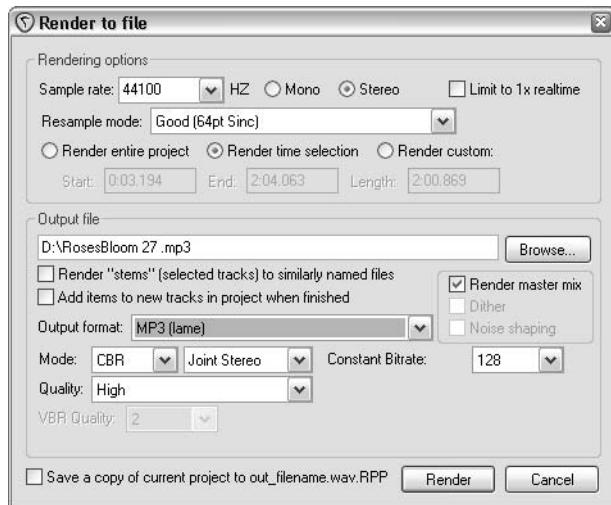


Figure 2.89 Rendering a project to an MP3 file. The typical settings are shown here.

4. Most of the remaining Output File settings together determine the output quality. For example, the higher the Bitrate setting, the better will be the quality, but the bigger will be the file size. The combination of settings shown here (including Joint Stereo) should give a satisfactory result. *Be sure to specify the directory required for your output file.*
5. Click on Render to render the file.
6. When rendering is finished, close the rendering window and exit REAPER.
7. Use Windows Explorer to find your rendered file.
8. You can now play this using a program such as Windows Media Player, or you can transfer it to your MP3 player or email it as an attachment to your friends!

## Assignment

Render your Happy Birthday project to audio CD and to MP3.

## Days 29 and 30: Consolidation Exercise

Congratulations! In just 28 days you have covered an enormous amount of ground and should now have established a solid base from which to progress into the world of home recording.

Before you do so, it's worth catching your breath and spending a couple of days consolidating what you have learned.

Record a new project. Figure 2.90 shows the guitar tabs for "Old MacDonald Had a Farm." You can use this or any other song of your choice (but keep it simple).

### Old MacDonald Had A Farm

Traditional

G                    C            G            D7    G  
Old MacDonald had a farm, e-i-e-i-o  
G                    C            G            D7    G  
And on this farm he had some pigs, e-i-e-i-o  
G  
With an oink, oink here and an oink, oink there  
G  
Here a pig, there a pig, everywhere a pig, pig  
G                    C            G            D7    G  
Old MacDonald had a farm, e-i-e-i-o

**Figure 2.90** "Old MacDonald"—your final assignment!

Record a small number of tracks—four to six in total.

Put into practice those topics and techniques that you have found most useful up to now. You do not have to apply every single technique and topic that you have learned during the last 28 days, only those that seem the most important for your own needs.

Mix the song and render it to CD audio and MP3.

And do it all in two days!

## The Next Step: Your Learning Plan

Over the course of these 30 days, you should have learned quite a lot about home recording. If nothing else, you should have some idea of the extraordinary learning curve that you have taken on. That doesn't mean you should be discouraged, rather that you should be realistic about your expectations and be prepared to give yourself time. There's still a lot more to learn.

I won't pretend that this 30-day course has introduced you to every aspect of home recording that you will ever need to know, but it has introduced you to the core topics. Where you should focus your learning from here depends very much upon your individual needs.

For example, if your main reason for getting into this is to record your choir *a cappella*, then it's not likely that learning more about MIDI editing will be your highest priority right now. On the other hand, you definitely will need to learn a lot more about techniques for recording and mixing vocals. If, on the other hand, you envision yourself making a great deal of use of synthesizers and samplers, then that is an area about which you will want to know a lot more.

To help you assess your priorities, I've included a list of topics in Table 2.2. Use this list as you see fit. One way might be to award marks to each topic in two categories. The first category measures importance. Award a mark from 1 to 5 to each topic, where 1 means that you have absolutely no interest in the subject and 5 means that you really need to understand this topic as much as possible. The second topic is confidence, where 1 means that you are totally confident of your current level of knowledge in this area and 5 means that you feel you know absolutely nothing. This scale of marking might seem counterintuitive, but it yields one benefit. Simply by adding together the two numbers, you will have an immediate overview pattern, which should help you to assess what your learning priorities should be, at least at first. As you get deeper into home recording, those priorities will doubtless change, but at least you will be learning in a

**Table 2.2 My Learning Plan**

Stage	Importance	Confidence
Projects and project files		
Single-track recording		
Multiple-track recording		
Overdubbing		
Punch recording		
Recording multiple takes		
MIDI recording		
Software and hardware synthesizers		
MIDI editing		
Creating and using loops		
Sampling		
Basic mixing techniques		
Editing recorded audio		
Frequency manipulation plug-ins, such as EQ		
Dynamic manipulation plug-ins, such as compressors		
Delay-based plug-ins, such as chorus and reverb		
Using submixes and folders		
Track routing and channel splitting		
Changing pitch and play rates		
Using productivity tools—markers and regions		
Using productivity tools—macros and custom actions		
Automation envelopes		
Managing the Master Track		
Rendering to MP3, WAV, or other format		
Other 1:		
Other 2:		
Other 3:		
Other 4:		

productive, practical way and a logical sequence, rather than just floundering about, wondering what to do next.

Note that this list is confined to issues involving the recording, editing, and mixing processes only. The technical issues that you will also need to understand (such as choosing suitable equipment) will be covered in the next chapter.

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# 3

# What Else Do I Need to Know?

There is surprisingly little technical knowledge that you need to acquire before you can get started seriously on your recording path. The way to go is to follow the same philosophy that we used during the 30-day course. The time to dig deeper is when you feel you need to. Right now, we're going to examine a small number of important issues in more depth. After that, it's up to you.

## Audio Formats

During the 30-day foundation course, you were introduced to two audio formats—WAV format and MP3 format. If you are using a PC (as opposed to a Mac), these are the two formats that you are likely to use the most. However, there are other formats that you may need to use from time to time. Let's have a brief look at some of the formats.

### WAV

WAV files are uncompressed audio, also known as *lossless* audio files. This simply means that the audio stream is recorded so as to preserve all of its original characteristics as closely as possible. Because the files are uncompressed and of high quality, most professional audio engineers use WAV files when working on a PC. This file format is also recognized by all quality audio editing software.

### AIF or AIFF

The *Audio Interchange Format File* is the Mac OS equivalent of the WAV file. It exhibits all of the characteristics of the WAV file, is of equivalent standard, and is undoubtedly the audio format of choice for professional audio engineers when working with Mac equipment.

### MP3

MP3s are digital audio files that are encoded using a lossy compression format. It is important to understand that *compression* in this context is an entirely different process from the compression that we encountered during the 30-day course when we used plug-ins such as ReaComp. Compressing WAV files to MP3 has more to do with

reducing file size. The term *lossy compression* means that when the file is decompressed, the data stored in the compressed file is actually different from the original data. In the case of MP3, those parts of the recorded audio that the human ear has difficulty hearing are actually discarded in the compression process.

The one great advantage of the MP3 format is its drastically reduced file size. This makes it suitable for distribution over the Internet. This is also the format most commonly used to store music in small portable devices, such as the iPod.

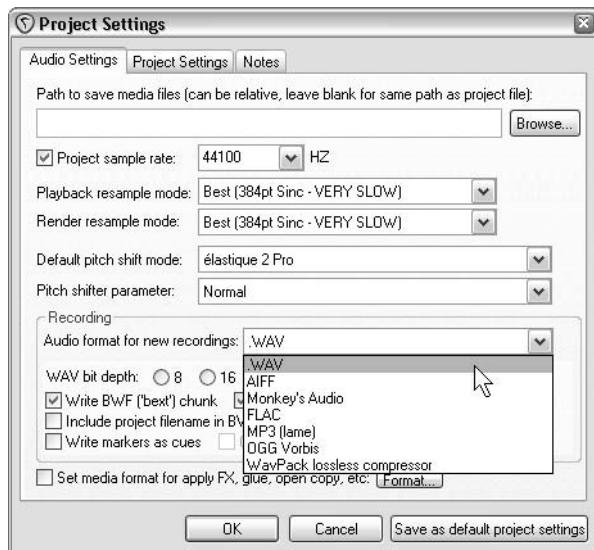
## **WMA**

WMA stand for Windows Media Audio. It is another lossy compression format and was developed by Microsoft. It was introduced as a competitor to the MP3 file format. Although it is true that an increasing number of programs and portable devices will accept files in WMA format, it is still not as widely used as MP3.

## **OGG**

The OGG Vorbis format is an audio codec (encoder/decoder) that is open source and free. Unlike MP3 and WMA, OGG uses lossless compression and is generally considered to deliver a superior audio quality. However, files rendered in this format cannot be played on any of the most popular portable audio devices.

REAPER lets you mix different formats in the same project file if you need to. By default, however, the format used for each new track that you record is determined by the Audio Settings page of your Project Settings (see Figure 3.1).



**Figure 3.1** REAPER’s Project Settings Audio Settings page.

WAV format holds the recorded audio stream more faithfully and with greater accuracy than any of the other formats. For that reason, it makes sense always to use this format for recording. If you later wish to distribute your material in another format (such as MP3), it can be rendered into that format without in any way downgrading the original recorded files.

Of course, it is possible to record in a lower quality format, such as MP3, and later render the material into WAV or AIF format. This practice, however, is not generally recommended. This is because although it is true that the conversion will produce files capable of being burned to audio CD, the sound quality will at best be no better than the original MP3, and certainly not of the same standard as material recorded in WAV or AIF format in the first place.

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**AUDIO FORMATS: KEY POINTS** As a general rule, you should record in WAV format. If you wish to distribute your music in a compressed format, such as MP3, OGG, or WMA, it is an easy process later to render a copy in your required format.

Converting material from MP3, OGG, or WMA to WAV format is possible but will produce an inferior result.

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## Sample Rates and Bit Depth

I spoke briefly about sample rates and bit depth quite early on, when we first looked at the steps you need to go through in order to prepare for recording. We don't need to go into either of these topics too deeply, but it is important that you understand generally what these terms mean and why they are important.

The *sample rate* refers to the number of discrete samples of your music that are recorded each and every second. As an analogy, think of each sample as being similar to a single frame on an old-fashioned celluloid movie. Obviously, the higher the sample rate you use, the more accurate a representation of the performance is recorded. However, you shouldn't simply assume that the highest possible sample rate is necessarily the best. This is for a number of reasons:

- There is a limit to the sensitivity of the human ear. Do you really think that you can tell the difference between, say, a recording sampled at 196,000 and one sampled at 88,000?

- The higher the sample rate, the more disk space is required for your recording and, perhaps more importantly, the greater will be the load placed on your computer's resources during the mixing process. Do you really wish to risk compromising on other matters (such as the use of FX and plug-ins) just for the luxury of using a high sample rate?

While we're looking at sample rates, I need to say something about the *Nyquist Theory*. Put quite simply, this states that the upper limit of the frequency spectrum that can be captured by the digital recording process will be limited by the sample rate. This limit will be set at half the sample rate.

For example, if you record at 44,100 Hz, frequencies above 22,050 Hz will not be recorded. On the other hand, a sample rate of 96,000 Hz will capture frequencies up to 48,000 Hz. This is an area in which there are fiercely debated differing viewpoints. Keep in mind that the absolute upper limit of human hearing is 20,000 Hz. It is indisputable, therefore, that if you record at a sample rate higher than 44,100 Hz, the extra information you are recording will not be able to be heard by your audience, unless perhaps it is a feline or canine one.

However, some argue that those frequencies we cannot hear are important because they can have an effect on those frequencies we do hear. For the time being, just pigeonhole this information, perhaps to return to it several years from now.

The simple truth is that it will be a very long time before there is any possibility of you being able to significantly improve the quality of your recordings by using a higher sample rate. Your choice of equipment, your experience as a recording engineer, and the environment in which you are recording will all be far, far more important. Different sample rates are required for different audio formats. At this stage in your career, keep it simple and use the required output rate when recording. For example, if your ultimate destination is CD, use a sample rate of 44,100 Hz. If it is DVD, use a sample rate of 48,000 Hz.

Bit depth is a different issue altogether. It's important not to confuse the two.

Bit depth is not a specific audio-related issue; it is generic to the computing environment. You probably are already aware that computers convert all information into a binary code (consisting of various permutations of the two digits 0 and 1) for processing. The *bit depth* represents the number of digits available for storing different data values. The greater the bit depth, the greater the number of permutations available and hence the more accurate the representation of (in the case of audio) each individual sample. Of course, it is also true that the greater the bit depth, the greater the file size.

At 16-bit the number of permutations available is 65,536. At 24-bit this rises to 16,777,216. It is therefore irrefutable that at 24-bit you get a more accurate representation than you do at 16-bit. However, as with sample rates, the question arises: *Does it matter?* The answer is that it does, but not for the reason you might think.

It is very doubtful whether the human ear can detect the difference between sounds recorded at 16-bit and sounds recorded at 24-bit. However, there is one very important advantage of recording at 24-bit. *It makes life easier for you.* That is because the extra range of values that 24-bit recording uses makes it much easier to avoid clipping than is the case at 16-bit. (We discussed clipping earlier, by the way, when we first looked at recording.)

Now that you understand about sampling and bit depth, we can look at bit rate. The *bit rate* is the volume of data per second that is required to hold the recorded data. It is calculated as a function of the sample rate and the bit depth. The numbers start getting very large at this point. By themselves they don't mean a great deal; what matters is how they translate into file size. Table 3.1 shows some examples for a recording of just three minutes' duration.

**Table 3.1** Disk Space Required for Storing Audio at Different Sample Rate/Bit Depth Permutations

Format/Bit Depth	Sample Rate	File Size (Three-Minute Mono)	File Size (Three-Minute Stereo)
WAV/16	44,100	15.15 MB	30.30 MB
WAV/16	48,000	16.50 MB	33.00 MB
WAV/16	88,200	30.30 MB	60.60 MB
WAV/16	96,000	33.00 MB	66.00 MB
WAV/16	176,400	61.20 MB	122.40 MB
WAV/16	192,000	66.00 MB	132.00 MB
WAV/24	44,100	23.10 MB	46.20 MB
WAV/24	48,000	24.90 MB	49.80 MB
WAV/24	88,200	45.60 MB	91.20 MB
WAV/24	96,000	49.80 MB	99.60 MB
WAV/24	176,400	91.20 MB	182.40 MB
WAV/24	192,000	99.60 MB	199.60 MB

Notice that the figures in Table 3.1 refer to just a single track. For the whole project you have to multiply those numbers by the number of tracks. For example, a three-minute song made up of 20 mono tracks recorded at 24-bit and sampled at 48,000 would require approximately one gigabyte of storage. Sampled at 96,000, it would require close to two gigabytes.

With storage being a relatively cheap commodity, you might not think this is too important. It can be an issue, however, when you play back and mix your project. All those tracks, along with their FX and plug-ins, are likely to use up a fair amount of your computer's CPU.

When getting started, therefore, I'd strongly recommend that you record in WAV format at 24-bit and 44,100 Hz for CD audio (or if your songs are later to be converted to MP3) and at 24-bit 48,000 Hz if you are recording for DVD audio.

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**SAMPLE RATES AND BIT DEPTH: KEY POINTS** The higher the sample rate and bit depth that you select for recording, the more faithful the recorded version of your music and the larger the file size will be. Once you move beyond a certain level, the differences are not noticeable to the human ear, and any possible benefit may be more than offset by the additional load placed upon your computer in processing this material.

For most purposes, and especially for the novice, first try recording in WAV format at 24-bit and 44,100 Hz.

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## Audio Drivers

Happily, there is not a great deal that you need to know about audio drivers, but what you do need to know is important.

Any device attached to your PC requires drivers, whether it is a screen (monitor), a mouse, a keyboard, or anything else. In most cases you can work quite happily without needing to be aware of this. With audio drivers, it is just a little bit more complicated. There are three main things you need to be aware of:

1. Every individual make and model of soundcard requires its own drivers. When you physically install a new soundcard into your computer, it is important that you also install the correct drivers. In most cases, the drivers should be installed before the soundcard is physically connected to the PC, but you should always follow the manufacturer's instructions before you do anything.

2. Like any other piece of software, drivers will contain bugs. There's no need to be paranoid about this. When you consider that the soundcard has to work with countless pieces of software and along with any of an almost infinite number of possible permutations of other hardware devices, it isn't surprising that from time to time problems arise. You should regularly check your soundcard manufacturer's website for bug fixes and performance updates.
3. Whether you like it or not, in most cases you have a choice of which set of drivers to use with your soundcard. This is because, depending on the purpose for which it is being used, one set of drivers might be more suitable than another. For serious home recording purposes, you should always select either Steinberg's ASIO or Microsoft's WDM Kernel Streaming drivers. These are more resource intensive but also vastly superior in their capabilities to consumer-level systems, such as DirectSound. Much, but not all, DAW software can be made to work with either ASIO or WDM drivers, but most also have a preference for one or the other. Cakewalk, for example, recommends the use of WDM drivers with its software; REAPER prefers ASIO (see Figure 3.2). That doesn't make one system of drivers inherently better or worse than the other; it just means that which system you should use will primarily depend on which DAW you are using.

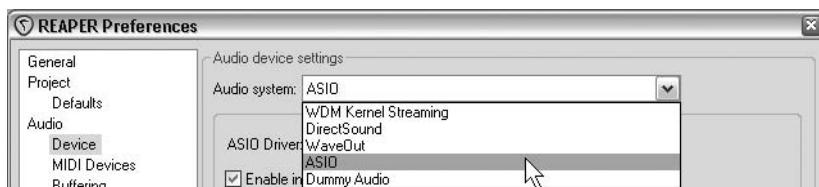


Figure 3.2 Selecting audio drivers in REAPER.

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**AUDIO DRIVERS: KEY POINTS** For home recording you should always use the ASIO or WDM Kernel Streaming drivers that are supplied with your soundcard. Which of these two you should use will depend on your choice of DAW.

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## Impedance

In this book, I am trying to keep the amount and level of technical material to an essential minimum, which means that I cannot afford to ignore the issue of *impedance*. We shall not be examining the topic in depth; instead, I'm content to introduce you to the

concept and to give you a basic understanding of what it is, why it so important, and what you need to do about it. In particular, you should arm yourself with an adequate understanding of this topic before you start spending a lot of money on expensive audio equipment.

Why does impedance matter?

Well, have you ever had the experience of being told by a friend something like, “Don’t buy such-and-such brand of equipment. I purchased a brand-new set of their speakers (or headphones, or amp, or mixer, or...), and after two weeks, it blew. I got it replaced, but three weeks later the new one blew as well. They’re just crap.”

Think about it for a moment. Of course, any company manufacturing in bulk will occasionally put out the odd dud or two. But if *all* their equipment is defective, why are they still in business? Maybe there’s another explanation. The most probable one is that your friend needs to learn a little about impedance.

It’s likely that you already have some grasp of the nature of electrical resistance. This is the resistance that a component or a circuit has to electricity being pushed through it. *Ohm’s law* provides us with a simple equation for measuring this:

$R=V/I$ , where  $R$  is the resistance,  $V$  is the voltage, and  $I$  is the current measured in amps. The value of  $R$  itself is measured in ohms.

From this we can go on to look at impedance. *Impedance* is the resistance a circuit puts up not to a direct current, but to an alternating current, such as an audio stream. This, too, is measured in ohms. Not making too much sense? Don’t worry; you’re not going to have to run around calculating for yourself the impedance for various pieces of equipment. This information should always be listed with the manufacturer’s specifications. However, a basic understanding of impedance is necessary for you to be able to ensure that you can send a clean audio stream as required from one device to another, without risking damage to your equipment.

For a purely passive circuit, resistance and impedance would by definition be the same as each other. This might be the case, for example, when the output from an amplifier is sent directly to speakers.

However, when dealing with most audio equipment (such as soundcards, mixing desks, and microphones), we also have to take into account the effect of components such as capacitors and inductors. These introduce one very important modification to our model—impedance will now vary according to the frequency of the signal that is being passed through the equipment.

Manufacturers of audio equipment aim as far as possible to keep the impedance at a consistent level across the entire frequency range. In practice, however, this cannot always be achieved. In particular, you are likely to find that your speakers will exhibit variations in impedance around the resonant frequency of the drivers.

Two terms that we need to come to grips with are *output resistance* and *input resistance*.

The same fundamental principles apply with any signal flow, whether, for example, we are talking about an electric guitar that is being recorded or a recorded guitar track that is being played back, perhaps through headphones. In the first of these examples, the guitar is the output device, and the soundcard is the input device. In the second example, the soundcard is the output device, and headphones are the input device. An even more interesting example is when you are using headphones for input monitoring. In that case, the soundcard serves as both an input device (from the guitar) and an output device (to the headphones) at the same time. Such a device will, of course, have different impedance levels for input and output.

The signal leaving the output device is subject to an impedance, which is the output impedance. If this output impedance were zero, you could impose any load you wanted on the output, and the signal voltage at the output terminals would not be affected. In practice, however, zero output impedance is not achievable.

At the other end of the chain, every input device also exhibits impedance. This is its input impedance. The output impedance is also known as the *source impedance*, and the input impedance as the *load* impedance. From here on in, it starts to get a little more complicated, so be prepared to take your time over this. Read through this section several times if you need to.

At this point I should also say a word about *unbalanced* and *balanced* interfaces. You may have come across the terms and wondered what they mean. Both types of interface use pairs of wires to carry the signal. In the case of an unbalanced interface, one wire is grounded and carries no impedance. The other wire carries the impedance. With a balanced interface, both wires carry equal impedance. Balanced systems also provide you with greater protection against interference than do unbalanced ones.

When you connect an output to an input, a source (the output) is connected to a load (the input), and a series circuit is created. This is called a *voltage divider*. The voltage will then be reduced by an amount that is determined by the impedances.

Let's now consider the issue of *impedance matching* when we are dealing with an unamplified audio signal. Theoretically, you might expect the ideal situation to be one in

which source impedance and load impedance equal each other exactly. Unfortunately, in this case you'd be wrong. Here's why.

The main object of the exercise is to transfer the maximum audio voltage from the input device to the output device, not the maximum amount of electrical energy. To this end, it is usual to aim for a load impedance that is five to ten times (or more) higher than the source impedance. Why so?

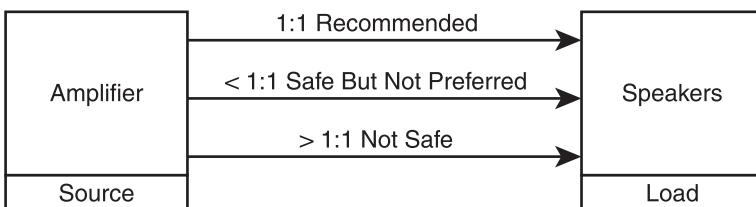
Well, here's one example. Let's return to our playback example. We might want the source (from the soundcard) to be fed to more than one load—for example, two or more amplifiers. We therefore need to ensure that the ratio of input impedance to output impedance is adequate to ensure that the source is not overloaded.

For this reason, you will sometimes find that the documentation for a piece of equipment, such as a soundcard, will include information about not only its input (load) and output (source) impedance settings, but also recommended impedance specs for the load device to which the source is going to be connected. We'll come to some examples of this later in this section.

There are other considerations. For example, taking care in the use of audio cables is recommended. Where a source has to drive a long cable, the impedance can adversely affect the transmission of higher frequencies. Shielded cable exhibits capacitance, which together with the impedance can create a low-pass filter that attenuates top-end frequencies. This undesired effect can also be alleviated by the use of high-impedance inputs with low-impedance outputs.

Let's now pull this theory together and take a look at how it might work out in practice.

#### **Impedance Summary: Amp to Speakers**



**Figure 3.3** Impedance matching an amplifier to speakers.

When speakers are matched to an amplifier, the load impedance should ideally be as close as possible to the source impedance. If the speaker's impedance is significantly higher than that of the amplifier, you will be unable to take advantage of the speaker's full capabilities. The maximum obtainable output will be restricted by the impedance of the amplifier. However, the system should still be quite usable. You are not risking damage to your equipment, and as long as you can obtain adequate output for your needs, there should be nothing to worry about.

However, the reverse situation is fraught with danger. If the impedance of the speaker is lower than that of the amplifier, then you run the very real and serious risk of overloading your speakers and causing them serious damage. Even more likely, damage to the amplifier would occur. Attempting to provide enough current to drive a load greater than the amplifier is designed for will often damage the output transistors and other power stage components. There are free calculators on the web to use in matching speaker loads to amplifier outputs. At the time of writing, <http://www.duncanamps.com/technical/impedance.html> contains an Excel spreadsheet very useful for this purpose, or you can Google for other sources.

When dealing with audio signals (at microphone or line levels), remember that you are concerned with the transmission not of power, but of the signal voltage. You should ensure that the source impedance is lower than the load impedance, ideally by a factor of at least 5 and possibly 10 or more.

Let's look at some real-life examples. To help you understand the concept illustrated in Figure 3.4, these are examples using the specifications of some actual items of equipment. The fact that these examples are quoted here in no way is intended to imply that

#### Impedance Summary: Audio Examples

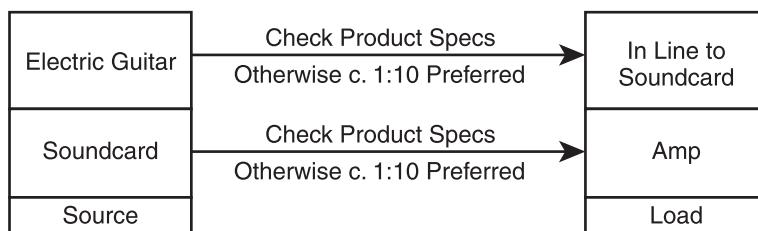


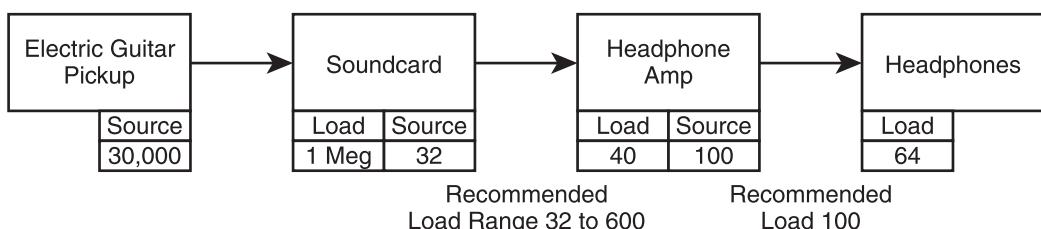
Figure 3.4 Impedance matching an unamplified audio signal.

these particular items are good, bad, or indifferent. They are quoted purely for the purposes of illustration.

- **Recording with a microphone.** The Alesis iO|26 is a soundcard whose microphone input (load) impedance is stated to be 1,200 ohms. The Bluebird condenser microphone (from Blue) has an output (source) impedance of 50 ohms. This is considerably better even than the suggested 10:1 ratio. The Studio Projects C1 condenser microphone exhibits a stated output impedance of less than 200 ohms. With a ratio of input to output better than 5:1, this should also be good to use, although we are running a little closer to the margins here. This soundcard is well suited to handling low-impedance microphones like these, but you wouldn't want to use it with a high-impedance microphone, whose impedance typically is 10,000 ohms or more.
- **Recording inline or line-level signals.** The principle, of course, is exactly the same as when you are recording with a microphone, but the actual numbers are considerably different. Typically, for example, the impedance of an electric guitar pickup will be in the range of 20,000 to 40,000 ohms. The iO|26 does provide special guitar inputs with an impedance of 1,000,000 ohms. Provided you use these special inputs, you will again be all right. However, if you tried recording it through a normal line input (whose impedance is typically around 15,000 ohms), you would experience a severe loss of level and probable distortion.

Incidentally, when recording an electric guitar (or other high-impedance source) inline, you should use a cable designed for guitar use and one as short as possible.

Let's now examine one example of an audio signal flow, together with an analysis of the impedance levels that might occur when you are recording an electric guitar and using headphones for input monitoring at the same time. The actual values used in Figure 3.5 are in each case based on the specifications for one particular model of that type of equipment. They should not be taken as typical or representative. You should *always* consult the manufacturer's specifications for details of the impedance levels that will apply to that particular device.



**Figure 3.5** Recording an electric guitar with input monitoring.

In Figure 3.5, the specifications for the guitar pickup tell us nothing about the recommended load. However, because the input impedance on the soundcard is well in excess of the 1:10 ratio, we should have no concerns here.

The relationship between the soundcard output impedance and the headphone amp input impedance (32:40) might at first appear to be a cause for some concern. However, on checking the manufacturer's specifications, we can see that 40 ohms is within the recommended range for use with this particular piece of equipment.

Finally, we are using a set of headphones whose input impedance, at 64, is considerably lower than the 100 recommended in the specifications for this particular headphone amp. Remember that at this stage, this is an amplified signal. We have already seen that in cases like this, the load impedance should match the source impedance. The manufacturer's specifications merely confirm this.

This combination, therefore, is definitely not advisable and could eventually lead to damage to the headphones, the headphone amp, or both. You would be well advised in this example to obtain either a different set of headphones with a higher impedance rating or a different headphone amp with a lower output impedance rating. Of course, this does not mean that there is necessarily anything inherently bad or wrong with either the headphone amp or the headphones used in this example. The impedance mismatch merely means that they should not be used *together*.

Please note that in this section, we have talked only about recording from analog sources. Most of the time that is what you will be doing. If you also need to record data from digital sources, such as a DAT machine, you should obtain specific digital cable for this purpose.

---

**UNDERSTANDING IMPEDANCE: KEY POINTS** Impedance is the resistance that a circuit puts up to an alternating current.

When speakers are matched to an amplifier, the load resistance should be matched equally to the source. If the load (speakers) resistance is lower than the source (amplifier) resistance, you risk damaging the amplifier, your speakers, or both.

When the signal being transmitted is an unamplified audio signal (for example, from a microphone to a soundcard), the load resistance should be higher than the source resistance. Check the manufacturer's specifications and, if in doubt, apply a rule of 1:5 or 1:10 or higher.

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## Understanding Microphones

Later we'll be looking at the various items of equipment that you'll need to purchase. If you're anything like serious about getting into home recording, you'll find that microphones will eat up a large slice of your budget. When the time comes, we'll look at how you go about deciding which microphones to buy, at least to get you started.

In this section I'll help you to understand the most important qualities and characteristics of microphones and what gives different microphones different characteristics. To begin with, there are four fundamental facts that you need to know about microphones:

- The more proficient you become at recording, the more important the microphone becomes. All other things being equal, the microphone is likely to be the weakest link in your recording chain.
- You can never know too much about microphones. Recording a track with a badly selected or poorly placed microphone will create problems that it probably will not be possible to correct.
- There is no such thing as "the best" microphone. Which microphone is preferred in which circumstances depends on a number of factors, including the instrument or voice being recorded, the environment within which it is being recorded, and the musical genre. Hence...
- You can never have too many microphones. And no matter how many microphones you have, you will always want more.

### Types of Microphones

In one sense, you could say that all microphones work in the same way. All microphones work by converting sound into an electrical signal. They do this by responding not simply to the sound directly, but to the changes in either vibration or air pressure that the sound creates.

However, what distinguishes different types of microphone is the way in which they do this. Viewed in this way, there are three main categories of microphones that are used for studio recording: dynamic, condenser, and ribbon. There are other microphone types, but these are less likely to have a place in a home studio.

#### *The Dynamic Microphone*

This type of microphone uses a wire coil attached to a cone within a magnetic field. The generator effect creates a voltage that is determined by the sound pressure. Thus, the dynamic microphone is one type of pressure microphone. See Figure 3.6.



**Figure 3.6** The Beyerdynamic TG-X47, an example of a dynamic microphone. Photograph courtesy of Beyerdynamic.

However, what is important to you isn't so much how the dynamic works as what the implications of this are.

- The dynamic microphone is relatively inexpensive to produce.
- It is sturdy, robust, long-lasting, and can withstand a fair amount of rough handling.
- Its frequency response is not as good as that of the condenser microphone. I'll explain what frequency response is shortly.

- It can withstand relatively high sound pressure levels. This makes it ideal for recording drum kits and guitar amps. A dynamic microphone can also come in useful when recording an untrained vocalist who mistakes shouting for expression.

### ***The Condenser Microphone***

This is the other main type of pressure microphone. Unlike the dynamic microphone, it uses an electrically charged solid plate with a thin metallic membrane placed in front of it. Changes in sound pressure cause changes to the spacing between the membrane and the plate. This in turn causes changes in the charge and forces a current through resistance. There are two classes of condenser microphone: large-diaphragm and small-diaphragm. Each is more suited to different applications, as you'll see both later in this section and in Chapter 4.

Again, what matters to us is not so much the process itself, but the implications and consequences of that process. A condenser microphone will exhibit the following characteristics:

- The condenser microphone gives the best overall frequency response of any microphone type. This makes it the microphone type of choice for most studio applications.
- It tends to be more expensive than a dynamic microphone, although budget-priced condenser microphones suitable for home recording are becoming increasingly available.
- It requires a power source. With some models this can be supplied by a battery, but most commonly the condenser microphone requires phantom power. I'll say more about phantom power in the next chapter.
- It is easily damaged if not handled gently. In many cases the use of a shock-mount when recording is recommended. Also, you should always use a windshield or windsock when recording vocals with this sort of microphone.
- It needs to be kept in a moderately warm and not overly humid environment.

### ***The Ribbon Microphone***

This microphone type is less likely to find a place in the newcomer's home studio, largely because of its cost. It is a vibration microphone. It works on the principle that air movement brought about by sound pressure causes a metallic ribbon to move around within a magnetic field. This particular microphone exhibits these characteristics:

- It is especially suitable for applications in which close miking is required to add warmth to a recording.

- It is less susceptible than some other microphones to picking up low-frequency noise.
- It is expensive.
- It is fragile.

## **Microphone Specifications**

Every individual microphone will have its own unique characteristics as well as those characteristics common to its genre. These are explained in the next few paragraphs, but before you read on, there is one important point to be made about understanding and interpreting microphone specifications.

You can never tell from the specifications alone whether a particular microphone will be a good one for the job for which you are thinking of using it. You might be able to tell whether it appears that it might be suitable, but specifications alone give little indication as to the quality of the microphone and its ability to handle any specific task.

### ***Frequency Response***

You can play the same CD back through six different stereo systems, all of equivalent price and quality, and guarantee that each one will color the sound of that CD in at least a slightly different way. So it is with microphones. Every microphone has its own pattern of frequency response. For reasons we don't need to go into here, there is absolutely no such thing as a completely "transparent" microphone. Usually a microphone's printed specifications will include a graph indicative of its frequency response.

Understanding frequency response can help you to identify potentially suitable microphones for different situations. For example, a microphone that adds a little presence might be a good vocal microphone.

### ***Sensitivity***

Sensitivity is a measure of a microphone's ability to convert sound pressure into electric voltage. This indicates the voltage that a microphone will produce at a given sound pressure level. A microphone with high sensitivity yields a high voltage and does not need as much gain to be added as does a microphone with low sensitivity.

### ***Impedance***

This topic was pretty well covered earlier in this chapter, in the section headed "Impedance."

### Polar Pattern

Most microphones use one of three types of patterns for recording. The pattern determines the width or narrowness of the area around the microphone from which sound is picked up. These are described in the following sections.

**Unidirectional.** This is the type of microphone most commonly used in a studio situation. Its main purpose is to record the sound that is being created in front of the microphone (rather than, for example, all around it). Members of the unidirectional microphone family include cardioid, supercardioid, and hypercardioid. If you can, imagine a V pattern emanating from the front of the mic, and that the area contained within that V is the area from which sounds will be recorded. As you move from cardioid to supercardioid and then hypercardioid, the width of that V becomes narrower.

Characteristics of this kind of microphone include the following:

- Less susceptible to room and background noise.
- Will give you better separation when you are recording several instruments together—less “bleed.”
- Close miking tends to increase the level of lower frequencies that are captured. This is called the *proximity effect*.
- Better gain-before-feedback in a sound-reinforcement system.
- Good rejection of sound behind the microphone.

**Bidirectional (Figure of Eight).** This kind of microphone records from the front and the rear but not the sides. By analogy, if you can imagine the effect of placing two unidirectional microphones back to back, that is what the figure of eight microphone does.

Characteristics of this kind of microphone include the following:

- Accepts sound from front and rear.
- Rejects sounds from both sides.
- Suitable for recording two vocalists together onto a single track.

**Omnidirectional.** As its name implies, this kind of microphone records everything that goes on around it—front, back, and sides. An omnidirectional microphone is characterized by the following:

- It picks up everything around it.
- It gives a good pickup of natural reverberation.

- It offers you very little isolation.
- It is less sensitive to breath sounds.
- It is not subject to the proximity effect.

If you are having difficulties, particularly with plosives (the sounds of p and b) and proximity effect, when recording a vocal with a unidirectional microphone, it can sometimes be worth trying out an omnidirectional one.

Note also that some microphones are switchable between different patterns (see Figure 3.7). Some multi-pattern microphones do this using the flick of a switch; others are supplied with interchangeable heads.



**Figure 3.7** The condenser microphone shown here can be switched between different patterns. Photograph courtesy of Studio Projects.

### ***Other Features***

Sometimes you'll see that a microphone has little switches on it with mysterious names such as -10 or Low Cut. Each of these can be useful in the right circumstances.

**The –10 Pad.** A –10 pad can be used to lower the sensitivity of the microphone by 10 dB. This can be required on a condenser microphone if it is used to record items with high transients, such as cymbals, loud vocalists, and the like. In severe cases, such as when a vocalist appears to mistake shouting for singing with feeling, it might be safer to always use a dynamic microphone.

**The Bass Roll-Off.** This reduces the level of the lower bass frequencies that the microphone captures. It is especially useful for eliminating or at least minimizing the unwanted effect of low-frequency sounds in your recording environment. This might include external noise, such as the rumble of traffic that seeps through the wall or the sound of an air conditioner. Removing unwanted bass from a track after it has been recorded is possible but often difficult. It can make more sense to remove it at the source.

### ***Microphone Cables***

Cables with balanced connectors should always be used when recording. We discussed balanced and unbalanced inputs when we looked at impedance. The use of balanced inputs allows the use of long cables without increasing susceptibility to unwanted noise. Balanced connections use three-conductor connectors, either the XLR (cannon) or TRS (jack) plug. XLR connectors tend to be used with microphones because they are very durable. TRS plugs are more commonly used for mixer inputs and outputs, but they serve exactly the same purpose as their XLR counterparts.

---

**UNDERSTANDING MICROPHONES: KEY POINTS** Every different microphone colors sound differently. No one single microphone will ever be capable of giving you the best results in every situation.

Dynamic microphones are especially suitable when recording an item with loud transients, such as a percussive instrument or a loud and erratic vocal. In most studio situations, however, condenser microphones are preferred.

Using a microphone with a unidirectional pattern (such as a cardioid) is preferred in most recording situations because it will focus on the subject (vocal or instrument) being recorded. However, there are some occasions when a bidirectional or omnidirectional microphone will be preferred.

A microphone's specifications may serve as a guide to help you assess its capabilities, but by themselves will not be enough to enable you to determine the mic's suitability for any given situation.

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## Understanding MIDI

During the 30-day foundation course, I introduced you to the concept of MIDI and to working with MIDI. We had a brief and fairly superficial look at using software synthesizers, using samplers, recording MIDI data, playing back our recorded data, and using a MIDI editor. In all this, you were asked to remember one key fact: When working with MIDI, you are working not with recorded sounds, but with a stream of computer data.

In this section we will look a little more closely at MIDI. We still won't be going into any great depth. Rather, we will be building on what we have already learned so as to help you assess what your future MIDI needs and requirements might be.

### There's More to MIDI Than You Think

It might surprise you to know that even if you expect to be working exclusively with recorded audio music and you have no perceived need for keyboards, synthesizers, and the like, you will almost certainly still need to have some understanding of, and make some use of, MIDI. Why is that so?

Remember again the one key lesson that you learned earlier about MIDI. It is used to send a series of coded instructions to your computer. The examples that we have encountered up to now have been primarily examples of MIDI instructions being used to synthesize music. However, there are other ways in which MIDI data can be used to work with your DAW and enhance its capabilities, as you are about to find out. It's very likely that at various stages of the recording and mixing processes, you will want to make some use of these.

Many an acoustic musician has cut himself off completely from MIDI because he does not expect to be using it to make music. To do that would be to make a potentially big mistake.

The term MIDI stands for *Musical Instrument Digital Interface*. This technology was developed in the early 1980s, originally with the goal of enabling one synthesizer to play under the control of data sent from another. This was achieved using 16 channels of data, a setup that immediately opens up all sorts of other possibilities.

MIDI data is transmitted as a sequence of events, with each individual event in itself being quite simple. Examples of different types of events are note events (to play individual notes), volume events (how loud to play notes), program change events (to change the “instrument” being played), and so on. Your DAW’s MIDI editor should include an Event view in which you can see all these events listed.

## Keyboards and Synthesizers

When you record a MIDI item using a keyboard, a series of events is recorded. You might recall that you did this on Day 12 of the 30-day course. Using the MIDI editor, you can change anything you don't like. For example, you can correct a long note, make it longer, or make it louder.

Remember that your MIDI system is capable of handling 16 channels. That means that you can have up to 16 different things going on at the same time. It's therefore well within the system's capabilities, for example, to have five or six instruments being played at the same time.

Here are a couple of terms for you to learn: MIDI In and MIDI Out. They're not difficult. When you are recording MIDI data from a keyboard synthesizer, the MIDI Out data from the keyboard becomes the MIDI In data for the computer. If you then edit this data and play it back through that same synthesizer, it becomes the MIDI Out data for the computer and MIDI In for the synthesizer. There's also something called MIDI Thru. That's a term used when MIDI data is passed through but ignored by a MIDI device.

Until relatively recently, special MIDI cables were required to convey MIDI data from one physical device to another. Both devices needed to include a MIDI port to which that cable would need to be connected. If you are using equipment that has such ports, there's no reason why you shouldn't carry on using it.

Over time, the range of options other than using hardware synthesizers with MIDI ports has gradually grown significantly. One such option is the use of software synthesizers with virtual ports. This software is installed on your computer like any other, then accessed within your DAW program (REAPER or whatever) by selecting its virtual port. We did this on Day 12 of the 30-day course, when we used the Virtual Keyboard.

One more word about keyboards before we move on. Up until now, we have spoken about the keyboard as if it were exclusively a MIDI-only device, but in fact this need not be so. Most keyboards that you are likely to find in a studio also are capable of internally converting the MIDI data they generate into an audio signal and then transmitting that via an audio TRS-to-TRS (jack-to-jack) cable or pair of cables directly to the sound-card. In such cases you would expect the keyboard to include not only a MIDI Out port, but also two audio (line) out ports (left and right). Usually one of these ports can optionally be used for stereo output if that is preferred.

The difference between using the keyboard as a MIDI or audio device is quite simple and consistent with what we have already learned.

If you use a keyboard to transmit to your computer using its MIDI ports, then you will record a stream of MIDI data that can be edited in all of the ways we have discussed. For example, it can be recorded as a piano and later edited (using a program change event) to become a trumpet.

If you record using the line out or audio out port from the keyboard, then what you will be recording will be either a single WAV file (such as we discussed earlier in this chapter) or a pair of WAV files. These files will exhibit exactly the same characteristics as any other WAV file, just as if they had been recorded with an instrument and a microphone.

In more recent years, other devices that use MIDI messages have been developed that take the application of MIDI well beyond the synthesizer. These devices usually use USB (*Universal Serial Bus*) ports to transmit and, if appropriate, receive data. This development has stimulated the use of MIDI for a newer and wider range of purposes. These uses generally involve the transmission of control data in real time for purposes other than creating synthesized music.

We'll be looking at some of these in more detail in the next couple of chapters, when we take a look at what equipment you're likely to want or need, but here is a general introduction.

## Control Surfaces

One term you will encounter over and over again is *control surface*. A control surface is a device that is used to control how certain aspects of your DAW behave by transmitting MIDI control data to the computer. For now, there is one important thing that you should know about control surfaces. They come in all sorts of shapes and sizes and for all sorts of different purposes. Just as there is no single homogenous device called a musical instrument, there is no single homogenous device called a control surface (see Figure 3.8). Rather, the term refers to a family of devices, each of which shares certain common characteristics with other family members but also exhibits its own important and specific features. Here are just a few examples:

- **To assist in audio recording.** One kind of control surface can be used to help you create tracks, name tracks, specify inputs, arm tracks for recording, even start and stop the recording process—all without you having to be seated at your computer at the time. In this case, the control data it transmits is used to execute certain commands and actions for which you would otherwise have to use the DAW program's menu.
- **To adjust track controls.** One of the most frustrating things about using a computer music program is being restricted by the obvious limitations of the mouse. For example, with the mouse you can only adjust a single pan fader at a time. With a



**Figure 3.8** The Novation Nocturn, one example of a control device or control surface. Photograph courtesy of Novation Music.

mixing control surface, you can adjust one with each hand (and more if more hands are available). This makes it easier if you are trying to balance one track against another.

- **For two-way control.** Remember what I said earlier about MIDI In and MIDI Out? Well, some control surfaces, such as a synthesizer, can function as both a MIDI Out and a MIDI In device. In other words, they can receive and act on MIDI messages as well as send them. One such example of this is the use of a control surface to function as a mixing desk. This allows you to use the faders on this mixer to adjust the volume of your different tracks while your music is playing and to have these movements remembered as a series of control instructions. When next the project is played, the changes in volume that you recorded are implemented, and you can see the faders of the mixer moving up and down accordingly as the fader movements are played back along with the music. Because the whole process involves a two-way flow of information, this is sometimes known as *duplex mode* or *full duplex mode*.

There's one aspect of this that often confuses the new user:

How can MIDI instructions be used to control audio playback? After all, aren't audio and MIDI supposed to be entirely different things?

The answer is really quite simple when you think about it. In the examples that I have given, the MIDI instructions are not used to make any changes to the actual recorded audio items themselves, only to the way in which they are played back. The WAV file itself is not altered in any way, shape, fashion, or form. The instructions recorded from the control surface are instructions to the DAW software to adjust (in this example) only the virtual faders on the track, not the recorded material itself. Adjusting such faders is a perfect example of how MIDI instructions generated by a control surface can be used to improve a DAW's workflow.

There's one more thing to say about control surfaces, and it's one that you are going to like.

In the main, devices like control surfaces are quite easy to set up and use (see Figure 3.9). You don't have to be an expert in understanding what all of the various MIDI control messages mean in order to be able to use them. In almost all cases, they come equipped with a learn mode. This means you can teach the control surface what you want it to do and how you want it to behave with a few twiddles and mouse clicks. In some cases, you can get into the control surface and code things up for yourself if you wish, but it is most unlikely, especially at first, that you will find this necessary.

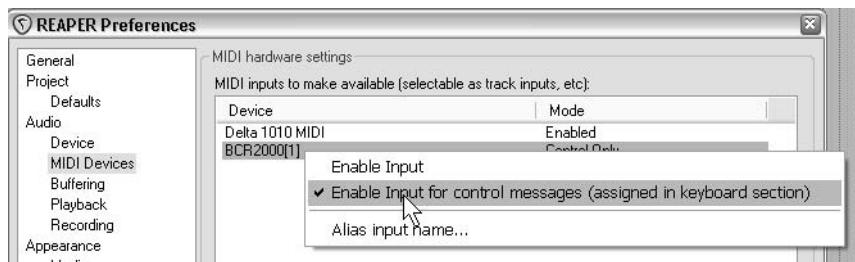


Figure 3.9 Setting up a control surface in REAPER.

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**UNDERSTANDING MIDI: KEY POINTS** MIDI consists of data that is transmitted between various MIDI devices (such as keyboards, synthesizers, and control surfaces) and the computer.

A MIDI device can be a physical piece of hardware or a virtual device generated by software.

MIDI devices and the MIDI language can be used not only to create synthesized music, but also to control your DAW's behavior.

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# 4

# Setting Up Your Studio

People take up home recording for all sorts of different reasons. For some, it is simply an interesting hobby. For others, the primary aim might be to help develop their musical and songwriting skills. Perhaps you are a member of a band aiming to be able to record and produce your own demo. You might even be aiming to ultimately achieve a standard where you are able to offer a professional service to others.

With so many varied expectations and goals to meet, the answer to the question, “How much am I going to have to spend?” is about as open-ended as any question can be. It depends on your needs, your expectations, and not least of all, how much money you have available. Suffice to say for now that within reason, there are realistic home studio setups available to suit most pocketbooks.

## The Need to Budget

The advice in this chapter certainly won’t take you all the way to setting up a recording studio at a serious professional level. No matter, because before you can get that far, you’ll need to go on a fair journey. The advice given here should help you take more than just the first few steps of that journey. Before you read on, though, you should take one point on board.

Not everybody agrees about everything. What works best for one person doesn’t necessarily work best for someone else. All of the advice offered here is based on experience and offered in good faith. Some of it might not apply to you. There are areas where opinions differ. Always be prepared to challenge any advice you are given to test whether it is the best for your own situation.

This chapter will include guideline prices for many items. These prices are based not on the manufacturers’ recommended or list prices, but on those actually available from audio suppliers such as Sweetwater or zZounds. They are (or should be!) accurate at the time this is being written, but you should be sure to check for current prices before making any purchasing decisions.

In this chapter you will find you a lot of information, suggestions, and advice. Some of it will be in the context of a best-of-possible-worlds scenario, where you are not overly restricted by cost considerations. The reality is that you will probably need to make compromises. I'll help you to identify those areas where you can compromise the most and suffer the fewest negative consequences. In the end, however, where you make those compromises will be up to you. You will need to interpret and adjust what you read here to suit your own personal circumstances, needs, and ambitions.

Here's my golden rule for spending when it comes to outfitting your home studio. It's one you're probably going to like:

*It is more important to spend wisely than to spend big.*

You could consider two budgets for a home studio—one of, say, \$5,000, and the other of \$20,000. It would be the easiest thing in the world to spend both budgets in such a way as to get a better outcome for the \$5,000 spent well than for the \$20,000 spent rashly. That doesn't necessarily mean using the larger budget to purchase equipment that is overpriced. More likely, it would mean not prioritizing well and not matching equipment effectively. This brings us to our next golden rule:

*First work out your total budget, then develop a plan that will deliver you the best possible outcome within that budget. Quality equivalence is one important technique that you will need to embrace in order to realize this.*

## Equipment Matching

So what is quality equivalence? Let's develop the concept step by step.

It's a general rule that applies in many areas of life that a system will only be as good as its weakest link. Let's take an example as simple as home stereo equipment. We'll go back to good old pre-digital days for this example because it's a good one.

Much equipment was sold as a complete integrated system, with turntable, amplifier, and speakers all in one box. In the best-value systems, all of the various components were pretty much of equivalent quality.

Now suppose that instead of purchasing a prepackaged system, you decided to put together your own system by purchasing the three items separately. Let's also suppose that you were considering two options. And let's also assume that the more expensive the item, the better its quality.

- Option 1 might include a \$100 turntable, a \$200 amplifier, and a \$200 set of speakers, all of equivalent quality.
- Option 2 might include a \$600 turntable, a \$900 amplifier, and a \$50 set of speakers.

You can almost guarantee that despite costing around three times as much, Option 2 would yield the less satisfactory outcome. This is because the wonderful work done by the turntable and the amplifier will be completely undone by the inferior speakers.

Why have I bothered explaining all this when it is really so obvious? Because so many people don't seem to apply this reasoning when it comes to their studio. They become obsessed with having to have the most expensive microphone, the very best speakers in the store, or the finest preamp that money can buy. They spend big on one or two items and then find they have to skimp on everything else. The result is usually an overall system that is inferior to what they would have if they had thought a little about quality equivalence and planned their expenditure wisely.

Also keep in mind that as you go up the financial scale, diminishing returns set in. A \$5,000 microphone isn't going to give you a result 10 times as good as a \$500 one would, for example.

This brings us to our next golden rule:

*Being able to make the best of what you have is more important than having the best.*

That applies, of course, within reason. You will never be able to get a good recording of an opera singer or a classical guitarist with a \$20 microphone. But a decent \$200 or even \$150 microphone used well will give you better results than a \$5,000 microphone used badly.

That said, there are a few places where if you really are on a tight budget, compromising on quality is wiser than in other areas. If you really are forced into a corner, remember that you should give input devices priority over output devices. Why so?

If you can't get a decent-quality recording in the first place, forget the rest. Perfect monitors will just enable you to hear badly recorded material more perfectly. But if you have a good, clean recording, there's a halfway decent chance that you'll be able to get a halfway decent sound even if the equipment that you're using is...well, only halfway decent. And that good, clean recording will still be there later, when you're able to upgrade your other gear.

## Planning Your Studio

This is one area where you might well be able to keep your costs to a minimum, perhaps initially zero. Unless the environment that you're using is just plain awful, don't spend a penny in this area until you've got your system up and running and you've experimented a little.

Keep this in mind as you go about setting up your home studio:

*Concentrate on getting your layout right before you start spending too much money.*

Let's talk first about selecting a location.

Where you are going to locate your studio and how it is to be laid out should be your first considerations. This is because it is probably the aspect over which you will have the least control. Subject to having a reasonable budget, you will have much control over the decisions that you make regarding which equipment to purchase. However, you will have little or no control (at least at first) over the environment within which you will be recording. You will therefore need to ensure that your choice of equipment is suitable for that environment.

Depending on your domestic circumstances, it's possible that the choice of your recording environment may already have been made for you. You might have no choice, for example, other than to set up in the attic, the basement, or the spare bedroom. On the other hand, if you're living on your own or with a partner or housemate who shares your passion, you might have more room for maneuvering. Why not consider sleeping on the couch if it frees up your current bedroom to be a dedicated studio and if that room offers the best recording environment? Let's take a look at what room characteristics you are likely to look for in choosing where to locate your recording equipment.

It's easier to describe the characteristics of a bad home recording environment than it is a good one. Moreover, in the typical home, rooms exhibiting many of these characteristics are quite often the norm. If you can, look for a room as unlike the one described here as possible.

- **Noisy location.** I once had a home studio on the main Sydney airport flight path. To say that the location was not ideal would be an understatement. Eventually I moved houses. But until I did, I found ways of working around the problem. There was simply no alternative.
- **Square, regular shape.** Square is worse than rectangular, and rectangular is less than ideal. Alcoves and odd-shaped walls are a blessing. That's because the way sound waves are reflected within a room, the more regular the shape of the room, the greater the extent to which unwanted reflections will take on a life of their own.
- **Flat ceiling.** A flat ceiling will compound the problem of excessive reflections. If you have a flat ceiling in a room where room height equals room length equals room

width, then think about moving out. Sloping or cathedral ceilings, on the other hand, can be great.

- **Low ceiling.** Higher ceilings (within reason) are generally better than lower ceilings.
- **Uniform, symmetric design.** Don't be afraid of windows—windows can be your friend as long as they don't let in too much noise, and especially if they come in different shapes, sizes, and positions within the room.
- **Uniform surfaces.** Bare walls, floors, and ceilings can be guaranteed to help those reflections bounce around with a terrifyingly determined vigor. On the other hand, covering every surface will be likely to yield a dull, dead, muffled effect. Try things such as carpeting the floor but not the ceiling. Some do-it-yourself acoustic treatments, such as hanging an otherwise unwanted blanket here and there (on walls that are not opposite each other), might also help if you're still getting too many reflections.

How do you know what's right? Try positioning yourself in various places within the room and clapping your hands loudly. If you hear the sound bouncing back at you, it's not right. If it's too muffled, it's not right. If it's neither of these, you're doing well.

Another issue to consider is how you deal with excess bass. This is a common problem in many home studios. Don't make the mistake of thinking the more bass the better. You want to record the natural bass of your instruments, not whatever distortions your room does to it. We could talk about bass traps, but you might not need to go there. The odd piece of furniture or two positioned diagonally in corners might help give you what you need.

In considering a few makeshift solutions (such as using blankets and furniture), we have skated around the topic of serious acoustic treatment. There's a reason for this. There's no doubt that acoustic treatment of a studio can often yield significant benefits, especially when you are mixing. It is questionable, though, whether this should be a high priority for the inexperienced newcomer. You could end up spending a lot of money for little (if any) perceived improvement. In most cases the time to consider this is probably later, when you feel that you are truly doing the best you can with the equipment that you have and that the room acoustics are holding you back. That said, there are exceptions, especially if you have a truly awful room (acoustically). Even then, using a better room might be an easier answer, at least in the short term.

In all this discussion, we've assumed so far that you don't have the luxury of a separate control room. This means that the room where your desk and computer are set up is the same room that you will be using for your microphones and musicians. Don't let this

worry you right now. That's the way at least 95 percent of home studios start out. Many manage to continue perfectly well with this arrangement, although studio layout does then become a slightly more challenging and important (but not overwhelming) issue.

There might or might not come a day when your home studio can benefit from spending a lot of money on further acoustic treatment, but that day will not come before your knowledge and skill levels are at a sufficient level to deserve it. If and when that day arrives, you'll know it. Quite possibly, you'll be living somewhere else by then.

Toward the end of this chapter, we'll return to our studio design and look at how to go about creating a layout that gets the best out of your recording environment, together with a few budget options for some acoustic treatment if you still need it. Before you can do that, though, you're going to need some equipment.

## DAW Software

Selecting the best software to meet your needs can be one of the most difficult decisions for the new user to have to make. This is because usually you really need to get quite deep down into a program before you can figure out not only whether it is capable of giving you what you want, but also whether you like its interface, its design paradigm, and its way of doing things. Let's nail one myth from the outset.

*There is no such thing as a single DAW software program that is indisputably the best choice for everybody in every situation, regardless of their needs, interests, and circumstances.*

In one sense, most of the many widely used programs that are available are pretty similar. Whether you're looking at Ableton Live, Cubase, REAPER, Samplitude, SONAR, or many others, they all provide the same or very similar core features. Each has its strengths and its weaknesses, but most users could probably get by with any of them.

However, in no less an important way, they are all different. Each has its individual design philosophy, its way of working. As a general rule, the more expensive products include more third-party bundled products, such as FX plug-ins and soft synths. Whether the extra cost is worth it depends in no small measure on how useful these items are to you.

One issue with DAW software that comes up time and again on forums is that of stability. Nothing is more frustrating than trying to run a studio with DAW software that keeps crashing on you. There are some important facts that you need to understand about this issue:

- Most DAW software is stable and reliable for most users most of the time.

- No DAW (or any other) software is 100-percent stable all of the time in every single environment and under every circumstance for every single user.
- Any software will crash under certain circumstances. Those circumstances are likely to be different for different programs. Therefore, if you read a post to the effect that, "I tried using *xxx*, and it just crashed all the time," it doesn't *necessarily* mean there is anything inherently wrong with that program. There are all sorts of issues involved. Here are just a few: Was it installed correctly? What else was running on the computer at the time? Were the correct audio drivers selected?

It really is important, therefore, that you try out as fully as possible the software that you are considering on your machine and in your circumstances.

It's not surprising that not everybody gets their choice of software right the first time. What is unfortunate is that getting it wrong can be costly, especially for those programs that cost \$1,000 or more.

During our 30-day foundation course, we have used REAPER. If you've been happy using it up to now, you may wish to consider purchasing your license and moving forward from there. A non-commercial license will currently cost you \$50; a commercial license will cost you \$225. Both licenses entitle you to use exactly the same software. The difference essentially comes down to whether you are doing it for fun or for money. If you wish to consider alternatives, then you should do that. Here are some issues to keep in mind if you do.

Well-meaning friends will bombard you with advice, usually to the effect of, "You must use this program—it's easily the best." Beware. Of course, informed advice can be helpful, but always check two things.

- What do they use it for? If they have no need for synthesizers and samplers, and you're going to be up to your ears in them, then that person's advice isn't very helpful for you.
- What other programs have they used? It is amazing how many people will enthusiastically advocate their own choice of product as "the best," even though they have never actually seriously tried anything else.

If your musical collaborators are using different software, think about whether it will be easy to transfer projects. Ultimately, however, you must choose the DAW that is right for you.

Here is a checklist of some other points to look out for:

- Not all companies offer “try before you buy” versions of their software. In the case of those that do, the trial versions are usually crippled in some way, somewhat restricted in functionality, and/or time expiring. For example, some trial software will not allow you to save your work. That alone can make attempting a serious evaluation frustrating and difficult.
- Some companies make available what they call *entry-level* versions of their product. These are budget-priced, stripped-down versions of their flagship products with certain features removed and limitations imposed (usually on the number of tracks you can include in any project file or on your use of plug-ins). If you like them, you can later upgrade to the fully featured version, but the path can be an expensive one.
- You may need to share files with programs other than your main DAW host. For example, if you intend to get heavily into tempo-based composition (using loops and samples), it is likely that at some stage you might want to be able to import files in REX format (as produced by a program called ReCycle). If you know, or consider it likely, that you need to use files in any specific format, check to make sure your proposed software will support this.
- At some later stage, you might need to run your main studio DAW simultaneously in tandem with a different stand-alone program (as opposed to plug-ins). This is not something everybody needs to do. However, for this to be possible, both programs will need to support a feature such as ReWire. In the case of REAPER, the program also supports ReaRoute, which brings added flexibility to this particular application.

Software prices vary enormously, and it would be a mistake to think that a more expensive product is necessarily going to give you a better outcome. How much you will spend depends on which program you choose. Some examples at current prices (as of September 2008) are listed in Table 4.1.

Another software cost factor is the cost of upgrades. These tend to be released each year, usually in the fall. Different company policies vary. At the time of writing this, for example, the purchase of a REAPER license includes a free upgrade to the next version. On the other hand, an upgrade from SONAR 7 Producer to SONAR 8 Producer, for example, costs \$179.

What's more, it doesn't end there. As well as your main DAW software, you might also need some additional third-party software, such as FX plug-ins, synthesizers, and samplers. As a general rule, you can expect that any third-party product available in VST,

**Table 4.1** Cost Comparison of Different DAW Software

Product	Lite Version	Full Version
Ableton Live 7	LE: \$199	Full: \$599
Cakewalk Home Studio 7	Basic: \$139	XL: \$209
Cakewalk SONAR 8	Studio: \$369	Pro: \$619
Cubase 4 Lite	Essential: \$195	Studio: \$499
Cubase 4		Full: \$999
REAPER 2.5		Non-comm'l: \$50 Comm'l: \$225

VSTi, DX, or DXi format that you might need will work with any of the main host PC DAW programs, such as those listed in Table 4.1, but it's advisable to make sure first. Let's look at each category in turn.

## FX Plug-Ins

FX plug-ins in Steinberg's Virtual Studio (VST) or Cakewalk's DirectX (DX) format should run on most PC-based DAW software. Some VST software may also be Mac-compatible, but you should never assume this. For some Mac-based software, such as Pro Tools, RTAS format is required. Mac's own Audio Units (AU) format is commonly used by other Mac DAW host applications.

You could spend just about every dollar you earn on plug-ins if you had a mind to, only to find out that most of your money had been wasted. There are third-party plug-ins available for all sorts of purposes, including many you didn't even know existed. These start at the applications that we have already encountered (EQ, compression, delay, and so on) and take off from there.

For example, there are plug-ins specifically sold for use with vocals, others for use with guitars, and so on. Very often these are no more than some permutation of perhaps EQ, compression, and delay bundled together and marketed in such a way as to get you to part with your cash.

The best advice I can give you right now is that unless you are absolutely 150-percent certain that you really do need an item, do not purchase any of these at first. Otherwise,

you'll probably end up not needing most of them and not knowing how to use those that you do need.

Start by getting to know the plug-ins that are supplied with your DAW. Consult the product's documentation, join a web user forum or two, and get a useful book. Learn what they do and how to use them. Seek the advice of others. Practice and experiment. This will yield two main benefits. First, you'll be surprised at how much you can get with what you've already got. And second, you'll be in a much better position to assess what else you need.

Sure, this takes time. I never said it wouldn't. But taking time is better than wasting money.

## **Software Synths**

It would be possible to fill an entire book with what could be said about software synthesizers. The best advice I can give you for now is to tread cautiously at first. Don't be overly ambitious or jump straight into the deep end. Some people need a lot of software synthesizers, others only a few, and others none at all. Take your time. At this stage, keep your purchasing down to items that you are certain you absolutely cannot live without and that you are confident you can master. Remember that you can usually download free-trial evaluation versions of many products before you commit to buying.

If you find the topic overwhelming, you can start by dividing what's available into a number of categories that define the main different types of software synths out there. This should help you evaluate your needs. The first thing you need to know about software synths is that they can be very expensive. The second is that they come in a number of different formats.

PC-based software uses synthesizers in VSTi and DXi formats. Pro Tools on the Mac requires RTAS, and most other Mac applications use AU. Not all synthesizers are available in all formats, so it is important to check this before you buy.

That said, let's now take a look at what's available. Understanding this becomes easier when you realize that many software synths are at heart virtual devices that aim to re-create within your computer the behavior of some piece of actual hardware that exists (or at some time existed) out there in the real world. It might pay to seek out some freeware synths to cut your teeth on first. This should help you to identify which kind(s) of synths you may later need to purchase.

### **Simple General MIDI Synths**

A simple General MIDI synthesizer is the easiest type of analog software synthesizer to use. It consists entirely of presets that are used to replicate the sounds of different

instruments, ranging from a piano to a piccolo, from a cello to a choir, from a steel drum to a sitar, from a glockenspiel to a guitar, and everything in between. Typically, with such a synthesizer, the user has little ability to modify or customize the synthesized instruments other than with perhaps a little reverb or chorus. Generally, these synthesizers are capable of producing up to 16 different instruments simultaneously.

One example of such a synthesizer is TTS-1 from Cakewalk.

Most software synthesizers put more creative power in the hands of the user than does this type. Let's lift up the hood and see what's really there.

## Other Analog Soft Synths

These are software items that aim to re-create the sound of hardware analog synthesizers. Of course, it can only be a simulation, not least because analog synths work using voltage, not data.

An analog synth uses oscillators to generate a fat, full waveform in a pattern such as saw, sine, rectangle, or triangle. It then uses a number of techniques to apply filters to that waveform to remove parts of the sound. For this reason, it is said to work *subtractively*.

This kind of synthesizer is especially popular for creating bass sounds, but it has many other applications as well, including pads, organs, and strings. It is at its best when it's used to create "mushy" sounds rather than crisp, precise ones. Two of the best-known examples of analog synths are the Moog Modular 5 and the Minimoog 5 from Arturia (see Figure 4.1). These both retail at about \$250 each.

## Frequency Modulation Synths

Amongst the best-known examples of frequency modulation synthesizers are Yamaha's DX and TX series, which first appeared in the 1980s. They work not by creating a fat waveform, like an analog synth, but by producing a series of thin sine waves that are put together. The characteristics of the sounds produced are initially determined by the frequencies selected for the original sine waves. They can then be molded by the application of any number of effects.

This method makes available the synthesis of an almost unbelievable range of sounds, especially where precision and clarity are key requirements.

An example of a popular synthesizer in this category is the Native Instruments FM8, which retails at around \$200. (See Figure 4.2.)



Figure 4.1 Minimoog.

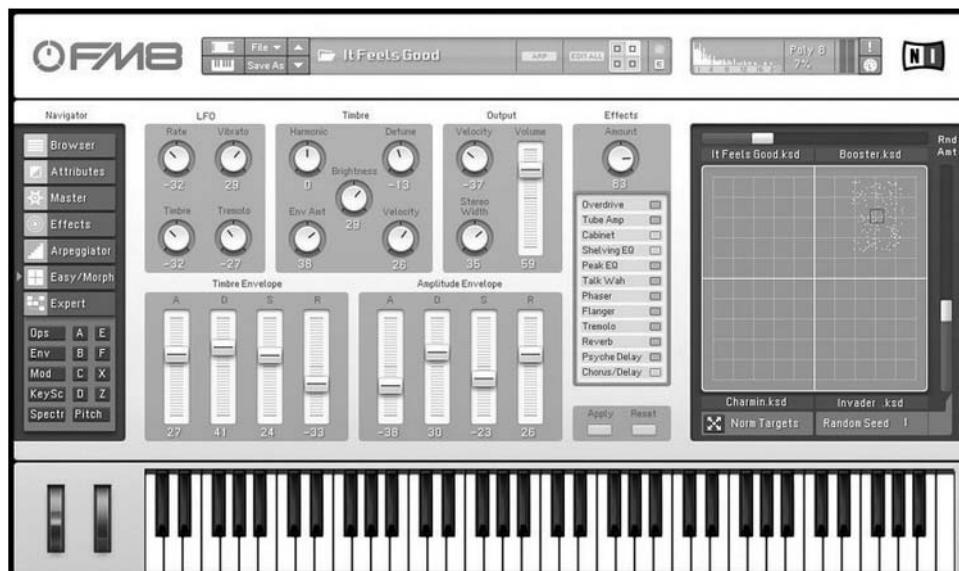


Figure 4.2 Native Instruments FM8.

## Sample Playback Synths

We came across a very basic example of using audio samples to create synthesized music during the 30-day foundation course. It probably won't surprise you to learn that sampling can get a lot more sophisticated than that. Digitally recorded samples of actual instruments are manipulated using a combination of analog envelopes and digital synthesis techniques, such as low-frequency oscillators and digital filters.

The number and variety of products available in this category are so vast that it really deserves to be considered as a major topic in its own right. There are products that deliver a full digital orchestra (and more), such as Cakewalk's Dimension Pro, which retails at around \$300 (see Figure 4.3). This is supplied with a staggering 7-gigabyte library of audio samples. Dimension Pro is also included free of charge with SONAR 8 Professional Edition.



Figure 4.3 Cakewalk's Dimension Pro.

Of no less significance are the products that focus on just one particular instrument, or family of instruments, or sounds appropriate to a particular musical genre, such as drum kit. These really do cater to all tastes. The variety of content and application is quite

extraordinary, ranging from Native Instruments Akoustik Piano (about \$200), which models the sound of four of the world's best grand pianos, all the way to products like E-MU's Proteus X2 (around \$100), whose 3-gigabyte library is focused on the sounds of urban and hip-hop music.

## **Wavetable Synths**

By now you're probably wondering how there can possibly be more, but there is. Much more!

Wavetable synths take sample playback synths into a territory of their own. They use not just one single sampled instrument to create their sounds, but a whole set of instruments. The software is capable of morphing from one instrument to another.

That, of course, is just the beginning. Wavetable synths incorporate the features we have met in other synth types, such as envelope generators, oscillators, and filters. One example is the Korg Legacy Digital Edition, which retails for around \$200.

## **Software-Only Synths**

This category of synthesizer merely takes what we had up to now a logical step further. Put simply, there is another disparate group of synthesizers that are not based on any existing or previously existing hardware. They are designed as software products from the bottom up. Each has its own distinct characteristics. This sort of product is taking us somewhat beyond our terms of reference, but just so that you know what's there, I'll mention two—Absynth and Reaktor, both from Native Instruments.

Absynth comes in at around \$80. The manner in which it combines sampling with synthesis lets you experiment with music in ways not possible with conventional synthesizers. For example, it offers multiple synthesizers and multi-sampling techniques, together with routing options that open up incredible possibilities.

Reaktor retails around \$450, and I won't even try to begin to describe its capabilities—it would be a lot easier to say what it can't do! In short, it gives you a complete system for music creation, taking synthesizers way beyond the constraints of their earlier days. A product like this is, in effect, a whole software suite, inasmuch as it allows you to use any number of different modules as needed to construct your music. If you're new to all this, don't even think about using software like this just yet.

That's the software out of the way. Now we can get down to the nitty gritty of deciding what real equipment you are going to need.

## The Studio Overview

Once you've decided on your choice of software, it's time to work out your studio plan. To begin with, we'll look at just the basics. In the next chapter, we'll go on to look at some of the other equipment that you'll probably end up wanting.

The diagram in Figure 4.4 represents the setup in a fairly typical home studio. There can be many variations on this model, but this gives us a good starting point.

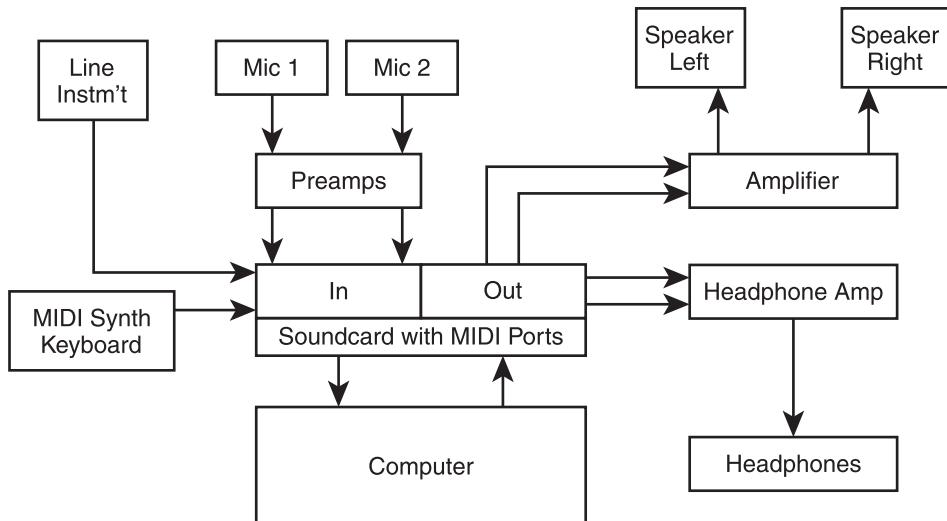


Figure 4.4 Basic home studio overview.

We'll examine each piece of equipment in some detail quite soon. Here is a general overview:

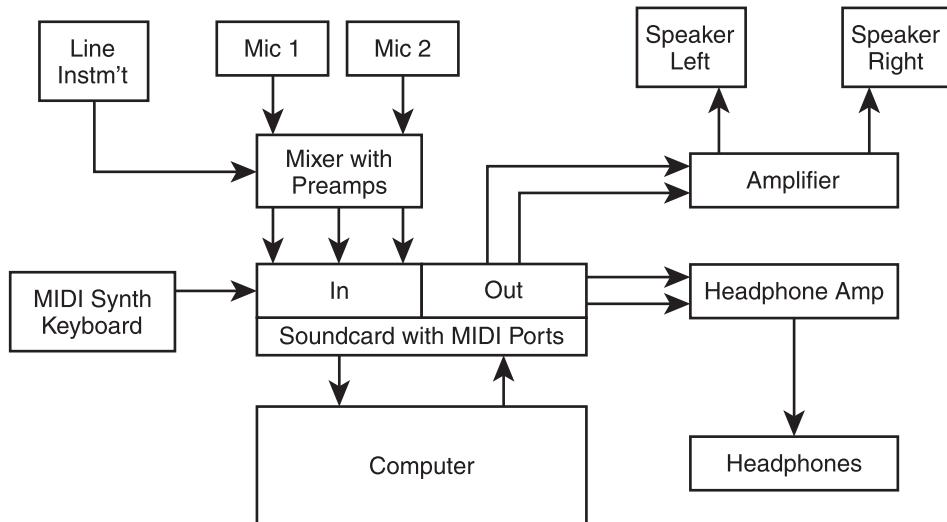
- **The computer.** This is the heart and nerve center of your studio. You would be making a serious mistake if you tried to run your studio on an underpowered computer.
- **The soundcard.** Study the diagram in Figure 4.4 carefully, and you will see that this is the hub that every other piece of equipment uses to connect to the computer. Setups in which some items bypass the soundcard and connect directly to the computer are not uncommon. We'll look at a couple in a moment. Even with these setups, the soundcard is still the single most important purchasing decision that you will have to make.

If you make a bad choice in purchasing, say, a set of headphones, at least the rest of your system will still be able to operate at its best. In contrast, a bad choice of soundcard has the potential to seriously restrict and limit just about everything else you do.

- **The keyboard.** You might need a keyboard to use purely as an electronic piano to record WAV files. You might need it as a synthesizer, a MIDI input, and perhaps an output device. You might not need one at all.
- **The preamps.** If you are intending to use condenser microphones (for example, to record vocals or acoustic instruments), you will need preamps. These may come in the form of a separate item of equipment, or they may be built into your soundcard.
- **The microphones.** We've said a fair bit about microphones already. In this section, we will be looking at some examples and options.
- **The cables.** These are required to connect anything to anything else. Different kinds of cable will be required for different purposes.
- **The headphone amp.** The soundcard that you choose may or may not include a built-in headphone amplifier. Even if it does, you may also need a separate headphone amplifier if you are intending to record perhaps two or three live musicians at the same time.
- **Amplifier and speakers.** Amplification is required to amplify the audio signal received by the speakers. Active (powered) speakers include their own amplification. If you are using passive (unpowered) speakers, you will require a separate amplifier.
- **The headphones.** These can be used to hear the playback and are also essential equipment for input monitoring and overdubbing. Depending on your choice of soundcard (audio interface), you might also need a separate headphone amp.
- **The other things.** This includes such items as surge protectors, microphone stands, microphone clips, shock-mounts, and windshields. The benefits of surge protectors are obvious, yet for the sake of laying out a few dollars, you'd be surprised how many people put their valuable equipment at risk. If you don't know what shock-mounts and windshields are, don't worry. You soon will.

Let's now look at some possible variations to this studio model. Take a look at Figure 4.5.

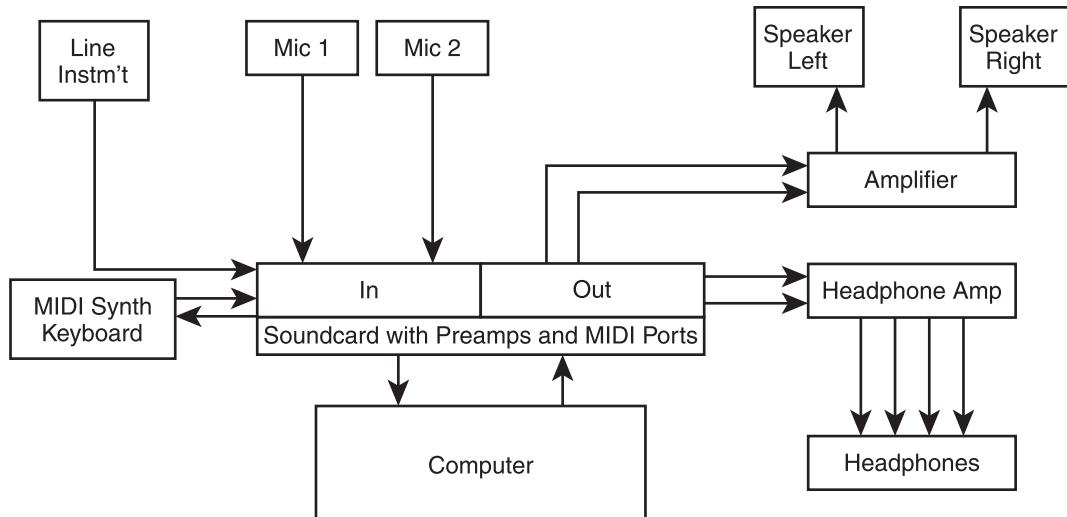
In this second example, our input flow has been made a little more sophisticated by using a mixer with preamps rather than just simple preamps. This arrangement readily gives you fingertip control for adjusting the strength of incoming signals. It also enables you to easily mute or solo different channels, thus making it easier to monitor levels and identify any possible problems with an individual microphone or line.



**Figure 4.5** Home studio with dedicated preamps.

The use of a mixer pre-input is not essential. If your budget or your physical workspace is tight, this is one piece of equipment that you can get by without. It is, however, a useful piece of equipment to have.

The third scenario (see Figure 4.6) introduces two other changes to our model.

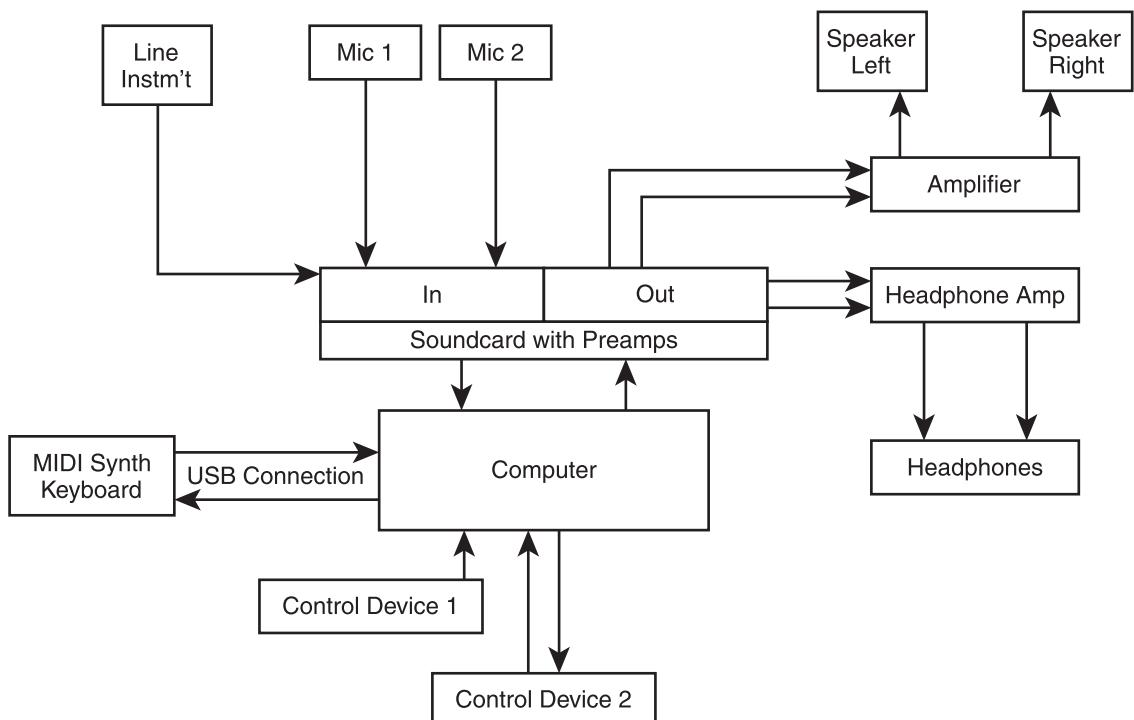


**Figure 4.6** Home studio with soundcard with preamps.

The first is that we are now using a soundcard (or audio interface) with built-in preamps. This avoids the need for separate preamps or for using a mixer to monitor and control input. Such a device usually is operated using a combination of physical controls, such as knobs, to adjust the strength of the incoming audio signal on each channel and generally includes virtual mixing software—for example, to solo and mute channels as required when you are checking levels or troubleshooting.

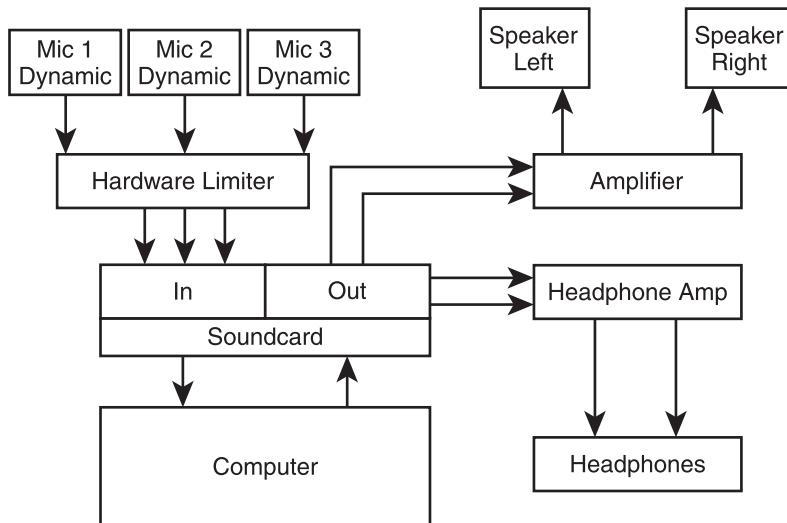
The second change is that in this case, our keyboard connects to and communicates with the computer by means of USB rather than a MIDI port. One implication of this method is that you need not rule out purchasing a particular soundcard if it meets your requirements well in all other respects but does not have MIDI inputs or outputs.

In the fourth example (see Figure 4.7), one or more control devices have been added to our home studio. The use of control devices is not essential, and you can get by without them. However, it is almost certain that the more you get into recording, the more you will appreciate the substantial benefits that such devices can yield.



**Figure 4.7** Home studio with control devices added. If you are going to be recording yourself, one type of control surface worth considering from the outset (if your budget will stretch to another \$200 or so) is one that will allow you to set up and record your tracks without needing to be physically sitting at your computer. We'll look at one of these shortly.

Recording live drum kits is a special case and an art form in its own right. Absolutely endemic to the process are problems with very loud transients, which can cause clipping. Hence, this particular recording model uses a limiter. The example in Figure 4.8 shows the use of three microphones (in practice, it would be likely to be more), in this case all dynamic.



**Figure 4.8** Home studio with external hardware limiter. By using a hardware limiter, you can prevent the incoming signal on any channel from exceeding 0 dB (or any other level that you set below that), thereby avoiding clipping. A limiter is not essential, but if you are recording live drum kits (or anything else with potentially very loud transients), then its use is strongly recommended. This is because once incoming digital audio is clipped (due to its being too loud or too "hot"), the damage is irreparable and cannot be undone. An analog limiter can trim back the audio signal to prevent clipping before it is converted into digital format.

Single-channel limiters (for use with a single microphone) start at about \$70, whereas a four-channel limiter (more appropriate for a drum kit with up to four microphones) comes in at around \$200.

Although there is no exact one-size-fits-all home studio setup, most layouts will start out looking substantially similar with perhaps a variation or two. With that in mind, we can now examine more closely just what it is that you will need.

## Equipping Your Studio

We'll now move on to a closer and more detailed examination of the main items of equipment that you will need to purchase for your studio. The core items that you

will need to pay the most attention to will probably be the computer, the soundcard (audio interface), the microphones, the speakers, and the amplifier, but there is other equipment that is also important. I'm aiming to give you as much information as I can to help you. Keep in mind, however, that no two people have identical needs. As you read through this section, you should repeatedly aim to relate the information given to your own circumstances.

## The Computer

You've already seen how important it is to take into account manufacturers' specifications when making purchasing decisions. Nowhere is this more important than when it comes to matching your computer to the needs of your preferred DAW program.

Usually you will find two sets of specifications—minimum requirements and recommended requirements. Don't even waste your time reading the minimum specs. For all intents and purposes, this information is of little practical value. This is because in most cases what will place the greatest load on your computer's CPU and RAM will be not the host program itself (REAPER, SONAR, or whatever), but the plug-ins and extras (such as samplers, synthesizers, compressors, and reverb) that you run in conjunction with the host.

It is most unlikely that you will ultimately be satisfied just running the basic host DAW alone. Also keep in mind that you are going to want your system to serve you for a good few years to come. As future software upgrades are released and as new products are developed, the demands made on your computer's CPU could well increase. The safest course of action is to regard the recommended specifications as the absolute minimum. It is better to plan for a surplus of processing power than to risk a shortfall.

Somewhere there will be a tradeoff between cost and performance. This is an area in which technology is still changing rapidly. Any detailed advice offered here would most likely be out of date before you even got to read it. Check out the audio forums and see what other people are doing.

One more word about DAW specifications: Don't read too much into them. Take them for what they are and no more. Consider Table 4.2 as an example. It compares the key recommended specifications for Samplitude, REAPER, and SONAR as of September 2008. In every case, the specifications stated are for running the software on a PC with Windows XP.

Don't be tempted to use figures like these to make any comparisons between the relative performances of the different products. The matter is far less simple than that.

**Table 4.2** Comparison of Recommended Specifications for Three DAW Programs

Item	REAPER	Samplitude	SONAR
Processor	Multiple processors	Pentium 4 2.4 GHz or higher	CPU speed 1.5 GHz or higher
RAM	1 gigabyte	1 gigabyte	1 gigabyte or more
Disk space	More than 1 gigabyte	1 gigabyte or more	6 gigabytes

Notice that in Table 4.2, REAPER is the only one of the three products where dual or multiple processors are recommended. In fact, dual or multiple processors will yield a better performance for almost any DAW software.

Here, then, is a list of issues you should keep in mind when selecting your studio computer:

- Ideally, it should be a dedicated machine, separate from the computer that you use for your email, word processing, spreadsheets, and so on. It is best kept away from the Internet.
- Dual- or multiple-core PCs will give better performance.
- Treat recommended specifications for RAM and CPU as the absolute minimum.
- Your system should preferably have at least two physical high-speed hard disks—at least 80 gigabytes for system and software files and 200 gigabytes for your recorded material.
- Ensure that the PC operation is as silent as it is possible to obtain within your budget. The amount of noise that escapes from your PC will largely be determined by a permutation of the amount of noise generated (especially by the processors and disk drives), the type of fan used, and the type of case. This is another area in which products change, but you should be able to obtain up-to-date information at [www.quietpcusa.com](http://www.quietpcusa.com).
- You should also get at least one external hard drive or other similar device for backup purposes, and preferably two. One advantage of using external disk drives for tasks such as this is that they can be switched off during recording, thus helping to keep noise to a minimum. If you really need to cut down on cost, backing up to CD or DVD alone may be an alternative, but it's slow, and you'll soon tire of it.

- Working with DAW software, especially during the mixing stages, can require—or at least be made a lot easier with—multiple windows displayed on the screen at the same time. If you don't get a system with dual widescreen monitors, it won't be long before you'll wish you had.
- You almost can't have too many USB and FireWire ports. In a short time, we'll be looking at some of the devices that you are likely to want to use in your studio, which connect via FireWire or USB. However many you identify now, it's virtually certain that as you get deeper and deeper into recording, you'll find more.
- Disable the onboard soundcard. You won't need it. By disabling the onboard soundcard altogether (so that Windows doesn't even know it is there), you will avoid any potential conflicts with the pro standard soundcard with which you are going to replace it.

These points are general and will usually apply to almost anybody wanting to set up a home recording studio. In addition, your requirements might create further specific needs of your own. Don't be afraid to seek the advice of someone who knows.

What's a computer with these specs likely to cost? Prices change so much that anything put into print here would likely be out of date before you even get to read it. If you need a ballpark figure to start with, you could pencil in something like \$1,500 to \$1,800, but don't rely on that figure. Check out some dealers for accurate, up-to-date prices.

Right, that's got the computer sorted. Let's move on to the soundcard.

## The Soundcard

Your choice of soundcard (also known as an *audio interface*) is the single most important decision that you have to make after the computer. It is the hub through which all audio signals (and perhaps MIDI data) will enter and leave your computer. Happily, it is also an area in which it is a relatively straightforward task to assess your needs. To help understand why, we can compare the issues involved in choosing an audio interface with choosing a microphone.

Your choice of microphones will depend on a number of qualities, some of which are difficult (if not impossible) to assess without first trying the product out. Even then, a microphone that is good in one situation will not be anywhere near as good in another.

In contrast, when it comes to selecting an audio interface, there are certain specific criteria that you can use to determine which models are likely to be suitable and which are not. If you get down to a shortlist of two or three reputable products, then the chances

are that any of them should do the job for you pretty much as well as any of the others on your shortlist.

*Should?* Maybe you don't like the sound of that. Words like *will* are much more comforting than *should*, aren't they? Unfortunately, even after you think you've done all your homework, there is one more hurdle to overcome. There is always a possibility, however slight, that the audio interface you have your eye on will give you problems on your system. We'll look at some of the reasons for this shortly. For now, you just have to keep this fact in mind and remember that this is a perfect example of how useful audio forums can be. Try to find out the experiences of other users whose systems (hardware and software setups) are as similar as possible to your own.

Audio interfaces can be categorized according to their physical characteristics. The three main types are PCI card, USB, and FireWire.

The PCI card, as its name implies, is a card that is physically installed into one of the computer's PCI slots. The other two types are both stand-alone devices that are connected to the computer via a USB port or a FireWire port. However, your choice of an audio device will depend more on the features and specifications of that device than on which of these categories it belongs to. For example, any audio interface that you are likely to choose will provide software that includes a virtual mixer for such tasks as input monitoring. It's likely that you'll find some of these more to your liking and easier to use than others.

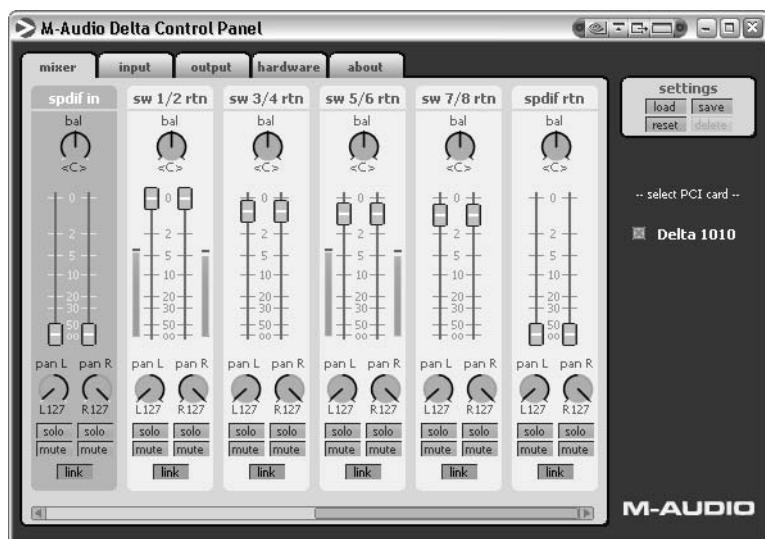


Figure 4.9 The virtual mixer software supplied with the M-Audio Delta 1010.

A common mistake made by newcomers is to underestimate the number of analog inputs and outputs that you will need. The most inexpensive and basic audio interface will give you two such inputs and two outputs, regardless of whether its type is PCI, USB, or FireWire. For most people, this will ultimately not be enough.

Consider when you are recording. Two lines out from a keyboard (if you are recording it as audio) will immediately account for both inputs, which is a bit of a shame if you want to sing at the same time. Even if you just want to record an acoustic guitar and a vocal together, to get the best results you will want to use two microphones for the guitar, one for the vocal, and probably a line out from the guitar as well.

Now consider mixing. Remember that your outputs should normally be considered as stereo pairs. A limitation of two outputs will not allow you to send separate mixes to headphones and speakers.

If your budget is so very tight that you are forced to make do with a 2-in/2-out interface, then at least make sure that it is one you can add to.

Let's now look at some examples of actual products. It must be emphasized that they are only examples. Any of the products discussed here is capable of being a good choice of soundcard for a particular individual in particular circumstances. That doesn't make any of them necessarily the best choice for you, nor does it mean that there is anything wrong with the many audio interfaces that are *not* discussed here. I am aiming to help you to understand the issues that you must resolve in selecting an audio interface, not to select one for you!

One example of a basic entry-level product is the M-Audio Audiophile 2496, a stereo 2-audio-in, 2-audio-out PCI soundcard. It also includes a MIDI In and a MIDI Out port. There is nothing wrong with its quality, and at around \$100, it represents excellent value if this really is all you need. Its downside is that if you later discover you really do need extra inputs or extra outputs, your only option is to take it out and replace it.

An example of a basic USB device is the Tascam US-122L. You can pick one up for about \$130. It, too, provides two analog inputs and two analog outputs, as well as MIDI In and MIDI Out. For the little extra that you pay, you also get phantom power for use with condenser microphones. This avoids the need for separate preamps. Incidentally, when considering any USB device, make sure it is a USB 2 device.

Now take a look at the PreSonus Inspire 1394 (see Figure 4.10). This is a FireWire device that allows you to record audio to up to four tracks at the same time (of which no more than two may be from microphones) and again provides only one pair of analog outputs. Like the US-122L, it includes preamps for use with condenser



**Figure 4.10** The PreSonus Inspire 1394 audio interface. Picture courtesy of PreSonus.

microphones. It sells at around \$150 but brings you what may well be a significant benefit. Unlike the PCI card, this product is modular. You can daisy-chain up to four of these devices together. The daisy-chained devices are controlled using software provided with the product. The system can grow along with your needs. For many users, this might represent a wiser investment.

Of course, if you know from the outset that you are going to require eight audio inputs to record your band, then a better solution than the modular one is to purchase up front a system capable of giving you what you need.

Take a look at another FireWire device, the Alesis iO|26 (see Figure 4.11). For around \$450, this provides you with MIDI input and output, eight balanced analog audio inputs (with phantom power), and eight balanced analog audio outputs. Its other features include a phono input (for use with a turntable), ADAT Lightpipe inputs (for use with a variety of ADAT-enabled equipment), and S/PDIF connectors for use with such devices as CD players, DAST machines, and MiniDisc recorders.

Another example is the Focusrite Saffire Pro 10 I/O. Only a little more expensive than the Alesis iO|26, it provides similar MIDI and analog audio inputs, and it comes bundled with some quite handy FX plug-ins.

Back at the PCI end of the spectrum, you will find the M-Audio Delta 1010, which sells for around \$600. It is a PCI soundcard that also offers eight audio in and eight out (but no phantom power), along with MIDI output. Typical of such a product, it comes with a breakout box. This is necessary because of the impracticality of trying to connect so many leads and cables directly to a soundcard.



Figure 4.11 The Alesis iO126 audio interface. Picture courtesy of Alesis.

Stepping up to about \$700 brings you within range of the Tascam FW-1082. This 10-channel FireWire audio interface provides you with six line inputs, four microphone inputs with preamps, separate outputs for headphones and monitors, as well as a real mixing desk for fingertip control.

If you want something on a larger scale, for about \$1,500 there's the Yamaha n12, whose 12 channel inputs include eight microphones with preamps.

Right now you might be thinking that on the evidence so far, FireWire and USB seem to give you more for your money than PCI does. Before reaching any conclusion like that, though, we need to dig a little deeper. Things aren't always quite as simple as they seem. For starters, PCI cards handle heavy traffic better than FireWire or USB. For most people's needs this isn't likely to be a problem, but it is something to keep in mind if you're planning to record the London Philharmonic Orchestra with 24 or more simultaneous inputs.

There are other issues that we also need to consider. Before we do, let's just clear up a couple more points.

- **Analog and digital inputs and outputs.** Up until now, we have concentrated exclusively on analog inputs because for most users they are what matters. These are inputs that take an analog audio signal (such as is generated by a microphone) and convert it into a digital format for processing by the computer. Similarly, the outputs take the digital data that was fed into the audio interface from the computer and has been converted to an analog format as required by the amplifier.

It is possible that you may also need to accept or transmit audio directly in digital format—for example, to or from DAT tape. That's where digital inputs and outputs can get to be important.

- **Bundled software.** You'll often find that an audio interface is supplied with a lite (or LE) version of a DAW software program. Although this can be useful if no other software is available to you, you are likely to find this very limited in its capabilities and restrictive in terms of what it lets you do. For example, it will impose a limit on the number of tracks you can include in a single project. It might be of use to you, however, if you just want to get an idea of the flavor of that product, of how it works, and of whether you feel comfortable using it.

### ***Audio Interfaces and Sound Quality***

All other things being equal, the quality of the sound produced by any one soundcard or audio interface compared to any other will depend on three factors:

- The preamps
- The digital audio converter
- The software drivers

Let's look at each of these in turn.

#### ***The Preamps***

Some audio interfaces include microphone preamps, and some don't. It's fair to say that in most of the well-known audio interfaces sold by leading audio dealers—Alesis, Edirol, Focusrite, PreSonus, Tascam, M-Audio, and so on—the quality of the built-in preamps will be more than adequate to meet the needs of the new user. Nevertheless, the golden rule applies: If you can, check out how satisfied existing users of these products are with their quality.

If, after purchasing your computer, you are on a restricted budget of, say, \$2,000 to \$3,000, then it definitely makes sense to choose an audio interface with such built-in preamps. Remember what I said about quality equivalence and doing the best with what you've got. A \$2,000 preamp isn't going to make a \$100 microphone sound like a \$1,000 one. What's more, getting the optimal gain and trim settings right for your soundcard's built-in preamps will give you better outcomes than will thrashing around with your \$2,000 outboard preamp without really knowing what you're doing.

If, after purchasing your computer, you've got a fair bit more to spend on your gear than, say, \$6,000 plus, and you have heaps of knowledge and buckets full of experience, and you're really setting your heart on the best, then it might be worth considering a separate preamp. To achieve any significant and noticeable improvement over what one of the soundcards mentioned earlier will give you, you'll be looking in the range of at least \$1,000 to \$2,000 for a dedicated 8-channel preamp.

### ***The Digital Audio Converters***

These are the items that handle the conversion of audio signals from analog to data and vice versa. Poor converters will tend to produce a brittle sound that lacks warmth. These are what we have to thank for the widespread myth that still prevails in some quarters that digital recording is brittle and lacks warmth. In fact, it is recording with poor-quality digital converters that creates a brittle sound lacking warmth, not an inevitable consequence of the digital recording process itself.

With the introduction of techniques such as oversampling and error correction, the quality of DACs used in modern soundcards has improved substantially, to the point that most of the interfaces you are likely to consider should be capable of giving you a result that is better than acceptable. Much the same can be said about DACs as I said about preamps. If you're really loaded and you want the very best, you can have it...but it will cost you. Personally, I'd really struggle to justify laying out \$3,000 or more for dedicated digital audio converters in a home recording situation. If you're not convinced, just make sure that the audio interface you purchase has an S/PDIF interface. This will enable you to connect it to outboard digital audio converters at some later date if you really feel the need to.

Even if you do use an S/PDIF interface, you'll find that there's a sting in the tail with these. You have no way of hearing as such the sound produced by the digital audio converters alone. Before you get to hear anything, the fruits of their labors need to be handled and processed by the audio interface's audio drivers. And do they have a story to tell.

## Audio Drivers

Read this mantra and engrave it upon your heart.

*More than any other single factor, the quality of the software that we call audio drivers will determine how happy you will be with your audio interface.*

Of course, that doesn't mean things such as balanced inputs and outputs, good preamps, decent DACs, and so on do not matter. But they will all come to nothing if the drivers are not up to the job. It is the drivers that have the job of ensuring that the CPU is up to the task of transferring audio data between itself, your DAW, and the soundcard. We can talk more about buffers and buffer size if you like, and more besides, but you're probably happy just to accept that this is not an easy task. This is probably the main reason why you should not rule out the possibility of ending up with a boring old PCI soundcard after all.

Compared to the drivers, life for devices such as preamps and DACs is just a breeze. They just sit there working like a sausage machine all day as the data trundles through them. The poor old drivers, on the other hand, have to handle all sorts of stuff being thrown at them, often unexpectedly and unpredictably and from every direction, and they have to deal with difficult customers, such as all sorts of different idiosyncratic DAW programs and different computer components (such as motherboards) that all have their own eccentric ways and all make different and sometimes conflicting demands. All right, I'm exaggerating for effect, but if it helps you to get the point, it has been worth it.

Getting drivers right is not easy. If you don't believe this, go to the website of any reputable audio card that has been around for, say, five years or more, and have a look at its driver history. It should teach you one thing: It is absolutely inevitable when a new audio interface is released that problems will occur when that interface is used with certain hardware and software combinations. Of course, these things are rigorously tested before release, but the number of permutations that can and will occur in the real world makes the appearance of some problems inevitable. It is also, of course, not possible to predict what these problems will be.

I don't want to exaggerate this. All else being equal, the probability is extremely high that without doing any research, you could purchase a soundcard that has only just been released last week, and that it will run perfectly well on your computer and with your software of choice. But that probability gets even higher if you choose a model that's been around for a few years and research it well. Then don't forget to check their website regularly for updates. You should always install and run the latest drivers for your soundcard. At the same time, keep older versions backed up somewhere in case you need to go back to them. That can happen as well.

Okay, it's time to run through a checklist of some points for when you're evaluating your choice of audio interface. These points are general and should apply to most people who are starting out on the home recording path. It's possible that not every question will be important to you. In addition, you may need to identify other concerns not listed here, but which might be important to you.

- Never assume anything. If you have an issue that appears not to be addressed in the product feature list or specifications, then you should make further inquiries.
- Are there any known issues with using the product with your particular DAW, CPU, motherboard, and operating system combination?
- Will installation be a problem? For example, does your computer have the necessary FireWire or USB port or spare PCI slot that your soundcard requires?
- Are both ASIO and WDM drivers available?
- Is there a website where driver updates can be downloaded as they become available?
- What sort of support does the manufacturer offer? For example, email support, telephone support, or web forums?
- What maximum bit depth and sample rate do you need? It is unlikely that you'll need better than 24-bit/96,000 Hz at the most.
- Does the device have enough analog inputs and outputs for your needs?
- Does it include special inputs for sources such as guitar pickups, which require high impedance loads?
- Does it provide phantom power/preamps for condenser microphones?
- Do you require MIDI inputs and outputs also?
- Can the system be expanded if you later need more inputs or outputs?
- Does it have special outputs for headphones?
- Does it have digital inputs and outputs (ADAT, S/PDIF)?
- What are its impedance ratings?
- What are its other technical specs?
- Does it make available latency-free input monitoring?

- How good/user friendly/capable is the product's software control panel? Does this include an internal patch bay (or equivalent) to facilitate routing?

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**Soundcards and Audio Interfaces: Key Points** There are three main types of audio interface that you may wish to consider—PCI card, FireWire, and USB. In all probability the performance specifications and features of the individual item being considered will be more important to you than its type.

In particular, it is important that a soundcard's drivers work well with your computer and with your DAW and other software.

In most cases, you can expect that the quality of the built-in preamps and digital audio converters will be more than adequate to meet your needs. Using outboard DACs and preamps can yield better results, but only at a cost and only if you know how to get the best out of them.

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## The Preamps

We've already discussed the issues involved in whether to go with your audio interface's internal preamps or whether to go for a dedicated outboard device. If you choose a soundcard that does not include internal preamps, then of course you will need to purchase such a device. Budget around \$500 to \$600 for this. M-Audio, PreSonus, ART, Mackie, Studio Projects, and Focusrite are some of the brands you might wish to consider.

There is another possibility that you might want to consider, one that enables you to gain extra practical benefits from having your preamps external to your soundcard. That possibility is to use a mixing desk (provided there's plenty of room in your studio). If you take this route, you'll need to ensure that your desk provides not only preamps, but also a separate dedicated output for each channel. (Remember, you should assume nothing.) This gives you real fingertip control when setting up for recording and monitoring and is a more intuitive and flexible way of working than using a mouse with a virtual desk on your screen. The more tracks you are recording at once, the more this is the case.

One example is the Behringer Eurorack MX2642A. Very much at the budget end of the market, it sells for about \$500. For \$900, you could look at the Mackie 1604-VLZ3, whose specifications, including its preamps, are rather impressive. Don't forget, though, that if you go along the mixer path, you will need to do a little homework on impedance matching.

## The Keyboard

Keyboards come in such a range of models and with such a range of applications that the word *keyboard* really should be thought of as describing a generic and very broad family of instruments rather than describing any particular instrument as such. The fact that all keyboards have keys is not the point. More significant is what the keyboard is used for.

At one end of the spectrum, you might want your keyboard to serve you as a simple MIDI input device and no more. Its sole purpose is to record tunes in MIDI format. In such a case, you would use this in conjunction with one or more of the software synthesizers (such as were discussed earlier) and your DAW's MIDI editor. You'd probably be looking at about \$100 to \$200 or so. You might find that something like the M-Audio KeyRig 25 will give you all you need (but as always, check for any compatibility issues). What you will almost certainly find useful in such a keyboard is for it to include assignable knobs and buttons, which with some patience you can set up and use to control your DAW transport and other parameters.

At the other end of the scale, you have items such as the Korg OASYS, which is just about a complete music production system in its own right. Its features include umpteen built-in synthesizers, audio inputs and outputs, MIDI control, MIDI sequencing, sampling, effects processing, built-in MIDI editing, and much more.

It would be impossible to explore all the possibilities in anything like adequate detail here. Chances are that if you need a keyboard, you've already got one. If not, it's your call to determine what you need and how much you want to spend.

## Microphones

In the previous chapter, we took a brief look at some of the technical aspects of microphones that you need to understand before you are really able to even begin making informed decisions about which microphones to purchase. Even armed with that information, you'd be forgiven for being somewhat bewildered about how you start deciding which microphones you should set about getting.

A good place to start is with another golden rule, rather a humbling one this time. It's this:

*Ask not only if the microphone is good enough for you. Ask also if you are good enough for the microphone.*

By analogy, imagine that you're putting a digital camera into the hands of someone who has never taken a photograph before. What do you think would be the safest bet for that person—a \$100 point-and-shoot basic automatic or a \$5,000 state-of-the-art advanced model?

Now let's consider the example of a microphone that is to be used primarily for recording vocals. One exceptionally good vocal microphone is the Neumann U 87. With a recommended retail price tag of around \$4,000, it's no doubt beyond the budget of most home recording enthusiasts, but let's say you've just won the Lotto.

Microphones like these need to be treated with the utmost respect. They are at their best when used in a perfect acoustic environment. They will faithfully pick up every sound, every creak, every shuffle, every rustle, every whisper, every sniff, every wheeze, and every acoustic disturbance, regardless of whether you want them. If you can get beyond that, and if factors such as your microphone placement and the singing and microphone techniques are spot on, it will reward you well. Get any of these wrong, and it is likely to be unforgiving.

Taking this argument to its logical conclusion, if your recording environment is acoustically so bad that despite your best efforts you're still getting overwhelmed by excessive reflections, or your singer's vocal techniques are causing you problems, then in such situations it is worth considering not using a condenser microphone. Keep in mind that dynamic microphones are generally less sensitive to reflective sounds than are condenser microphones. In those circumstances, good-quality dynamic microphones, such as the Shure SM57 and SM58 or the Sennheiser MD421, will almost certainly give you better results.

That said, in most home studio situations and with most singers, a good-value budget condenser microphone in the \$120 to \$250 range is probably a good place to start. Remember what I said earlier about color—every model of microphone will color sound differently. If you can only afford one microphone, you'll have to make do with one, but ideally you should have a selection available so you can select whichever seems best for the particular situation.

A selection of different microphones of generally equivalent quality is likely to be more beneficial than would be several copies of the same microphone.

Let's look at some examples, assuming that you don't have unlimited amounts of cash. We're looking at some pretty good items that represent good value for money. Of course, which of these will be relevant to you will depend as much as anything on what it is you want to record. Use the suggestions here as a starting point, not as a destination. Seek the opinions of others, especially those whose experience you respect and whose musical genre is similar to your own.

### ***The Vocal Mic***

The vocal is probably the most difficult thing to record well. If you're going to record vocals, you'll need at least one decent vocal microphone. Which one you should choose

depends on any number of things. Male or female vocals? Hip-hop, country, blues, or opera? How much do you have to spend?

That said, here are some possible candidates. If you need a vocal microphone that can withstand a lot of abuse, and you don't have a lot to spend, think about the Shure SM58. This dynamic microphone will set you back no more than \$100, and the good news is that it will double as a good microphone for a guitar amp or a kick or bass drum. The Shure SM57 and SM58 microphones over the years have acquired a reputation that is little short of legendary. You'll find them in just about every studio, large and small. They are sturdy, solid, and reliable and will last for a lifetime.

For most vocals and vocal styles, however, a large-diaphragm condenser microphone should definitely be preferred. For around \$200 to \$250, you could consider the Rode NT1-A or the Studio Projects C1—or both. Both come with shock-mounts and windshields. Both also work well with acoustic stringed instruments, such as the guitar or mandolin. If you're really strapped for cash and can't stretch to get a dedicated vocal microphone, consider using a general-purpose microphone, such as the ones we'll be looking at shortly.

### ***The Drum Kit***

If you're going to be recording live drums, then you really do need a set of microphones designed for this purpose. As much as anything, this is because the acoustic properties of the sounds produced by a drum kit are very unlike vocals, stringed instruments, and so on.

The least expensive set you can get away with is likely to be Samson's five-microphone drum kit, which costs less than \$200 (see Figure 4.12). Don't be put off by its low price;



**Figure 4.12** Samson's five-microphone drum kit. Picture courtesy of Samson.

it's a better piece of gear than that. But, of course, if you've got the money, you can do better. About \$400 will get you Shure's PGDMK6-XLR six-microphone drum kit.

### ***General-Purpose Microphones***

How many of these you need and which items you choose will depend on which instruments you need to record. You need to understand that when a microphone is said to be good for the violin, or good for the saxophone, or good for the cello, that is really only an indicator and no more. With careful microphone placement and plenty of practice, you'll find that as long as the microphone is a reasonably good one, you should be able to get very satisfactory results by doing the best with what you've got.

That said, the better the variety of microphones that you have, the easier it will be to get good results and to make interesting recordings. You don't want every track colored in exactly the same way if you can help it. Let's look at a few, again very much in the budget range. We'll start with the least expensive and work our way up.

- **Samson C01 (about \$80).** This is just about the cheapest condenser microphone you can get, and it probably wouldn't be your first-choice microphone in many situations. Nevertheless, its response tends to have something of a high-end bias, which can lead to surprisingly good results with an otherwise dull or muddy instrument. As an alternative, consider the C03, which at \$120 is switchable between cardioid, omniod, and figure of eight. You'd be unlikely to use either of these as your main studio workhorse, but it can offer an interesting option at times.
- **Shure SM57/SM58 (about \$100).** This is a dynamic microphone whose greatest strengths are its durability and its versatility. It is equally at home in front of an acoustic guitar or a guitar amp. If funds are really tight and will stretch to just one general-purpose microphone, this could be a worthy candidate.
- **Audio-Technica AT2020 (around \$100).** This is a no-nonsense, all-purpose, medium-diaphragm cardioid condenser microphone, quite suitable for a range of acoustic instruments. It's a fairly handy all-around microphone.
- **AKG Perception 120 (\$120).** This is a surprisingly well-featured microphone that includes a bass roll-off switch and a 20-dB pad. It's worth considering as an alternative to the AT2020 if you can't stretch to both.
- **Studio Projects B1 (\$120) or B3 (\$150).** One of the most impressive features of the B1 is that it seems to retain clarity even in situations where close placement is required. The B3 is basically the same microphone as the B1, but unlike its cardioid cousin, it is switchable between three patterns.

- **AKG Perception 170 (\$140).** You should probably aim to have at least one small-diaphragm microphone in your collection—especially if money is tight, this is worth considering. The small-diaphragm microphone is especially good as an overhead, for choirs, pianos, ensembles, and so on.
- **Beyerdynamic Opus 53 (\$180).** This is another small-diaphragm condenser microphone. The manufacturers recommend it especially for use with wind instruments, but it also does a good job as a general instrument microphone.

If money really is desperately tight, and you really can afford only one budget-priced condenser microphone to start out, then you should go for one with as wide a range of applications as possible. Depending on your priorities, the Studio Projects B3, with its clear response and switchable patterns, could be one possible choice. At \$180, the AKG Perception 220, with its warm, clear sound, could be another serious contender.

Finally, I'll say it again. These are just *some* examples of low-priced microphones that are worth considering to get you started. There are others out there not included in this list that are just as good—and for some applications better. I'm certainly not suggesting that you should rush out and buy anything without first doing your homework. What you'll need and how many is up to you.

In the next chapter, we'll take a look at some of the microphones you might want to look at a little later on in life, after you become more experienced.

## Amplifier and Studio Monitors

We have already seen that different speakers, like different microphones, color sound in different ways. There are some very important lessons to be learned from this.

The first is that choosing your studio monitors is an entirely different ballgame from choosing a home entertainment system. The latter is likely to be loaded toward boosting those frequencies that the ear likes the most. When mixing in the studio, however, you want to be able to hear as accurate and transparent a mix as possible. Home entertainment systems can be useful late in the process as a means of testing your mix, but they should never be used for producing it.

Unfortunately, there's one other thing that I have to say: Regardless of how much money you spend on your speakers, you will never achieve the Holy Grail of a system that gives you a 100-percent transparent response in all circumstances.

It is therefore important that you get to know your speakers well and in particular their idiosyncrasies and individual characteristics.

That takes time, experience, and more time. Be prepared to experiment and learn. Test your mixes on different pieces of home electronics equipment as far as possible, all the way up from a cheap portable boom box (if you can find one) to a top-range audiophile system. Compare them to the sound that you are getting from your studio monitors, and be prepared to make adjustments accordingly if need be.

Another decision you will have to make is whether to go for powered (active) speakers or a separate amplifier with unpowered (passive) speakers. Remember that if you purchase your amp and speakers separately, you will need to ensure correct impedance matching. Powered speakers are probably the simpler option.

That takes care of that. So how much is it going to cost? By now you know the answer to that one—as much or as little as you like. Arriving at a sensible figure depends largely on what other equipment you have and how much power you need. However, if we’re talking about a system with maybe a \$600 to \$1,000 audio interface (with built-in preamps and DACs) and a modest but proud collection of four or five \$120 to \$200 condenser microphones, then you’re probably looking at around \$600 to \$700 for a pair of powered speakers like the Mackie MR8s or the Yamaha HS80Ms.

If you really can’t stretch that far, you could try out what’s available for about \$300, such as the Alesis M1 Active MKII, to see whether there’s anything you like.

## Headphones

Your headphones will basically have to serve three purposes:

- Input monitoring for the person behind the desk recording
- Input monitoring for the person being recorded
- As an aid to mixing, to complement your use of monitors

The first two of these are very similar, but each is likely to require a different mix. One useful trick to remember especially when recording a vocalist is that by lowering or raising within the mix the level of the signal of their own live performance that they hear in their headphones, you are able to encourage them to sing (or play their instrument) louder or quieter.

During input monitoring, it is likely that you will want to send a mono signal to your headphones. During mixing it is more likely to be stereo.

The type of headphone most suitable for studio use is called *closed*, or *closed back*, or *closed ear*. These three terms are essentially synonymous. The ear pads are known as *circumaural* or *sealed*. This simply means that the entire output of the headphones is

contained, thus ensuring that possible bleed back onto the track being recorded is kept to an absolute minimum.

This brings obvious advantages as far as recording is concerned, but also a less obvious disadvantage. Because the performer's ears are cut off from the outside world, she may quite easily make all sorts of distracting and unwanted noises, such as shuffling or foot or finger tapping, that you definitely do not want to hear. Recording with headphones is a technique that requires patience, practice, experience, and experimentation.

Then there is the question of comfort. If the headphones are for your own use only, it should be relatively easy for you to assess the degree of physical comfort (or discomfort) you feel when trying out one set of headphones as opposed to another. If they are to be worn by various band members at different times, it obviously becomes more difficult. Look, then, for a pair that appears to be easily flexible and adjustable.

The other big important issue, of course, is that of sound quality. You will get better results and a more expressive, inspired performance, especially when recording vocals, if the top end can be heard clearly and the bass is thick and full. Listen for a good, even frequency response. You don't want a sound that's too tinny at one end of the scale or too muddy or boomy at the other.

How much do you need to pay for a good set of headphones?

To begin with, cheap \$20 headphones are definitely not recommended for studio use. They really are an economy that will come back to haunt you. Their sound quality is at best uninspiring, and they will bleed all over the place. This can add an irritating constant sizzle to the background of your tracks. If you have any, throw them away.

If money is absolutely tight, try out something like Yamaha's RH3C, which comes in at about \$40. They represent good value for the money, and you might be happy with them. If you can go up to \$100 or \$150, you'll notice the difference with something like the Sennheiser HD280 Pro, the Audio-Technica ATH-M50, or the AKG K 141, any of which would likely be a good choice.

## **Headphone Amp**

If your soundcard includes a headphone output, you can get by without a headphone amp if you work exclusively on your own. If it has two independent outputs, you might get away without one even if you need to record others. As well as studio-quality monitors, you can hook up domestic hi-fi equipment to get another "view" of your mixes. If you do, you might be able to squeeze one extra headphone output from the hi-fi amplifier. If you need to record more than one other at a time, you will definitely need a headphone amp.

In the simplest of cases, you can pick up a Samson or ART 4 channel amp for as little as \$50 to \$60. Something like this might be sufficient for your basic needs, especially because it includes an independent volume control for each channel. However, there are important benefits to be had from a headphone amp that go well beyond simply making more outputs available.

If your budget can stretch to \$200 or \$250, it is definitely worth considering something like the ART HeadAmp6Pro. For the extra money, you get six independently adjustable channels, each with its own volume, bass, treble, and balance controls. These can be used to fine-tune with precision not so much the mix as the quality and color of the sound that each person individually gets in his or her headphones.

Don't underestimate the role that a good headphone sound can play in coaxing the very best performance out of people, or the contribution that a headphone amp can make to that effect. You'll find the extra \$150 or so money well spent. If you can't afford a headphone amp for now, make sure that it goes pretty high on your wish list.

## **Extras and Accessories**

Of course, there is more. There's always more. Microphone stand, speaker stands (unless they're going to be wall-mounted), leads, cables—you name it, you'll need it...

## **Also Worth Considering**

So now you've bought and paid for the absolute essentials. Can you stretch to a further few hundred dollars? If you can, you won't regret it. We're going to look now at a couple of items that, strictly speaking, you don't actually need. It's just that if you *do* get them, you'll very quickly wonder how you ever managed without them.

### ***Handheld Recorder***

You want to rough out a song, a very quick demo to take to the rest of the band. Audio quality doesn't have to be the best. It's more important to strike while the inspiration's hot. In these circumstances, do you really want to go to the trouble of setting everything up and booting up the computer? This is a perfect example of when a handheld recorder can be just what you need.

A number of models are available to suit different needs and different budgets. It's unlikely that you'll need a topnotch one. You'll just want something simple that you can use to help you work out your songs or to put down at band practice. You can even use it to record a guide track, which you can later upload to your computer when the time comes to make a proper recording.

Shown here is the Zoom H2, which, for about \$200, should do the job nicely for you.



**Figure 4.13** Boss's Zoom H2 handheld recorder. Picture courtesy of Boss.

### ***Frontier TranzPort***

Up until now, I've tried to maintain a fair degree of neutrality, presenting you with examples of products but not necessarily recommending these examples as being any better than anything else. When it comes to the Frontier TranzPort, I'm just going to have to break that rule. This little device really is in a class of its own.

If you're in the situation of having to record yourself, you'll soon find out how limiting it can be to have to be both musician and sound engineer at the same time. Ordinarily, you have two choices. One is to record while sitting uncomfortably at the computer and picking up the computer noise along with whatever else is going on. The other is to keep running back and forth between the microphone stand and the computer, keeping your fingers crossed and hoping that you've got things like sound levels right.

For about \$200, enter the TranzPort. It is the perfect device for when you are tracking alone in the studio. It works with Cubase, REAPER, SONAR, and a host of other DAW software (but as always, check first if yours isn't listed). It allows you to run your DAW without needing to be at or near your computer—in fact, you can even be in a different room if you want to be.

## Budget Summary

Now is the time to get out your calculator and total what this little lot's going to add up to. Sit down with a strong coffee or a stiff whiskey—you're going to need it because we now come to the one absolutely infallible rule that applies to financing your studio.

*You always end up spending quite a lot more than you were expecting.*

The following tables might serve as a useful guide. You've gathered by now that when it comes to the home recording studio, there really is no such thing as a "typical" arrangement. All will share certain characteristics, but each will be different in a combination of ways that will make it unique. Tables 4.3 to 4.5 show three checklists—one for the solo home musician on a shoestring, one for a small band with a bit more cash to spare, and a third one that gets us closer to a dream home studio. In each case, you'll find examples of what you might end up spending. I have deliberately refrained from mentioning any specific brand names or models in these lists so as not to appear to be favoring one manufacturer over any other.

**Table 4.3 Example 1: The Shoestring Home Studio (for the Struggling Musician)**

Item	Approximate Cost
DAW software license	\$50
Basic PC with single monitor	\$800
Two-analog-input soundcard with preamps	\$120
One condenser microphone	\$120
Accessories (stands, leads, and so on)	\$200
Powered monitors	\$300
Headphones	\$40
Headphone amp	\$50
Approximate total	\$1,680

**Table 4.4 Example 2: The Budget Home Studio (for the budding band)**

Item	Approximate Cost
DAW software license	\$50
Software synth	\$200
PC with dual widescreen monitors, dual hard drives, and external hard drive	\$2,000
Eight-analog-input soundcard with preamps	\$600
Drum kit mic set	\$180
One dynamic microphone	\$100
Two condenser microphones	\$240
Powered monitors	\$600
Headphone amp	\$250
Headphones @ \$40 (2)	\$80
Headphones @ \$100 (2)	\$200
Accessories (stands, leads, and so on)	\$400
Approximate total	\$4,900
<b>Also worth considering...</b>	
Four-channel limiter	\$300
Extra microphones	\$240
Handheld recorder	\$200
Frontier TranzPort	\$200
Approximate grand total	\$5,840

In each case, I am assuming that a keyboard is already owned or not needed.

Decide on a realistic budget. Draw up a table similar to one of these and work out how you are going to allocate that budget between the various items that you will need.

Don't get demoralized if your budget looks a lot more like the first example than the last. Keep at it, and you'll be surprised at what you'll be able to do. You can also gradually add more and better-quality gear to your setup over time. You'll get there.

**Table 4.5 Example 3: The More Ambitious Home Studio (for the Bigger Budget)**

Item	Approximate Cost
Handheld recorder	\$200
DAW software license	\$50
Software synths	\$600
PC with dual widescreen monitors, dual hard drives, and external hard drive	\$2,000
Four-channel limiter	\$300
Twelve-analog-channel soundcard	\$1300
Eight-channel outboard preamp	\$600
Drum kit mic set	\$400
Three dynamic microphones	\$400
Six condenser microphones	\$1,200
Powered monitors	\$1,000
Headphone amp	\$250
Headphones @ \$140 (5)	\$700
Accessories (stands, leads, and so on)	\$500
Frontier TranzPort	\$200
Control surface	\$500
Approximate total	\$10,200

## Studio Layout

To close this chapter, we are going to return to the topic of studio layout that I mentioned earlier. For those who want to get really serious, whole books have been written on this topic, such as *Home Recording Studio: Build It Like The Pros* by Rod Gervais (Course Technology PTR, 2006). In this section, I will just explain a few of the key points and considerations that you will have to keep in mind.

### Noise Issues

The question of noise and possible acoustic treatment is a big one. We have already touched on some of the issues involved, and the best advice I can give at this stage is

to tread carefully at first. Remember that you will be addressing a number of different objectives, such as:

- Minimizing problems caused by sound leaking into or out of the room. Not only do you want a quiet environment for yourself, but you probably don't want to start a war with the neighbors, either.
- Ensuring a good acoustic environment for recording.
- Minimizing problems caused by hum and buzz.

### ***Dealing with Leakage***

There's not much I can say here about how you interact with your external environment. It depends on your studio location, neighborhood, and environment. You might wish to consider double-glazing or acoustic batts in the overhead if you are encountering real problems.

### ***Diffusers and Absorbers***

There are two ways in which material can be used to improve the acoustics of your room—by using diffusers and absorbers.

The role of the diffuser is to prevent excessive reflections—for example, by unwanted bouncing of sound from wall to wall. By attaching diffusers to the walls at a slight angle, you prevent the sound from being bounced straight back at itself. Simple diffusers can be constructed from plywood.

Absorbers, as the name implies, work by absorbing excessive levels of unwanted frequencies, thus removing them from the studio environment. This is the principle used in the construction of bass traps, for example. These issues are covered in depth in *Acoustic Design for the Home Studio* by Mitch Gallagher (Course Technology PTR, 2006).

### ***Hum and Buzz***

Problems with hum and buzz are not uncommon in home studios. These are the constant bass tone that you hear resonating at around 50 or 60 Hz. Unfortunately, the more equipment you have set up for use at the same time, the more likely you are to encounter this problem. Without getting too technical, let's take a look at some of the ways you can set about dealing with this.

Unfortunately, the cause can sometimes be something beyond your control, such as a neighbor using power tools. If this happens, all you can do is to try to negotiate some sort of give-and-take arrangement. You let me know when you're going to be planing and sanding, and I'll let you know when I'm going to be mixing.

More commonly, however, the cause lies within your own environment. Here are a few tips:

- Use only equipment with balanced inputs and outputs and balanced TRS and XLR cables. This alone might be enough to solve all your problems.
- Keep audio cables and electrical cables away from each other as much as possible, and especially away from power adapters and splitters. If your audio and electrical cables really do need to exist in close proximity, at least make sure they do not run parallel to each other. If they need to cross, make it at a 90-degree angle.
- Make sure you run all of your studio equipment on the same power circuit. This will ensure that all the items will find the one path to the ground, thereby hopefully avoiding a ground loop.

If you find that problems persist, there's a good chance you have one piece of equipment that is the culprit. By using a process of basically trial and error, you should be able to identify which one it is. Companies such as Behringer and Ebtech manufacture fairly inexpensive hum eliminators, which might solve the problem.

## Room Symmetry

Very commonly, the most serious and persistent problems with room acoustics will be found at the bass end of the spectrum. In short, the problem is that the level of bass response that is heard will vary substantially from one part of the room to another. A well-planned studio layout is one important aspect in addressing this problem.

Both speakers should be symmetrically positioned within the room; otherwise, each will have a different frequency response from the other. This will adversely affect your mixing. You will also get better results if they are placed with their backs to the shorter wall, facing out into the room. You will assuredly get a better representation of sound this way than if they are placed facing diagonally out into the room. They should, of course, be positioned at an angle so that their sweet spot, or optimum listening position, will be just at the point where you are sitting.

A good starting point is to obey the 38-percent rule. This suggests that the best place for listening will be 38 percent into the length of the room. That is the position at which you can expect to find the flattest frequency response. The theory behind this is quite complex, but essentially it depends on finding the position where the balance between peaks and nulls for different frequencies is at its optimum.

Other than that, be prepared to experiment and to learn from this. For where you're at now, you should be able to achieve an acceptable acoustic environment without the need to spend lots of money. That might come later!

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# 5

# The Road Ahead

**N**ow that you've taken your first steps in setting up your home studio, it's time to sit back for a moment and think about the future. In this chapter, we'll take an overview of how you can best prepare and be prepared for what lies ahead. We will be looking first at a learning path, then at those areas where you are most likely to need to plan for some further expenditure.

## The Learning Path

You've probably already realized from the 30-day course that the amount of learning that lies ahead of you is quite substantial. It will be worth your while to consider what options you have for making this easier. These might include one or more of the following:

- Read the documentation supplied with your DAW software. It's there to help you! If you encounter sections that you don't understand or that are too difficult at first, skip over them. You can return to them later. Understanding 50 percent is a lot better than understanding 0 percent.
- Enroll in a course. Keep your ears and eyes open for any opportunities that you might have to learn firsthand from real professionals. It's not uncommon for studios to offer part-time evening courses for very reasonable fees.
- Look out for more books, especially covering those aspects that are most important to you.
- Find online learning opportunities. Search the web for relevant sites, articles, videos, and so on. Participate in web forums.
- Practice, practice, practice. Then practice some more.

## Future Expenditure

When it comes to your studio and your equipment, the longer you get involved in recording, the more the following statement becomes true:

*You will never be satisfied with what you've got.*

It won't be long before you realize that the budget you allocated for setting up your studio was just the beginning. In a few pages from now, you'll be on your way. I'd just like to leave you with some thoughts that will help prevent you from wasting your money in the future and ensure that you get the best value from every single dollar you spend.

Let's take a look at five main areas where the cash will just fly out of your bank account if you're not careful.

- More plug-ins
- More microphones
- More gear
- More gizmos and gadgets
- More like a real studio

### **More Plug-Ins**

Spending excessive amounts of money on more and more plug-ins probably accounts for more wasted dollars than any of the other areas that we are considering. If you're not careful, you can end up spending 10 or 20 times the cost of your DAW software on plug-ins, and you may not necessarily get too much value from them. Let's look at a few tips that might help you keep your plug-in collection under some reasonable control:

- It is more important to understand and know how to use the plug-ins that you have than it is to have a big selection from which to choose. The rider to this is that the more plug-ins you have, the less you will understand them all. Make absolutely sure, for example, that you really can't get what you want from your DAW's native compressor or EQ plug-in before you splash your money around on another one.
- There is little point in having multiple plug-ins that essentially perform the same functions. It is true that different EQ plug-ins may color sound slightly differently, and different compressors may shape sound differently, but ultimately it is difficult to see how you can get much benefit from having more than two or three different items that perform similar functions. There's a good chance that one of each might suffice.
- There is no shortage of plug-ins that you can buy, often expensive, which are marketed as being uniquely suitable for a particular application, such as improving female vocals or creating crisper percussive sounds. However, the reality is that when you examine them closely, you'll often find that there's not all that much special, unique, or dazzling about their capabilities, only their packaging. Most

audio plug-ins essentially fall into one or more of three categories. These are EQ, dynamics, and time-based. Those plug-ins that are marketed for specific purposes in many cases do nothing that you could not do yourself with some combination of perhaps EQ, dynamics (such as a noise gate and/or compressor), and delay.

- It doesn't matter how clever, how good, and how widely praised a plug-in is, always stop and ask yourself the questions "Do I really need this?" and "Am I actually going to use this?" before purchasing it. You'll be surprised how much money this single piece of advice will save you.
- *Never* buy a plug-in over the Internet (or anywhere else for that matter) on impulse. Remember that unlike with many hardware items, if you change your mind, you don't like it, or it crashes your DAW, you can't return it for a refund. *Always* research first!
- Build your plug-in collection slowly, gradually, and over time. Its growth shouldn't be allowed to outpace the growth rate of your own knowledge, experience, and confidence.

There's any number of ways that you can categorize plug-ins. For the purposes of this discussion, we can categorize them broadly by cost. You can think of three categories: freeware plug-ins, budget-priced plug-ins, and full-price plug-ins. Let's look at each category in turn, in many cases along with some examples. In every case, the examples quoted are just that—*examples*. They might or might not be suitable for you, and I'm certainly not suggesting that there aren't any other products out there that might meet your own particular needs better than these.

### ***Freeware Plug-Ins***

This category of plug-ins is a real potpourri. You'll find some pretty good stuff here, some pretty awful stuff, and much in between. To help you in your quest, consider where they mostly come from.

- Reputable established software companies. It's not unusual for companies with a reputation for producing quality plug-ins to release and promote a freeware plug-in or two. This might be a lite version of an expensive big-sister plug-in (to tempt you into reaching for your credit card), or it might be a fully functional plug-in in its own right. Either way, it should be a quality product.
- Enthusiasts whose skill and knowledge levels are good and whose motivation simply isn't financial. You'll be likely to find some good stuff here.
- Well-meaning novices and amateurs. Don't say I didn't warn you.

One example of the first type is Kjaerhus's freeware Classic Reverb plug-in. You can find out more about this and other Kjaerhus plug-ins at [www.kjaerhusaudio.com](http://www.kjaerhusaudio.com). Coming from such a good pedigree, you can be reasonably assured that this will be of a decent quality. Incidentally, reverb is one area where there may well be advantages for you in getting hold of more than one product.

### **Budget-Priced Plug-Ins**

Budget-priced plug-ins as a rule come in at about \$20 to \$50 each. Often for \$100 or thereabouts, you can pick up a bundle of a half dozen or so different plug-ins from the same company.

For bread-and-butter plug-ins, such as EQ, noise gates, and compressors, it won't often make sense to purchase plug-ins in this category. This is because in all likelihood their functionality and quality will be no better than those supplied with your DAW.

Examples of plug-ins in this category are the Sonitus range. These are generally good products. You should not assume, however, that their EQ, Gate, or Compressor will necessarily do anything for you that your DAW's native plug-ins won't do. Where you might find them useful is in filling any gaps in the functionality provided by your existing plug-ins.

An example might be something like the Sonitus Modulator, which retails for about \$39. It contains a selection of modulating effects, such as flangers, phasers, and tremolo. If you need to apply such effects often, perhaps to a guitar, then a plug-in such as this might well find a role in your collection. Sonitus plug-ins are bundled in with Cakewalk's SONAR 8 (Studio and Producer editions) or can be purchased separately. More information can be found at [www.cakewalk.com/Products/Sonitus/sonitus.asp](http://www.cakewalk.com/Products/Sonitus/sonitus.asp).

Taking this idea a step or two further, Stillwell Audio produces an interesting range of plug-ins. Characteristically, they have an edge to them that pushes the functionality of audio plug-ins into interesting, quirky, and creative areas. These plug-ins may be budget priced, but there's nothing budget about their quality.

Olga (a frequency modulation synthesizer) and Bad Buss Mojo (whose primary purpose is to roughen up a recording that sounds too clean and too perfect) are two of the many interesting plug-ins available from this stable (see Figure 5.1). In both cases, you can download an evaluation copy for free and try before you commit to buying.

You can find out more about these and other items at [www.stillwellaudio.com](http://www.stillwellaudio.com).



**Figure 5.1** Olga—a quirky frequency modulation synthesizer.

### Full-Priced Plug-Ins

How much is full-priced? Probably from around \$200 and upward for a single plug-in. For that sort of money, you'd expect to get something special and to be absolutely certain that you're spending well.

One example of a product range with a good reputation is Wave Arts Power Suite 5. Its five individual components (TrackPlug, MasterVerb, MultiDynamics, Panorama, and FinalPlug) sell at \$199 each, or at \$599 for the complete collection. If you're a beginner, don't even think about purchasing anything like this yet. You'll know when the time comes to consider software as powerful as this.

It's worth making a couple more points here, though. First, take a close look at TrackPlug in Figure 5.2.

Does it look bewildering to you? If so, look again, more carefully. This plug-in is really five plug-ins in one, chained together. It includes EQ, a noise gate, two compressors, and a limiter, each of which can be enabled or disabled and can have its settings individually adjusted. This can be a very powerful and effective tool for getting the best sound out of different instruments (once you know what you are doing), and there is no doubt that having everything together in one window is convenient. You can learn more about the Wave Arts series of plug-ins at [wavearts.com](http://wavearts.com).



**Figure 5.2** TrackPlug from Wave Arts—five plug-ins in one.

Nevertheless, it's likely that *if you have enough knowledge and you know what you are doing*, you could get quite satisfactory results by simply chaining together these various component elements yourself, using the plug-ins (such as EQ, compression, and noise gate) supplied with your DAW. This can often be the case with many plug-ins.

The second example concerns mastering plug-ins. These include products such as Ozone and T-RackS.

I am mentioning this only because *mastering* is a term of which you have possibly heard, and in that context we cannot entirely ignore it. The process of mastering consists of bringing your final mix up to a professional standard, where it is ready for distribution. This entire topic is way, way beyond the scope of the beginner, and as such really doesn't merit any further consideration just now.

There are other items that one day you may aspire to be good enough to use and wealthy enough to afford. When you have learned everything you need to know about microphone selection and placement, recording techniques, audio editing, mixing, EQ, dynamics, delay, and so on, then you might wish to think about whether you could use some of this professional-level software. Depending on your goals, you might one day have a need for it.

## More Microphones

The type of budget-priced microphones we discussed in Chapter 4 should stand you in good stead for at least the first year or so of your home recording venture. However, the time may come when you'll want to invest in some microphones that are just a little bit better. Whether this happens will depend as much as anything on your needs and objectives. This is another one of those issues that you will need to address later on down the track but which you should at least be aware of now.

A budget of around \$500 to \$800 brings you within reach of an impressive selection of microphones. When used well, a tube microphone can add clarity and body to your recordings, especially to vocals. It has its own independent preamp and power supply. Examples of this kind of microphone are the Studio Projects T3 (which also has a fully adjustable pattern) and the Rode K2.

Other microphones in this price range worth considering include the Baby Bottle (from Blue), the C 214 from AKG, or, if you're thinking about ribbon microphones, the Groove Tubes VELO 8.

Whether it's worth your while to ever spend any higher than this will ultimately be your call. The determining factors have all been discussed before—your budget, your recording environment, the musical standard of the person being recorded, your ability and experience as an audio engineer, as well as the standard for which you are aiming are all issues that come into play here.

## More Gear

There's no end to your possible wish list. Always remember to reign yourself in with a reality check. Will that \$2,000 preamp really make you any better at what you're trying to do? Of course, better quality gear can, in the right circumstances, help you secure better outcomes, but it can never compensate for any limitations in your knowledge, skill, and experience.

## More Gizmos and Gadgets

Look through any audio store's catalog or website, and you will be amazed at the number and variety of gadgets and devices that you can purchase for your home studio. As ever, ask yourself the questions, "What do I really need?" and "Will I actually use this?"

At some stage in the future, you will almost certainly reach the point of considering (and quite possibly make the commitment to purchasing) a control surface. There is no doubt that one or perhaps more of these devices can make life a lot easier and more enjoyable when you are mixing your projects, tweaking your parameters, and so on. That will be the time to explore your options in detail, but for now here is a general overview of the subject.

You can think of most control surface devices as fitting into one of two categories: general purpose or special purpose.

### ***General-Purpose Control Surfaces***

This type of control surface usually is used for operating the everyday controls of your DAW software, such as adjusting the volume and pan faders of different tracks, using the solo and mute functions, starting and stopping playback, and so on. It effectively provides a real desk for your interface, as opposed to having to use just the mouse and PC keyboard all the time.

The prices and capabilities of these devices vary enormously. At the budget end of the scale are products such as the Behringer BCR2000 and BCF2000, priced at about \$200 and \$300, respectively. At the premium end, you'll find products such as Mackie's modular Control Universal Pro series, which sells at about \$1,200 per module.

### ***Special-Purpose Control Surface***

The other kind of control surface is developed to perform a specific task or series of tasks. One example of this is the Novation Nocturn, specifically used for controlling the parameters on your plug-ins. This enables you, when using your favorite EQ plug-in, to use real knobs to adjust parameters such as gain, frequency, and bandwidth rather than having to be dependent on the mouse, for example.

Another example of such a device is the Evolution UC-33e, which is designed specifically for use with software synthesizers.

### **More Like a Real Studio**

This is a subject that we've touched on from time to time throughout this book. In most cases it is quite literally the last area that you should look at when spending your money. That isn't to say that it's not important—rather, that features such as an acoustic vocal booth and a separate control room aren't going to serve you well if your equipment is of a poor standard or you don't know how to use it to its best advantage.

By way of example, I can truthfully say that one of the worst recorded tracks I ever had brought to me for mixing was of an excellent vocalist using an excellent microphone and was recorded in a vocal booth. The problem was that the microphone placement and her microphone technique were both way off.

That's it! Good luck, and if you get even a quarter of the pleasure that I've gotten out of recording over the years, then your journey will have been well worth it. Hopefully, after reading this you'll also get only a fraction of the frustration.

# Appendix: Glossary of Basic Audio and Recording Terms

**analog audio.** This is sound that is continuous, corresponding exactly to real sound. This is unlike digital sound, which is broken down into a series of ones and zeros. An analog signal is usually recorded using magnetic media, such as tape. Cassette tapes and vinyl LPs, for example, both use analog sound. Analog recordings cannot be copied without some loss of sound quality and are likely to contain more unwanted noise than digital recordings.

**ASIO.** Audio Stream Input-Output. This is an audio protocol originally developed by the Steinberg company and used by REAPER to communicate with professional-standard audio hardware devices and interfaces (such as PCI soundcards and FireWire and USB devices).

**bit depth (sample size).** The bit depth is the level of detail at which a computer samples analog audio to create digital audio. When recording WAV files, most commonly, you use 16- or 24-bit sampling. 24-bit audio is generally preferred because it gives a more accurate representation of the sound and makes it easier to avoid clipping, but it does take up more disk space. Bit depth should not be confused with sampling rate.

**bit rate.** This measures the number of kilobits per second of data in MP3 and other audio files. The bit rate you choose when creating an MP3 file determines the size and quality of the resulting MP3. The highest commonly used bit rate is 320 kbps. A file created using this bit rate will have excellent quality but will be fairly large. A standard bit rate for encoding MP3s is 128 kbps. A file created using this bit rate will have good quality and will take up about 1 MB per 1 minute of sound.

**buffer.** An audio buffer is a driver setting that helps determine the rate at which audio passes between your computer's processor and its soundcard. Reducing the buffer size can help reduce the amount of latency while recording and monitoring audio. Increasing the buffer size can help prevent pops and clicks while recording.

**burn.** This is the process of writing data or files onto a recordable CD using a hardware device called a *CD burner*. Generally, you create either an audio or a data disc when you burn a CD (although hybrid formats such as CD Extra are also available). Audio discs

can be played in any standard audio CD player. A data disc contains computer files and can only be read on computers. REAPER supports the burning of audio discs.

**bus.** An internal pathway that may form part of your audio routing system. Although REAPER makes no inherent distinction between a track and a bus, a track can be considered to be functioning as a bus when more than one other track has been routed to it.

**channel.** A channel is a path through which an audio signal flows. One important feature of REAPER is its ability to use up to 64 separate channels with any single track or tracks.

**chorus.** An effect that makes one voice or instrument sound like many. Chorus works with all types of audio but is particularly effective with the human voice.

**clipping.** This is the unpleasant distortion you hear when output is too loud, causing the peaks of the audio signal to rise above the capabilities of the amplifier circuit. To avoid distortion, reduce the volume or gain before the stage in which the clipping occurs.

**CODEC (or codec).** Codec stands for compression/decompression. A codec is a program used to enable Windows to compress and/or decompress audio to and from different formats. For example, REAPER uses the LAME codec for creating MP3 files.

**compression (audio).** A process of reducing the dynamic range of an audio stream. Often used to make the loud parts of a track quieter and the quiet parts louder. Not to be confused with data compression, which is an entirely separate and unrelated topic.

**compression (data).** The process of packing digital data, such as computer files, more efficiently for the purpose of storage or transmission. MP3 is one example of a commonly used compressed audio format. Not to be confused with audio compression, which is an entirely separate and unrelated topic.

**control surface.** An external device connected to your computer by MIDI cable, Fire-Wire, or USB and that is used to physically control various parameters and functions of your DAW.

**decibel.** A unit used for the measurement of sound. Commonly, sound pressure levels (SPL) are represented as numbers from 0 dB (the softest sound that may be heard) to 120 dB and beyond (the level at which sound is perceived as pain).

**delay.** An effect that creates a delayed sound.

**digital audio.** Digital audio is audio that has been converted into a series of ones and zeros that can be processed by a computer. When analog sound is converted in this way,

it is commonly saved and stored as a WAV file. Digital sound is easier to reproduce and manipulate without loss in quality than analog.

**digital audio workstation (DAW).** A term used to describe a computer when it has been set up and equipped with the necessary software and hardware to function as a recording studio.

**DIRECTX.** A widely used plug-in format, generally less popular than VST.

**Docker.** The REAPER Docker provides a tabbed viewing area for several REAPER functions (mixer, FX browser, undo, routing matrix, and so on) and is accessible through the View menu. The Docker can be detached from the main REAPER window and moved, for example, to a secondary monitor.

**driver.** Software that works with your computer's operating system to control and use a particular piece of hardware, such as your soundcard. To enable it to function at its best, you should check the web regularly for any updated drivers for your soundcard.

**dry.** A term used to describe an audio signal to which no effects have been added. The opposite is wet.

**encoding/decoding.** The process of converting audio to or from a compressed format, such as MP3 or FLAC. The encoding and decoding processes require the use of codecs.

**envelope.** A technique used to control how any of a number of a track's attributes behave over time. For example, an envelope can be used to automatically fade a track's volume up and down as required for different parts of a song whenever it is played back.

**equalization (EQ).** Sound is made up of many vibrations that take place at the same time at different frequencies. An audio equalizer lets you separately adjust the volume of different ranges of frequencies, thus changing the makeup of the overall sound. Frequency is measured in hertz (Hz). The higher the number, the higher the pitch of the sound.

**fade.** A technique to bring sound into or out of a track gradually. Fade-in brings the sound in gradually, fadeout does the opposite. Cross-fade occurs when two media items overlap in such a way that one is faded in while the other is faded out.

**fader.** A device that enables you to control the level of an audio signal by sliding the fader up or down. Examples of the use of faders in REAPER are to control a track's volume and panning.

**flange.** This is an audio effect that distorts sound by applying both a short delay and variable modulation of the frequency.

**folder.** A folder is a collection of related tracks and serves as a simple and convenient means of controlling all of the tracks contained within it. For example, changes can be made to the volume level of a folder to affect the combined volume of all of the tracks contained within that folder. Changes made to the volume level of individual tracks within a folder will affect the volume of that track and the relative balance between all tracks in the folder.

**frequency.** The number of vibrations in a sound wave per unit of time. Frequency is usually measured in hertz, where one hertz is one cycle per second. High-pitched sounds have higher frequencies, and low-pitched sounds have lower frequencies.

**glue.** To glue a track causes REAPER to render the selected items to a new single WAV file, which replaces the original items.

**high-pass filter.** A high-pass filter removes all sound below a set frequency. It can be useful in removing certain kinds of rumble and hum.

**input monitoring.** The process of having the audio stream that is being recorded also fed directly back to the musician or singer through his or her headphones. If latency is too high, a perceived delay or echo effect will be noticeable.

**items.** Items (a.k.a. *media items* or *media events*) contain audio/MIDI information and reside in a track. Items can be empty or can have one or more *takes*, one of which is visible and “active.” Items are often called *parts* in other DAW software.

**latency.** Latency is a measurement of the time taken for audio to travel from the sound-card, through REAPER, and out again to your headphones or speakers. Usually measured in milliseconds. For input monitoring, a latency of less than 5 ms is usually preferred.

**level.** The amount of power that drives an audio signal. Common names given to varying levels of voltage (from lowest to highest) are microphone level, instrument level, and line level.

**lossless compression.** A method of compressing the size of audio files without losing any of its frequencies. Examples of lossless compression formats are FLAC and OGG Vorbis.

**low-pass filter.** A low-pass filter removes all sound above a set frequency. It is useful in removing certain kinds of hiss.

**lossy compression.** A method of compressing the size of audio files that includes stripping out those frequencies that the human ear either doesn’t hear or hears poorly. Examples of lossy compression are WMA and MP3.

**markers.** Markers, like regions, are used to identify important sections of a song (for example, verse, chorus, intro, and so on). Unlike regions, markers indicate only the beginning point of a section and are displayed in the Markers band of the Timeline. Markers are useful for quickly locating and playing a section.

**master.** The section of REAPER's mixer where the main mix is controlled.

**media item.** A media item is an audio or MIDI recording. In many DAWs, these are known as *clips*.

**MIDI (Musical Instrument Digital Interface).** A system that uses a synthesizer to enable your computer to play back music under program control. MIDI instructions can include which notes to play and which instrument to simulate. MIDI can also be used to enable a computer to control external MIDI-compliant instruments, such as synthesizers.

**Mixer view.** REAPER's Mixer view depicts a virtual representation of a traditional hardware mixing desk. This can be used to combine all of the signals from your various audio and MIDI tracks into a main mix, usually consisting of one single stereo-paired output.

**MP3.** MPEG audio layer 3. An audio format with data compression that is widely used to transfer music over the Internet. An MP3 file is much smaller than a WAV file, but it still sounds to most people's ears virtually as good as the original.

**noise gate.** A noise gate is a filter that detects sound levels in an audio stream and shuts out sound when the volume falls below a determined level. It can be useful, for example, to cut out unwanted background noise that may be present during an otherwise silent passage on a recording.

**normalization.** This effect permanently adjusts the amplitude (volume, essentially) of audio to a certain value. Normalization is particularly useful when creating a CD using various different tracks. Normalization can be used to help obtain a consistent level of volume for the different tracks.

**notch filter.** This filter removes all audio in a specified frequency band in a WAV file.

**OGG Vorbis.** OGG Vorbis is a free, open-source audio compression format. Though less widely used than MP3, it can provide greater fidelity.

**pan.** A control that lets you determine the relative left-right balance of a mono signal within the stereo spectrum.

**plug-in.** A small program used in conjunction with REAPER to enable you to manipulate the sound of recorded tracks. Functions carried out by plug-ins include EQ, delay, and chorus. Two widely used plug-in formats are DirectX and VST (Virtual Studio).

**real-time.** Refers to effects that take place as you listen, without a noticeable delay.

**receive.** The method by which audio sent from one track is accepted by its destination track or bus.

**regions.** Regions, like markers, are used to identify important sections of a song (for example, verse, chorus, intro, and so on). Unlike markers, regions indicate both the beginning and ending points of a section and are displayed in the Regions band of the Timeline. Regions are useful for relocating/duplicating existing sections of a song.

**reverb.** An effect that simulates natural reverberations (sound reflections) that occur in different rooms and environments to create an ambience or sense of space.

**reverse.** An effect that takes one or both channels in a sound file and plays them backwards.

**rip.** To extract music directly from your CD in pure digital form and save it directly to your hard drive.

**routing.** A term used to describe the path(s) taken by an audio stream that, on playback, takes it from the track on which it has been recorded all the way to its position in the final mix.

**Routing Matrix.** REAPER's overview of a project's entire routing network. Changes to routing can also be made in the Routing Matrix.

**sampler.** A hardware device or software application that uses samples to generate audio output. Samplers often use a number of samples together to create realistic-sounding reproductions of real sounds and musical instruments.

**sampling rate.** The sampling rate is a measurement of how many times a second your audio recording software captures the incoming audio signal. Commonly used sampling rates are 44.1 kHz, 48 kHz, 88.2 kHz, and 96 kHz. For example, at 44.1 kHz, incoming audio is sampled 44,100 times per second. The higher the sampling rate, the more disk storage required. If you're not sure which rate to use, try recording initially with a sampling rate of 44.1 and see whether you are satisfied with the results.

**send.** The method by which audio is routed from one track to another track or bus, where it is accepted via a receive.

**stems.** Stems are a group of selected tracks and/or folders. They allow you to render selected tracks to disk at the same time as (or in lieu of) the main mix. You can use stems to "freeze" FX on tracks or to render each track in a mix (a *stem*, as it is called) so that a mastering engineer (or whoever) can later adjust the mix.

**synthesizer.** A hardware or software device that artificially (using oscillators) generates signals to simulate the sounds of real instruments or to create other sounds not possible with real instruments.

**takes.** A take is a part of an item that contains a media source (audio, MIDI, or other type). An item can have multiple takes, which may refer to different recorded versions (where it gets its name). When you adjust an item (splitting, adjusting its start/end times, stretching, and so on), the action is performed on all takes, so that if you need to switch to a different take, the overall timing is correct.

**tracks.** A REAPER project is made up of any number of tracks that you record (or import). Tracks can contain multiple media items and envelopes. Tracks appear as horizontal bars on the Timeline and as vertical bars on the mixer. REAPER offers three types of tracks: 1) standard tracks, which hold items and envelopes; 2) folders, which hold items, envelopes, and standard tracks; and 3) stems, which include selected standard tracks/folders.

**Track Control Panel (TCP).** The area (to the left of the REAPER screen in Track view) where a project's various tracks are listed. So called because it is used to control the various track attributes, such as to arm for recording or to control volume on playback.

**VST and VSTi.** A widely used plug-in format, Virtual Studio Technology, developed originally by Steinberg. The term *VST* usually refers to FX that manipulate or modify the sound of audio in some way. *VSTi* refers to a virtual *instrument* used for sound synthesis.

**wave (.wav) files.** A wave file is an uncompressed audio file on your PC. This enables it to be as close a copy to the original analog data as is possible.

**waveform.** A visual representation of a WAV file.

**WAV editor.** Audio software designed for editing digital audio.

**wet.** A term used to describe an audio signal to which effects have been added. The opposite is dry.

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