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EDIT, MIX, AND MASTER WITH T H E FREE AUDIO EDITOR

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Record, Edit, Mix, and Master with the Free Audio Editor

by Carla Schroder

San Francisco

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Thank you, TJ! It just keeps getting better!

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Thanks everyone!

Audacity is an open source, free-of-cost, cross-platform audio recorder, editor, and mixer for Linux, Windows, and Mac OS X. It comes packaged in easy-to-use installers for Mac OS X and all versions of Windows, and Linux users will find it in the software repositories of their favorite Linux distributions. Visit *http://audacity.sourceforge.net/* for downloads, documentation, and mailing lists.

In this book, we'll be using Audacity 1.3.12 (and newer) on Ubuntu Stu- dio and Microsoft Windows XP, Vista, and Windows 7. The stable 2.0 release will appear soon and should look very much like what you see in this book because the 1.3.*xx* series is the run-up to 2.*x*. This book is based on the very latest releases as they came out, so it is as current as any book can be. The 1.3.*xx* Audacity releases are considerably advanced from the old 1.2.*x* se- ries. Every new release is full of wonderful improvements and bug fixes, so if you're still using those old 1.2.*x* versions, you should consider upgrading. Ubuntu Studio is Ubuntu with a huge set of multimedia applications. It is 100 percent Ubuntu-compatible, and it uses the standard Ubuntu software repositories. You can download Ubuntu Studio or simply add the Ubuntu Studio packages and artwork to any Ubuntu installation. There are several

INTRODUCTION

excellent multimedia Linux distributions, which you can read about in Chap- ter 13. You can use any Linux version you like; a few important system modi- fications you may need to make are covered in Chapter 13.

Windows requires some modifications too, which you'll find in Chap- ter 14. Since Windows XP continues to hang on and refuses to enter retire- ment, you'll find information for Windows XP, Vista, and Windows 7.

What Can Audacity Do?

Audacity is fast and easy to use. What can you do with it? A whole lot:

- Work with a wide number of different audio file formats and encod- ings, including WAV, AIFF, MP3, FLAC, AU, OKI, MAT4/5, Ogg Vorbis, WMA, M4A, and AC3.
- Record live audio.
- Convert legacy analog media to digital.
- Make movie soundtracks.
- Perform unlimited multichannel recording.
- Edit and mix multiple tracks.
- Overdub.
- Use special effects of all kinds: wah-wah, change pitch and tempo, bass boost, echo, reverse, phaser, and more.
- Add graceful fades, both in and out.
- Normalize volume levels.
- Fix defects such as hiss, static, pops, and hum.
- Perform frequency analysis.
- Write your own plug-ins for special effects.
- Cut, copy, splice, and mix sounds together.

Audacity can open and edit audio files faster than most other audio applications and has nearly unlimited undo and redo.

So, what can't you do with Audacity? Audacity does not support the RealAudio format, and it does not support MIDI. While it is wonderful for making mono and stereo recordings, it is not quite as good at making multi- channel surround sound recordings.

What This Book Covers

In this book, we'll use Audacity in a number of (I hope) fun and useful au- dio projects. In Chapter 1, we'll plug a microphone into a computer and learn the basics of recording, editing, playback, and Audacity controls. We'll also learn some important digital audio terminology and concepts. If you are new to digital audio production or new to Audacity, you should go through this chapter first.

In Chapter 2, we'll go into detail on audio gear, how to select it, how to connect it, and how not to spend too much money. The world of audio gear is vast and confusing, but this chapter sorts it all out for you.

If you're like me and have a hoard of treasured vinyl LPs, 45s, or vintage 78s that you want both to enjoy and preserve, read Chapter 3 to learn how to copy them to your computer. From there, you can transfer them to CDs or export to MP3 or any other digital audio format you want. You can do the same with any kind of legacy media.

Audacity is a great program for recording live shows or for editing record- ings of live shows made with portable recorders. Chapter 4 shows you how to clean up and optimize your recordings for compact disc or DVD-Audio. Chapter 5 goes into more detail on making audio CDs and compila- tion CDs. You'll learn how to normalize different volume levels, break long files into separate tracks, transition smoothly between tracks, and edit track metadata.

In Chapter 6, we learn how to author super high-fidelity DVD-Audio discs. DVD-Audio is a special audio standard for DVDs; it is not the same as the audio formats used on movie DVDs. With DVD-Audio, you can au- thor very high-fidelity DVDs or load several CDs' worth of music onto a sin- gle DVD.

Podcasts are all the rage, and Chapter 7 tells you how to make podcasts that sound good and are bandwidth-efficient, and it covers the basics of In- ternet streaming audio.

Chapter 8 goes into detail on making the highest-quality audio record- ings for distribution and tailoring your releases for different types of distri- bution, such as Internet radio, downloadable formats, and CD. It also offers some guidance on finding distributors and other business basics.

Audacity handles multitrack recording capably, so Chapter 9 shows you how to record multiple tracks, mix, dub, edit, and mixdown to your final mono, stereo, or multichannel surround release. You can play or sing along to an existing track, record as many tracks at once as your recording inter- face supports and your computer can handle, and mix separate recording sessions together.

Don't pay for ringtones—study Chapter 10 to learn how to make your own easily. Ringtones need to be not too big and not too small, and they can be any snippet of music or sounds or even your own voice. Learn some tips for tailoring your ringtones to sound better on the tiny lo-fi speakers of your phone.

You can go nuts playing with special effects in Audacity—strange noises, sound effects, echo, wah-wah, bass boost, tremolo, and so on. Chapter 11 introduces you to a number of them, tells you where to get more, and shows how to learn to write your own.

In movies and television, ace crime techs take shredded audio rem- nants and create detailed, high-quality, beautiful recordings as they natter about their magic algorithms. It's all hooey. But you can do a lot to clean up recordings afflicted with pops, hiss, and other defects, and Chapter 12 tells how.

Chapter 13 details how to select a Linux distribution for audio produc- tion, how to configure it for best performance, and how to troubleshoot and fix common problems.

Chapter 14 covers the important tweaks Windows users need to make for quality audio recording.

Audacity is easy to use, but it has a number of options that may not make sense to anyone who isn't already an audio engineer, so Chapter 15 goes in- depth into customization and configuration.

Appendix A is your hardware reference; you'll find examples of audio hardware in several price ranges that work on both Linux and Windows.

Appendix B is a glossary of audio terminology written for real people; that is, people who are not physicists or audio engineers and who appreciate clear explanations in plain English.

Appendix C debunks popular but silly audio myths and saves you from some common—and expensive—mistakes.

NOTE *Audacity is also available as a source tarball. What do you do with a source tarball? It contains Audacity's source code in a compressed archive. You can install Audacity from source code if you want and* *customize the compile-time options, examine the code, modify it, or even modify and redistribute it. If you're feeling adventurous and want to help debug daily builds, you can grab the newest Audacity version from Concurrent Versions System (CVS) and give it a test-drive.*

Audacity vs. Ardour

Another popular audio recording application for Linux (and Mac OS X) is Ardour, which calls itself a digital audio workstation. Ardour aims to meet the needs of professionals and competes with the likes of ProTools, Nuendo, Pyramix, and other expensive commercial audio applications. It has a more sophisticated mixer than Audacity and some nice audio-for-video tools. It has advanced dubbing abilities, synchronizes with MIDI sequencers, and supports control surfaces, which are hardware devices for controlling your software mixers. It has more automation, as well as a number of useful real- time features such as changing plug-ins on the fly and moving samples to different tracks or timelines while they are playing.

Which one is better? That depends on what you want to do. Both are 100 percent free software because they are licensed under the GPL, both are excellent, and both are getting better all the time. For complex multitrack mixing or precise video soundtrack synchronization, go with Ardour. For recording long tracks such as live shows, converting LPs and tapes to digital formats, cleaning up files marred by hiss or hum or other defects, making podcasts, making simple video soundtracks, and recording in the field, Au- dacity is an excellent, quality application with a short learning curve and a lot of useful and advanced features.

Let's fire up Audacity and make a record- ing. We'll begin with a quick-start tutorial and make a simple recording to demonstrate basic usage. Then we'll cover the fundamental Audacity functions in detail from start to finish: record- ing, performing common editing tasks, saving your work, and exporting to various audio file formats and quality levels.

We'll deal with fancy audio hardware later; for now, all you need is any Linux or Windows computer with an ordinary sound card and either a mi- crophone with a 1/8 mini-plug or a USB microphone or headset. Any microphone will do for this initial test, even a little cheapo computer mi- crophone. I recommend an external microphone because built-in computer mics sound pretty bad and are positioned inconveniently. Of course, if you have something better and know how to hook it up, by all means use it.

USB devices need to be plugged in before you open Audacity. If you change a USB device while Audacity is open, you'll have to close and reopen Audacity for it to detect the change.

1

AUDACITY FROM START TO FINISH

Figure 1-1: A new, blank Audacity window on a Windows PC

Figure 1-2 shows a new Audacity window on an Ubuntu Linux PC. As you can see, Audacity is nearly identical on both platforms. The only signif- icant difference is the recording and playback device chooser. The audio subsystems on Linux and Windows are very different, so the device choosers present different options.

Select **Edit** > **Preferences** > **Devices** to set up your default recording and playback devices. These can be overridden easily from the main Au- dacity window using the Device toolbar. Figure 1-3 shows a Plantronics USB headset selected on a Windows PC. (Chapter 15 goes into detail about con- figuring and customizing Audacity.)

USB devices always announce themselves by name, so you don't have to guess. For example, on both Linux and Windows, the recording device selector will say "Plantronics Headset." If you plug a microphone directly into an internal sound card, you will need to know the name of your sound card's driver. On Windows systems, don't select MME, which is the anti- quated, generic Windows audio interface. You want to select the modern Windows audio subsystem, which in the **Edit** > **Preferences** > **Devices** dialog

Audacity Quick-Start

Okay then, enough fiddling around (unless you're going to play a fiddle)! Let's make a quick recording, because that is more fun than sitting around reading about it. Figure 1-1 shows what a new Audacity window looks like before you make a recording on a Windows PC.

Figure 1-2: A new, blank Audacity window on an Ubuntu Linux PC

Figure 1-3: Select **Edit** > **Preferences** > **Devices** to set your default recording and playback devices.

appears as “Windows DirectSound” in the Host line. Figure 1-4 shows what the selections should look like on a laptop with an onboard SoundMAX au- dio chipset.

On Linux, you’ll have even more choices. “ALSA:default” on the Device lines will work for an internal sound card (unless you have changed the de- fault sound device for your Linux system; see Chapter 13 to learn all about Linux audio). Pick the device name for a USB device. When you’re finished, click **OK** to close the Preferences dialog.

lly start recording. Go to the Input Level Meter, shown in e 1-5. Click **Start Monitoring** and start making noise. e ana- log recording, with digital audio you don't need to your recording lev- els right up to the redline. Try record- a peak of −6 or −9 dB.

Figure 1-5: Input Level Meter

Now let's test recording levels be- fore we

You can use the Mixer toolbar to control recording and playback volume, sort of. It isn't really a mixer but a recording and playback volume control. This is the little toolbar with the speaker and microphone icons and volume sliders for each. It does not con- trol volume on all internal sound cards, because some low-end sound cards do not have drivers that support volume control. It may not control volume levels on USB devices either, again depending on what their drivers support. If this is the case, in Windows go to the Sound module in the Control Panel to control volume levels. Linux users should use alsamixer. (Remember,

Figure 1-4: Selecting your default recording and playback devices on Windows Vista.

NOTE *Doing digital audio production on a PC means you'll have to get very involved with your sound card drivers and in configuring your PC for good quality and perfor- mance. Visit Chapters 13 and 14 to learn how to tune your system for audio produc- tion and how to manage various operating system quirks for controlling volume levels, balance, and input and output devices.*

Before you start recording, save and name your new Audacity project by selecting **File** > **Save Project As**. It is a good habit to do this right away for every new recording.

In the next section, we'll learn all about all the tool buttons. For now, hover your cursor over the toolbars and buttons to learn their names.

Chapters 13 and 14 will help with these.) Or you can just make louder or quieter noises.

The Input Level Meter uses two different shades of red: bright red bars for displaying the average volume and dark red bars to show the peak vol- ume levels. The little vertical blue lines mark the highest volume levels at- tained during the session, and the little vertical red lines mark the peak volume levels of the last three seconds. On the right edge of the recording monitor are clipping indicators that will turn red when your recording level is too loud. They're pretty small and stay lit after your recording levels drop, which limits their usefulness. However, you do need to pay attention to clip- ping, which occurs when input levels are too high. Anything over 0 dB cre- ates clipping, and clipping causes distortion.

Now let's record some sounds. Click the red **Record** button and keep making noise. You'll see something like Figure 1-6. When you're finished, click the **Stop** or **Pause** button. With the Stop button, a new track starts the next time you click **Record**; the Pause button lets you pick up where you left off on the same track. If you stop when you meant to pause, don't worry— you can append to an existing track by pressing the SHIFT key and clicking **Record**.

Figure 1-6: At last, a recording session! When you see blue waveforms, you know it's working.

Naturally, when you're done recording, you'll want to hear what you just recorded, and Audacity offers instant gratification. Click the **Play** but- ton. If you don't hear anything, it's because you either selected the wrong playback device or have a volume control set too low. Stop playback before changing the playback device. The cursor changes to a little hand when you hover over the Time Scale, and you can click any point on the Time Scale to start playback again.

Figure 1-7: How to select a whole track

re 1-8: Normalization uniformly raises (or lowers) the volume level of

The final step is to export your new recording to a playable audio file. Audacity uses its own special file format that doesn't work in anything but Audacity, so you have to export to an audio file format that works in play- back devices. Select **File** > **Export** to export the project as a WAV file, which should be the default choice (Figure 1-9). Name your export file whatever you want, maybe something creative like *test.wav*. The WAV format is almost universal and will play on nearly any digital playback device or computer software media player.

Now you can play your *test.wav* file on your computer and hear it in all its glory. Windows users can use Windows Media Player, which is installed by default, or choose from a host of third-party programs. Linux users also have any number of media players to choose from: Amarok, Rhythmbox, VLC, Mplayer, and many more.

It is best to use WAV as your default export format because it is a loss- less, uncompressed format that provides the highest-quality recordings. WAVs stand up to a lot of editing without deterioration, whereas lossy for- mats (such as MP3 and Ogg Vorbis) lose information with each edit. You

In digital audio it is common practice to record to a low peak level, as low as −24 dB. The digital audio decibel scale is measured in negative num- bers up to zero. About the smallest change we can perceive is 1 dB, and −60 dB is as good as silence for most people, so a practical range to use is −60 to 0 dB. A +3 dB change doubles the volume, and −3 dB halves it.

A super-low peak such as −24 dB is useful when you're recording some- thing with unpredictable levels, such as a live performance. For other, more controlled circumstances, a good peak level is between −12 dB and −6 dB. Any sound level over 0 dB will result in clipping, which creates distortion. Avoiding distortion is very important in digital audio recording. The signal- to-noise ratio is extremely high, so you don't need to push your recording levels to the maximum just to keep noise at tolerable levels.

A low peak level means that your recording won't be very loud, but that is no problem. You can easily fix this. Select the whole track by clicking the track label (Figure 1-7). Then open **Effect** > **Normalize**. Check both boxes in the Normalize dialog and make the maximum amplitude 0 (Figure 1-8).

Figure 1-9: Saving your new audio file in WAV format

can always export from WAV to a lower-quality, lossy format, but you can't go from low to high quality.

All righty then, that's the short story. Read on to get the unabridged version.

Audacity in Detail

Keep in mind that Audacity supports nearly unlimited undo, so it is safe to experiment. Undo works even after saves; you lose your undo history only when you close your project file.

When you work on a project, Audacity does not operate directly on your audio files. Instead, it copies them into a temporary file, chops them into a lot of little pieces, and converts these to files with *.au* extensions that play only in Audacity. You can see this by viewing your project directories in any file manager. There is a single *.aup* file for each project; this contains all the metadata Audacity needs to put these little files back together with the cor- rect settings. When you open an Audacity project from your file manager, select the *.aup* file.

Suppose you have a recording of a splendid performance where you outdid yourself and brought tears to all eyes (of joy, not pain) and this re- cording is in WAV format. When you import this WAV into Audacity, it is copied and converted to Audacity's internal *.au* format. Your original WAV is safe and sound and won't be changed, as long as you don't overwrite it by exporting the project back to the same WAV file.

Figure 1-12: Edit toolbar

Table 1-1 lists all the buttons found on the Tools and Edit toolbars with descriptions of what they do.

Converting and splintering your files might sound like an odd thing to do, but operating on many little files is a lot faster than manipulating a few large files. Audio files can consume many megabytes and even gigabytes. Au- dacity has an automatic crash recovery mechanism, which you will see only if something bad happens; when you reopen Audacity, it will display a recovery message. Unsaved data are kept in a temporary file, so Audacity can usually recover them. Select **Edit** > **Preferences** to set an autosave interval; mine is at two minutes. And, as with everything we do on computers, good backups are essential. Hard disk space is cheap these days, so don't pinch pennies on storage.

Let's start our detailed tour with a look at Audacity's toolbars. All of the toolbars have handles on their left sides so you can drag them anywhere you want, even outside of the Audacity window. If you hover the cursor over the toolbar handles, the toolbar name pops up. Hover over the buttons to see their names. Select **View** > **Toolbars** to control which toolbars are visible. Figure 1-10 shows the Control toolbar, which has the Pause, Play, Stop, Skip to Start, Skip to End, and Record buttons.

Now let's meet the buttons on the Tools toolbar: Selection, Envelope, Draw, Zoom, Time Shift, and Multi-Tool (Figure 1-11). These affect the cur- sor functions.

Figure Figure 1-10: Control toolbar
1-11: Tools toolbar

Next to the Tools toolbar is the Edit toolbar (Figure 1-12), which con- tains the Cut, Copy, Paste, Trim, Silence, Redo, Undo, Link Tracks, Zoom, Fit Selection, and Fit Project buttons.

Table 1-1: The Tools and Edit toolbar buttons

Button Name	**Description**
Selection	Click to mark a playback starting point. Click and drag to select a portion of a track. Double-click to select a whole track. Click anywhere on the Time Scale to start playback (it changes to a little hand).
Envelope	Use for fine control of amplitude (volume levels) on a track and for creating fade-ins and fade-outs. Click to create control nodes, and then click and drag nodes to increase or decrease amplitude. Control nodes can be dragged both vertically and horizontally. Drag nodes past the track border to get rid of them.
Draw	Click the Zoom In button until you can see in- dividual audio samples and then use the Draw tool to manipulate them. Use this for very fine- grained smoothing out of clicks and pops.
Zoom	Left-click to zoom in, right-click to zoom out. Remember the Zoom buttons! You will probably use them a lot: Use Zoom In for precise edits and use Zoom Out to make long tracks manageable. See the View menu for more Zoom commands and keyboard shortcuts.
Time Shift	Synchronize tracks by dragging them backward or forward along the timeline. You can also drag a track or clip into another track, as long there is enough empty space to hold it.

Multi-Tool This is five tools in one, activated according to mouse position. Get the Selection and Envelope tools by moving the cursor vertically, the Time Shift tool by hovering over the track handles at the beginning or end of the track, and the Zoom tool by moving left into the decibel scale; the zoom view will center over the decibel number you hover over. The Draw tool appears when you zoom in far enough to see individual samples.

Continued on next page.

Table 1-1 (*continued*)

tool has a somewhat misleading name. You might think it's for
cting multiple tracks, but its function is to "link audio and label tracks,"
n is quite different. Link Tracks is enabled by default when you start
w project, and it keeps your audio and label tracks synchronized
n you modify a track. If you don't have a la- bel track, it does nothing.
it when you make changes that affect the track length, such as delet-
art of a track, inserting silence, or changing the tempo. Turn off Link
ks when you copy and paste entire tracks, because it will mess up
paste. Link Tracks appears in Audacity 1.3.9, will be inactive in the
eries, and is scheduled to reappear in the 2.1 series.

Link Tracks

Button Name	Description
Cut	This removes the selection and puts it on the clipboard.
Copy	This copies the selection without removing it and puts it on the clipboard.
Paste	This inserts the clipboard contents at the cursor position or replaces a selection.
Trim	This deletes everything but the selection.
Silence	This replaces the selection with silence.
Redo	Audacity supports nearly unlimited undos and redos, even after saving your project, so it is safe to experiment. The Redo button reverses an undo action or series of undo actions in sequence. You can't skip back to a selected action; you have to redo all of them in order.
Undo	This undoes your last action, or any number of actions before that in sequence, even after saving your project. You can't skip back to a selected action; you have to undo all of them in

order.

Table 1-1 (*continued*)

Button	Name	Description
	Fit Project	This tool sizes your whole project to fit horizontally in your Audacity window. Select **View** > **Fit Vertically** to fit your entire project into the window.

Figure 1-13 shows the Meter toolbar, which displays the recording and playback levels. When it's squished, the Meter toolbar might not display smaller values on its scale. In that case, grab it by the handle on its left side, move it somewhere with more room, then grab it by the resizing handle on the right side, and finally stretch it out until you can see the whole decibel scale.

Figure 1-13: Meter toolbar

Figure 1-14 shows the Mixer toolbar, which is not really a mixer. Instead, it is supposed to control the input and output volume levels on internal sound cards. However, these functions work only if supported by your sound card driver, so if they don't, blame your sound card maker. (For more information on operating system audio controls, see Chapters 13 and 14.)

Figure 1-14: Mixer toolbar

Button	Name	Description
	Zoom In	Magnify. You can zoom in far enough to see individual samples. Click the Selection tool cursor on the point you want to magnify, and the zoom will center on that spot.

Zoom Out Shrink. You can zoom out far enough to see your whole track at once. As with Zoom In, Zoom Out will center on the point that you clicked with the Selection tool.

ɜr when you need to select a small part of a long track

Fit Selection

can enlarge the selection to fit the window horizontally. This tool is

Figure 1-17: Selection toolbar

Managing Audacity Projects

Your first step on a new Audacity project should always be to name it us- ing **File** > **Save As**. Then you can press CTRL-S periodically to save changes or use **File** > **Save**. In addition to the *.aup* file, which is the project's mas- ter metadata file, Audacity creates a directory that holds the associated au- dio files. You can view these in your file manager; there will be many sub- directories full of files with the *.au* extension.

Adding Audio Files: Import vs. Open

Select **File** > **Open** to add an existing audio file to a new, empty project. After that, select **File** > **Import** to add more files. Selecting **File** > **Open** in a nonempty project opens the file in a new window.

The Transcription toolbar (Figure 1-15) changes the speed of playback. For example,

Figure 1-15: Transcription toolbar

you can use it to slow down when transcrib- ing lyrics or to sound sinister and evil. Or you can speed it up for giggles, like Alvin and the Chipmunks. This toolbar only affects Audacity playback and won't change your project file. (**Effect** > **Change Speed** behaves the same way, except it changes your project file. **Effect** > **Change Pitch** changes the pitch higher or lower without changing playback speed, and **Effect** > **Change Tempo** changes the speed without changing pitch.)

Figure 1-16 shows the Device toolbar, where you can select your record- ing and playback devices without selecting **Edit** > **Preferences**. If you plug in or remove a USB device, you need to restart Audacity, or it won't see the change.

Figure 1-16: Device toolbar

Finally, the Selection toolbar, shown in Figure 1-17, offers a number of different scales for precise timing and selection of portions of your audio tracks and for setting the correct frame rates for video soundtracks and com- pact disc audio.

Saving Your Work

Audacity projects are optimized for use as fast workspaces and are not suit- able for archival storage. There's no snapshot mechanism for preserving your work at different stages, and users have reported losing data when projects become corrupted. I use a belt-and-suspenders approach: I make backups of my Audacity project files, and I also make studio master files in WAV format, because each approach has its advantages and weaknesses. First we'll look at a method for saving Audacity projects at different stages, and then we'll look at how to make studio masters in WAV format.

You can create something akin to project snapshots by creating multi- ple Audacity projects from your original project. First, make a directory to hold related projects so they don't get mixed up or lost. Then select **File** > **Save Project As** and give the project a name to help you remember what's in it, such as Summer-Festival-1, Summer-Festival-2, or something more de- scriptive like Summer-Festival-No-Banjos or Summer-Festival-Mondo-Banjos. When you do this, you'll see a dialog like Figure 1-18. The crucial question here is "Copy audio from the following files into your project to make it self-contained?" Say yes by clicking the **Copy All Audio into Project (Safer)** button. This duplicates project files and uses more disk space, but it is the safest option. Sharing files across multiple projects saves disk space, but the headaches aren't worth it because changes in one project affect all projects. Even worse, you lose redundancy, which is your insurance against any one project becoming damaged and unusable.

Figure 1-18: Saving a copy of your project under a new project name

You can control this behavior in the **Edit** > **Preferences** > **Projects** di- alog: When saving a project that depends on other audio files. This offers three choices: “Always copy all audio into project,” “Do not copy any audio,” or “Ask user.”

Figure 1-19: Exporting your project to a 32-bit float WAV file

The resulting file is not a playable WAV file, except in Audacity and other audio editors and digital audio workstations that use 32-bit float for editing. However, it is great for studio masters because you can import and edit 32-bit float WAVs with very little loss of quality and export them to other audio formats: 16- and 24-bit WAV, Ogg Vorbis, MP3, FLAC, and so on. WAV supports a maximum of 32 tracks in a single file.

However, this has its drawbacks too. It works fine when you have only a few tracks to manage—my limit is four—because Audacity does not save the track names but instead renames all of them with the WAV filename. Let's say you have a four-track recording and the tracks are named *vocal*, *piano*, *violin*, and *vocal2*. Export this project to a single WAV file and name it

To make a high-quality studio master WAV file, export your project by selecting **File** > **Export**. You can do this as many times as you want during your work on a project, creating multiple masters to preserve your work at different stages (or until you run out of disk space!). Then you can import a WAV master whenever you want for further editing, and you can export to any other audio format from your WAV master. This also gives you the option of importing your WAV master into another audio-editing program, which you can't do with Audacity's project files.

The default export quality setting for WAVs is 16-bit integer, which is not the highest quality. Audacity's default recording quality setting (select **Edit** > **Preferences** > **Quality**) is a sampling rate of 44.1 kHz and a bit depth of 32-bit float. (Audacity terminology refers to bit depth as *sample format*, but *bit depth* is the correct term.) You can create a high-quality studio master by exporting to 32-bit float WAV. Follow these steps:

1. Select **File** > **Export**. 2. Select Save as type: Other uncompressed files. 3. Click **Options** and then select Header: WAV (Microsoft) and Encoding:
32 bit float.

You will see a window like the one shown in Figure 1-19.

testwav.wav. When you import *testwav.wav* into Audacity, all four tracks are renamed *testwav 1.wav*, *testwav 2.wav*, and so on. It also makes tracks 1 and 2 Right and Left, even if they were originally mono tracks. Figure 1-20 shows the before and after.

You still have all of your individual tracks, but you lose the track names. On multitrack projects, I rely very much on track names to stay organized, so combining them all into a single WAV file doesn't work for me.

For projects that have more than four tracks, I prefer to save each track as a separate WAV file. To do this, select the tracks you want to export and then select **File** > **Export Multi- ple**. (We will discuss selecting tracks in the next section.) Each track will be saved as a separa
track name will become the file- name of the correspo
do this, I put them in their own project directory so the
up with other projects.

Figure 1-20: Exporting your project to WAV (on the left is the original project before exporting to WAV, and on the right is shown what happens when the WAV is imported back into Audacity)

Selecting Tracks and Segments of Tracks

Now let's learn how to select tracks and parts of tracks. Audacity supports the usual editing functions computer users are used to—copy and paste, delete, select, and so on, but it will drive you crazy if you don't learn how to do them the Audacity way. A nice feature of Audacity is that it supports keyboard shortcuts for nearly all functions, so you can use the mouse or keyboard.

First, make yourself a new recording or import an existing audio file by selecting **File** > **Import** so you have some tracks to experiment with. Be sure the Selection tool is active. If you're using the Multi-Tool, move it up or down until it changes into the Selection tool, which looks like a little I-beam.

Track focus and track selection are two different things. A yellow track border shows which track has focus, but if the Track panel is light colored, then that track has not been selected. Having focus means that the track is ready to accept keyboard commands; the cursor line is active in that track, and you can move it back and forth with the arrow keys.

Figure 1-21 shows two tracks: The bottom track has focus, which is in- dicated by a yellow border, and the top one is selected, which is indicated by the shaded Track panel. The cursor line extends into both but is active in the bottom track. Having a selected track without focus and an unselected track with focus isn't much use. You can select a starting point for playback in the track with focus and move it back and forth with the arrow keys, but that's about it.

Figure 1-22: A segment of the top track is selected. Some laptop screens are not bright enough for the shading to be readily apparent in the Track panel, so look at the timeline as well.

Figure 1-21: The top track is selected but does not have focus, and the bottom track has focus, indicated by a yellow border, but is not selected.

When a track is selected, it becomes the target of any editing operations you perform, such as copying, cutting, or applying effects. These will be ap- plied to the whole track, even if it doesn't have focus.

There are two ways to select an entire track: You can double-click anywhere on the waveform, or you can click the track label in the Track panel (see Figure 1-7).

Most times you won't have to pay attention to focus and selection be- cause in the normal course of editing, they'll be where you want them to be. But sometimes things behave oddly, and being aware of this distinction should help you understand what's going on when Audacity seems to be re- sponding mysteriously.

You can also select part of a track. Figure 1-22 shows a track with only a segment, rather than the whole track, selected. Note the difference in shad- ing between the selected and unselected portions.

CTRL-A selects all tracks, and SHIFT-CTRL-A deselects them all. Double- click inside a track to select just that track, or left-click in the track label. SHIFT-click the track label to select and deselect multiple tracks one at a time, as well as nonadjacent tracks. In Figure 1-23, the first and third tracks were selected with SHIFT-clicking.

Figure 1-23: Selecting nonadjacent tracks by holding down the SHIFT key and clicking the track labels

To select part of a track, click and drag with the Selection tool. To make your selection bigger or smaller, move the cursor over either bound- ary of your selection, where it will turn into a horizontal arrow, and then click and drag that boundary (Figure 1-24).

You can navigate between tracks and ad- just selections with your keyboard's arrow keys. Pressing the SHIFT and the left-arrow or right- arrow key enlarges a selection; pressing CTRL- SHIFT and the left-arrow or right-arrow key makes it smaller. A slick trick for making a se- lection across several adjacent tracks is first to make the selection in the top or bottom track and then to press the up-arrow or down-arrow key to repeat the selection in the other tracks.

Figure 1-24: Changing the size of a selection by clicking and dragging a boundary

Figure 1-26: Using the Selection toolbar to select a precise section of an audio track

The Skip to Start and Skip to End buttons in your Control toolbar move the cursor to the start or the end of the track. Pressing the SHIFT key while clicking the **Skip to Start** button selects from the cursor position to the be- ginning of the track, and pressing the SHIFT key while clicking **Skip to End** selects from the cursor position to the end of the track.

You can also use the Selection toolbar to make precise selections based on various track parameters, such as time, samples, and various audio and video frame rates. You can see these parameters by clicking the drop-down menu in any of the three fields on the toolbar (Figure 1-25).

Figure 1-25: The Selection toolbar supports selections based on a number of useful track parameters.

Suppose you want to select a 12-second segment that starts 48 seconds from the beginning of the track. There are several ways to get to the 48- second mark—click with the Selection tool, navigate with the arrow keys, or use the Selection toolbar. Set Selection Start: seconds and enter 48. Se- lect the “End” radio button and enter 60 in the middle box. There is your 12-second segment (Figure 1-26.)

The up-arrow and down-arrow keys also change the numbers, and the right-arrow and left-arrow keys navigate back and forth.

Track Panel

The Track panel puts a number of useful shortcuts at your fingertips (Figure 1-27). The little *X* on the top left deletes the track. The arrow at the bottom collapses and expands the track. You can also grab and drag the track borders with the mouse to change their widths. The Gain slider am- plifies or reduces the track volume without permanently changing it, which is essential when you’re mixing multiple tracks. The Pan slider controls the left-right balance.

By default, Audacity plays all tracks on a project when you click the Play button. Use the Solo button to select one track for playback or the Mute buttons to silence tracks you don’t want to hear. This only affects playback in Audacity and does not change your project files.

The Track menu has an interesting grab bag of func- tions (Figure 1-28). You can use it to create track names— and when you’re working with a lot of tracks, you definitely want to name them. It also offers different waveform views; lets you split or join stereo tracks; lets you set mono, right, or left channel; allows you to move tracks up or down; and lets you change the bit depth (which Audacity calls the *sam- ple format*) and sampling rate.

A stereo track is split into two separate mono tracks with Split Stereo Track or Split Stereo to Mono. With Split Stereo Track, one will be Right, and one will be Left. With Split Stereo to Mono, two mono tracks are created. To create a stereo track, place two mono tracks adjacent to each other and then click **Make Stereo Track** in the Track menu of the upper track. Move tracks by clicking and drag- ging the track label or by selecting **Move TrackUp/Down** in the Track menu.

The vertical scale to the right of the Track panel is your guide to the volume levels of your tracks. The default display is Waveform, and you can change this to Waveform (dB), Spectrum, or Pitch (EAC) with the Track menu. Waveform is a commonly used visual scale for displaying the *amplitude* (the strength or volume of a signal) of your track. The Waveform vertical ruler has a *linear* scale of +1.0 to −1.0; anything that goes over these val- ues represents clipping, which means you’re getting some distortion. Linear means that all frequencies are given equal weight on the scale. Decibels are logarithmic rather than linear, so this is not a true representation, but it is easy to read.

Figure 1-27: The handy Track panel

Figure 1-28: Track menu functions

Waveform (dB) displays amplitude using a *logarithmic* decibel scale. Without diving into the mathematical details, logarithmic means that each 3 dB increment represents a doubling in loudness—hence, a sound mea- sured at 6 dB is twice as loud as a sound measured at 3 dB, and 9 dB is twice as loud as 6 dB. About the smallest increment of change that humans can perceive is 1 dB. (Read more about audio terminology in the glossary.)

There is a special decibel scale for digital audio, the Zero Decibels Full Scale. This represents the digital audio volume range with negative numbers up to a maximum of 0. In Audacity you can control the decibel range dis- played in the Waveform (dB) view and on the Meter Toolbar in the **Edit** > **Preferences** > **Interface** dialog. Click the Meter/Waveform dB Range drop- down menu to see your options. The smallest scale is −36 dB to 0, and the widest scale is −145 dB to 0. This only affects the display and does nothing to your audio tracks.

You can use either waveform display to monitor your recording levels; I think the default waveform display is easiest to read. You'll notice the dis- plays use two shades of blue, one lighter and one darker. The light blue rep- resents the RMS, or *root-mean-square*, which translated into ordinary English is the average volume over time. The darker blue represents the peaks, which are the transient extremes.

NOTE *RMS and peak ratings are (mis)used in the marketing of audio gear to make you think you're getting more than you really are. For example, a set of speakers is rated at 50 watts RMS/150 watts peak. Ignore the peak value—RMS tells you how much the speakers can handle continuously. The peak value indicates what the speakers can tolerate in very short (fraction of a second) bursts.*

The Spectrum view represents the energy level (amplitude) of the dif- ferent frequencies in colors. Red is "hot," or higher amplitude, and blue is "cool," or lower amplitude. If your waveform is mostly blue, it's not very loud, and if it's more red, it's louder. You can easily test this by selecting a track or a segment of a track, selecting **Effect** > **Amplify**, and giving it a neg- ative Amplification value of −30 dB. This should make it mostly blue. Give it a value closer to zero to make it more red.

Pitch (EAC) displays the contours of the pitch of your audio using the enhanced autocorrelation (EAC) algorithm. The EAC algorithm is interest- ing for doing pitch detection; if you are interested in learning more about this, *enhanced autocorrelation* and *pitch detection* are some good Internet search terms to start with. Audacity's implementation of this is pretty basic, so if this interests you, you'll probably want to find more sophisticated tools.

Cutting Out Unwanted Chunks

You can easily remove parts of tracks that you don't want. Just select a sec- tion and then press the DELETE key on your keyboard. If you want to keep only a small part of the track and remove the rest, select **Edit** > **Trim** or click the **Trim** button on the Edit toolbar. This saves the part of the track that you have selected and deletes everything outside of it.

Sometimes you might need to silence a large section of a track while leaving the track intact. In that case, select the part you want to convert to silence, and then click the **Silence** button or select **Edit** > **Silence Audio**.

Fade In and Out

When you delete part of a track, you might want to smooth the cut with grace- ful fades. Fades are integral to audio editing, and Audacity has two ways to create fades. The easiest way is to select a portion of a track and then select **Effect** > **Fade In** or **Fade Out**. You control the length of the fade, and Au- dacity does the rest.

The Envelope tool can fine-tune the amplitude levels; it is good for con- trolling fades and for fine-tuning amplitude anywhere on an audio track, including over relatively long segments. Figure 1-29 shows what this looks like. Click different locations to create control nodes. To get rid of a node, drag it outside the track border.

Figure 1-29: The Envelope tool creates graceful fades and gives you fine-grained control of amplitude.

Each node has four handles. The node handles can be moved in any direction. The outside pair behaves a little differently than the inner pair— use the outside handles to create more graceful, gradual curves.

The dotted lines on either side of the 30-second mark show where the borders of the envelope go outside of the track display.

NOTE *In addition to the Zoom buttons in the Editor toolbar, the View menu has some nice options for manipulating and navigating your tracks such as Fit In Window and Zoom to Selection, and it shows useful keyboard shortcuts such as* CTRL-*2 for Zoom Normal and* SHIFT-CTRL-*F for Fit Vertically.*

Figure 1-30: Using **Effect** > **Amplify** to raise or lower the volume of your selection

Another way to amplify a too-quiet recording is to select **Effect** > **Nor- malize**. Check “Remove any DC offset,” check “Normalize maximum ampli- tude to,” and set a maximum level up to zero. *DC offset* refers to the mean amplitude; if this is not zero, then normalization won’t be applied correctly because the amplitude levels will be unbalanced, and it might create some distortion.

The difference between Amplify and Normalize is seen when they are applied to multiple tracks. Amplify changes the volume on all tracks by an equal amount. If you amplify volume by +9 dB, a track that peaks at −20 dB will be raised to −11 dB, and a track that peaks at −9 dB will be raised to 0 dB. Normalize, on the other hand, adjusts all tracks to the same maximum volume level, so some tracks may be changed more than others.

The default maximum setting for both is zero. It’s useful to lower this to −12 dB or so on your studio masters in order to leave a bit of headroom for more tweaking without risking clipping. For example, when you downmix multiple tracks into a single track, the latter will have the combined ampli- tudes of all those tracks and get louder, maybe a lot louder. Experience will tell you how much headroom you need. Don’t normalize to zero until you are making your final exports.

Amplify and Normalize can also be used to lower amplitude. In the Am- plify dialog, enter a negative value, like −6. The Normalize dialog uses only negative values and won’t allow anything higher than zero.

Making Quiet Recordings Louder

Suppose your recording is too quiet and you want to amp up the volume. No problem! Select the part you want to amplify and then select **Effect** > **Amplify**. Audacity automatically calculates how much amplification can be applied without clipping; that is, without going over 0 dB (Figure 1-30). Don’t check the “Allow clipping” box unless you are very sure you want to do so.

Timer Record and Sound Activated Recording

Both Timer Record and Sound Activated Recording are in the Transport menu. To use Sound Activated Recording, select **Transport** > **Sound Activa- tion Level** and set the decibel level you want to trigger recording. It may take a bit of trial and error to figure out a level that balances capturing what you want without also capturing lot of sounds you don’t want. Then turn on the recording monitor (Meter toolbar) and click the **Play** button. When a loud enough sound is detected, Audacity will automatically create a new track and then use that track for as long as you leave Timer Record activated. Click the **Stop** button any time to stop Sound Activated Recording.

Timer Record is just as easy—just set the start and stop times for record- ing. You can use this together with Sound Activated Recording to set a start and stop range so that you can go away and leave Audacity running without worrying about it filling your hard drive.

Mixer Board

The Mixer Board is a new feature that first appeared in Audacity 1.3.8 (Figure 1-31).

Figure 1-31: The Mixer Board puts Pan and Gain controls front and center.

Figure 1-32: Using **File** > **Open Metadata Editor** to store useful information in your Audacity project

If you select **Edit** > **Preferences** > **Import/Export**, there is a "Show Metadata Editor prior to export step" option. If you check this, the meta- data editor will open for each track before it is exported so you can review or edit the metadata.

This is a slick little board with volume units (VU) meters for each track, plus Pan and Gain sliders. It puts your essential mixing controls within easy reach, without having to make your tracks extra wide so that you can access the sliders on the Track panels.

To use the Mixer Board, play your tracks in Audacity and adjust the relative volumes of your tracks with the Gain sliders, which are on the left side of the VU meter, and the left-right balance of each track with the Pan slider. Then make your export. The Pan and Gain sliders do not change your project files—they only affect playback in Audacity and how your ex- ported file will sound. See Chapter 9 to learn more about multitrack mixing.

Track Metadata

You can preserve useful data in your Audacity projects, such as song titles, date, artist name, and genre, with the metadata editor. Before your final ex- port, select **File** > **Open Metadata Editor**. You'll see a window like the one in Figure 1-32. Fill in any of the Artist Name, Album Title, Year, Genre, and Comments fields, and these will be applied to each song track. Audacity will fill in the Track Titles and Track Numbers fields automatically.

Final Mixdown

Usually your goal is to mixdown however many tracks you have recorded to a stereo track. However, Audacity also supports multichannel surround sound, which is covered in Chapter 9. Before you export, select **Edit** > **Preferences** > **Import/Export** and select the "Use custom mix" radio button. At export, an Advanced Mixing Option window appears, which is a simple channel mapper. Map your tracks to whichever channels you want. Your tracks can go to the left channel, the right channel, or even multiple channels. Chan- nel 1 is always the left channel when there are two tracks. (See Figure 1-33 for a simple two-track example.) When you are using this tool, you will be glad you named your tracks. Chapter 9 goes into more detail on multitrack mixing and channel mapping.

Figure 1-33: Using Audacity's mixer to map your tracks to the correct channels

Audio File Formats and Quality Settings

There are many different audio file formats, and Audacity supports a lot of them. Let's look at WAV, MP3, FLAC, and Ogg Vorbis. These well-supported, popular formats serve different purposes.

Understanding File Formats WAV files are uncompressed, high-quality pulse-code modulation (PCM) files. They are large. One minute of a CD-quality stereo WAV recording consumes about 10MB of disk space. WAV is the best-supported format and is the quality standard by which other formats are measured.

MP3 (MPEG-1 Audio Layer 3, not MPEG-3) is a popular compressed, lossy encoding format; an MP3 file can be as small as one-tenth the size of a similar WAV file and still sound pretty good. This means you can cram a lot more music into a portable player and have faster downloads and better online streaming. The trade-off is that quality is lost. See Chapters 13 and 14 to learn how to enable MP3 support in Audacity.

Although MP3 is extremely popular, it is encumbered with messy pat- ent problems. A number of different companies in different countries claim they own patents on MP3, and depending on where you live, you may be ex- pected to pay licensing fees if you want to distribute music encoded in MP3. The final patents won't expire until 2017. However, the patent situation is not clear-cut because many independent musicians distribute their music in MP3 format without paying patent royalties, and patents do not apply in all countries outside their countries of origin.

Free Lossless Audio Codec (FLAC) is an excellent open and free format. This lossless, compressed format is equivalent in quality to WAV but with file sizes that are one-third to a half smaller. FLAC is a great format for a PC media server, because you get great quality without eating up as much hard drive space. Online music services distribute their highest-quality downloads in FLAC. You can even use FLAC for your studio masters if you need to con- serve storage space. Although the FLAC format does not support 32-bit float, 24-bit FLAC files are still very high quality.

Ogg Vorbis was created as a high-quality, free, and open alternative to MP3. Ogg files range from about the same size as MP3 to about 25 percent larger. Ogg Vorbis is not as widely supported as MP3 and WAV, though its popularity is increasing. Linux, Windows, and Mac all have a number of software music players that support playing both standalone Ogg files and streaming Ogg. The iPod and Zune do not support Ogg (not a big surprise, coming from the two titans of lock-in), but a growing number of other play- back devices do.

Chapter 7 goes into detail about different quality levels for Ogg Vorbis and MP3, and Chapter 6 discusses WAV and FLAC.

The next section explains some important fundamental concepts of dig- ital audio and terminology that you'll encounter a lot, so grab a cup of tea, put your feet up, and read on.

Understanding Bit Depth and Sampling Rate Digital audio production can be summed up as converting analog signals to digital and back again. In other words, you capture sound from an ana- log microphone or electric musical instrument, run it through an analog- to-digital converter (ADC), and record the digitized bits to a hard drive or solid-state storage. The ADC can be a sound card, a preamp/ADC, a stand- alone ADC, or some other combination device. Somewhere down the road this digital data will be retrieved and converted to analog form for playback.

Your computer's sound card performs digital-to-analog conversion during playback, and so does an ordinary CD or MP3 player.

Your goal is to convert those analog signals as faithfully as possible. Once they are in digital form, you have a whole world of tools at your disposal to manipulate them in all kinds of creative ways, and you have a multitude of options for playback formats and media.

16/44.1, 24/96, 32-Bit Float Two common digital audio specifications are referred to as 16/44.1 and 24/96. Sometimes 16/44.1 is shortened to 16/44. These designations specify *bit depth* and *sampling rate*. Bit depth affects dynamic range, signal-to-noise ratio, and fidelity. Sampling rate determines the frequency range.

CD-quality audio is defined as 44.1 kHz, 16-bit, two-channel WAV, and 24/96 is higher-than-CD audio quality, such as digital audio tape (DAT), DVD audio, and studio master recordings. So, we should just go for the highest numbers to get the best quality, right? Well, no—there are a num- ber of factors to consider.

Sampling is done by the analog-to-digital con- verter; it samples the electric voltage in an analog audio signal at intervals and converts the measure- ments to digital form. The more times per second this is done, the more accurate the digital rep- resentation of the signal. So, a sampling rate of 44.1kHz means 44,100 samples per second per channel. This is Audacity's default. You can see a picture of this by enlarging a section of any Au- dacity waveform. This will look something like Fig- ure 1-34, where each dot represents a single audio sample.

Each audio sample is represented as a numeric value—in computers, everything is a number. In CD-quality audio, 16 bits is the range of possible values per sample, and 16 bits = 65,536. This is the *bit depth*. Each sample is not 65,536 bits in size but is given a single 16-bit value that is equal to or lesser than 65,535 (0 to 65,535.)

For 24/96 recording, which is often used in professional recording, 24 bits gives 16,777,216 possible values. Larger bit depths mean wider dynamic ranges and finer tonal shadings—and also significantly larger file sizes. One minute of a stereo recording is about 10MB (5MB per channel) at 16/44.1 and is about 34MB at 24/96.

Theoretically, the dynamic range of 16-bit digital audio is 96 dB, for a scale of −96 to 0. For 24-bit audio, it is 144 dB, and for 32-bit audio, it is 192 dB. In the real world, the actual dynamic range is lower because of limita- tions in electronic hardware: around 90 dB for 16-bit audio and 115 dB for both 24-bit and 32-bit audio.

Figure 1-34: Audacity waveform, enlarged so that the sample points are visible

The value of a wide dynamic range is not that you can shock your listen- ers with sudden extreme volume changes but in having a very high signal-to- noise ratio, which is also called "having a low noise floor." The more signal and less noise, the better.

As far as adjusting the dynamic range with the listener in mind, a range of 50 dB to 60 dB in a recording is the maximum most listeners will toler- ate, and that is under ideal conditions with a good system in a quiet environ- ment. A picky audiophile with good equipment and a quiet listening space will enjoy a symphony that uses every bit of a 60 dB dynamic range. A live symphony concert might encompass an 80 dB range.

Someone listening to music in a noisier environment or on lower-quality audio gear might be more comfortable with a dynamic range of 20 dB or even narrower. Audacity lets you tailor your recordings to any dynamic range you want. (See Chapters 6, 8, and 11 to learn more about dynamic range compression.)

There is a famous theorem in the audio world called the Nyquist- Shannon theorem. It is long and detailed—the part that matters here states that a perfect digital representation of an analog audio signal is possible when the sampling rate is at least twice as high as the highest frequency in the signal. The best human hearing can hear up to 20 kHz to 24 kHz, so a sampling rate of 40 kHz to 48 kHz can (theoretically) reproduce the entire range of human hearing.

Audacity's default recording bit depth is 32-bit float. Many digital au- dio workstations, including Audacity, operate at 32-bit float internally. It is important to understand that this is 32-bit *float*, not 32-bit integer. In con- trast, the 16- and 24-bit depths represent integer values. As usual, the math is complex, and audio geeks will bore you to tears telling you about it if you let them, so here is the oversimplified short story: Integers are whole num- bers, and *float* means floating decimal point. This 32-bit float number is a 24-bit mantissa plus an 8-bit exponent. This is significant in terms of audio production—it gives you a dynamic range of about 1500 dB, which means virtually no noise or clipping, and you get a smoother, more accurate re- sponse curve across the whole range of your analog-to-digital conversion. (In comparison, the dynamic range of 32-bit integer is 196 dB.)

Recording and editing at 32-bit float is beneficial even with a 16-bit re- cording interface. If you work to a peak of −24 dB, which is very low and safe to avoid clipping, your available dynamic range will still exceed what any hardware supports, and you'll have plenty of extra bits to throw away without harming quality. That means you'll be able to edit and manipulate your audio files as much as you want and still be able to make high-quality 16-bit and 24-bit exports. The more processing you apply to your recordings, the more you want all that extra headroom.

You will always export to a lower bit depth because there is no such thing as a 32-bit float playback device. Playable formats must be integer data, so your exports will always be to 8-bit, 16-bit, or 24-bit integer. (The word *linear* is often used instead of integer; they're referring to the same thing.)

In practice, many factors will affect the bit depth and sampling rate that you choose: How good is your hearing, how good is your gear, and what are the final format and playback medium going to be? How good are your recording techniques? What sort of recordings are you making—nuanced acoustic instruments and vocals or head-banging rock? Is your computer powerful enough to process larger audio files, and do you have enough stor- age for them?

You can get some good bargains on 16/44.1 recorders and ADCs be- cause of the "bigger is better" mentality. On the other hand, it doesn't hurt to have the extra bandwidth—you can always reduce quality and size, but (despite what they show you on TV crime shows) you can't recover what wasn't available in the first place. When you're experimenting, my recom- mendation is to try increasing the bit depth before increasing the sampling rate. If your hearing and audio gear are good, you should be able to hear the difference between 16-bit and 24-bit recordings, though I suspect it will take a side-by-side comparison to make the differences apparent. I record and save my studio masters at 32-bit float/48 kHz WAV and export mainly to CD-quality 16/44.1. Going to higher sampling rates makes no difference that I can hear, and it eats up hard drive space.

Audacity supports recording at 16-bit integer, 24-bit integer, and 32-bit float. For comparison, professionals record and edit at 24-bit, 32-bit, and even 64-bit depths.

NOTE *As we discussed earlier, exporting to 32-bit float WAV is a good option to consider for creating and archiving studio master files. Then you can import your 32-bit float WAV back into Audacity (or any other audio editor that uses 32-bit float), process it, and export to 16- or 24-bit without loss of quality.*

Bitrate, Bit Depth, and File Size *Bit depth* is a term with a specific meaning, which we just learned. Another common term is *bitrate*. These two terms mean different things and are often confused with each other. Bitrate is the amount of data per second needed to transmit an audio file and is most commonly expressed as Kbps or Mbps; 16/44.1 stereo is roughly 1.4Mbps, and 24/96 is about 4.6Mbps. You can easily figure this out for yourself:

bit depth × sampling rate × channels = bitrate (bit/sec) 16 × 44,100 × 2 = 1,411,200 bits/sec 24 × 96,000 × 2 = 4,608,000 bits/sec

MP3s (and other lossy formats) are described in terms of bitrates rather than bit depth/sampling rates. The MPEG-1 Layer 3 standard spec- ifies a range of bitrates from 32Kbps to 320Kbps. Pretty lo-fi, 128Kbps is a common MP3 bitrate left over from the early days of slower Internet down- load speeds and players with small storage capacities. Now 192Kbps is pretty common, and a person with good hearing and a decent MP3 player will hear the difference.

Total file size is an important figure to ace recording geeks. Digital au- dio files are large, and you can eat up a large hard drive in no time during a busy recording session with a lot of takes.

You can calculate the approximate file size with this equation:

bit depth × sampling rate × channels × (60 seconds) / 8 = file size in bytes for a 1-minute recording

One stereo minute at 24/48 kHz is about 17.3MB:

24 × 48,000 × 2 × 60 / 8 = 17,280,000 bytes

You must divide by 8 to get bytes because there are 8 bits per byte.

Now What?

At this point, you should have a good grasp of the basics of Audacity. Audac- ity is easy to learn; the hard part is learning audio concepts and terminology. I'll be talking about those a lot in the rest of this book, translating them into practical terms and showing you how to implement them in Audacity.

There is a saying in the photography world: The person behind the camera matters more than the camera. The same applies to mak- ing great audio recordings—the person behind the gear matters more than the gear. This doesn't mean that the gear doesn't matter, because it does. But merely owning the most expensive, elite audio equipment won't turn you into Tom Dowd or Rick Rubin or Quincy Jones or George Martin or whoever your favorite legendary music producer is. It won't turn average musicians into stars. When you're shopping for audio gear and getting drawn into the "higher specs and price tags is better!" zone, take a step back and reboot your mind. Take a deep breath, slow down, and concentrate on learning how to get the best out of lower-end equipment. Because today's average digital audio gear is better than the top-of-the-line analog studio gear of yesteryear, with more

2

BUILDING A GOOD DIGITAL SOUND STUDIO ON THE CHEAP

accuracy, fidelity, wider dynamic range, and less noise, it's a whole lot eas- ier to record, mix, edit, and apply special effects. You won't have to fuss so much with learning how to use your equipment, and you can concentrate more on learning how to make good recordings with artistic and technical fidelity. Then if you find yourself yearning for a better preamp, better mi- crophones, better speakers, or what-have-you, you'll have good reasons, and you'll know and appreciate better quality when you find it.

In this chapter we're going to turn a PC into a digital audio workstation, and we won't break the bank to do it. We'll also look at portable digital re- corders, which cram amazing fidelity and storage capacity into tiny devices and are endlessly useful in all kinds of circumstances.

Getting Sound In and Out of the Computer

One of the biggest hurdles with PC-based audio production is figuring out audio gear. There is a huge and bewildering selection of audio components with every imaginable combination of features and price tags. In this chapter I'll discuss the basic elements of a small recording studio. Appendix A goes into more detail on hardware, with examples of different models and brands in different price ranges.

There are several ways to interface audio gear with a computer: a PCI or PCI-E sound card for a desktop computer; a Cardbus or ExpressCard sound card for laptops; USB 1.1, USB 2.0, or FireWire audio interface for any com- puter. You will need one of these. Which one? For one- or two-track record- ing, any of them will do. For heavy-duty multitrack recording, USB 1.1 is not a candidate. But it's great for one- and two-track recording, and you'll have a lot of excellent, moderately priced devices to choose from. For serious multi- track recording, you need the faster protocols, especially if you're going to record at high bit depths and sampling rates, so we'll look at their strengths and weaknesses.

The most essential component in digital audio production is the analog- to-digital/digital-to-analog converter. The ADC/DAC is how you get analog audio in and out of your computer. It takes the analog signal from a micro- phone or instrument and converts it to a digital signal. Then it converts a digital signal to analog for playback. ADC/DACs come in a multitude of forms: The most low-end, cheapskate onboard sound chip has one, and of course so do higher-end audio interfaces. There are all kinds of USB and FireWire recording interfaces that connect microphones and instruments to your computer, and with them you don’t need to bother with an internal sound card at all. There are also slick little portable ADC/DACs for connect- ing turntables, cassette recorders, and stereo hi-fi amplifiers to a computer, as well as USB microphones and turntables with built-in USB ADC/DACs. And for the studio geek with a healthy budget, there are the more expensive rack-mount ADC/DACs.

Figure 2-1: My own little recording empire (the mics don’t really sit right there in front of the amp and mixer; that’s a pose just for the photo op)

Also not shown is a nice Focusrite Saffire Pro 26 I/O FireWire multi- channel recording interface. Focusrite makes great audio hardware and sup- ports Linux, Mac, and Windows. We’ll see more of the Saffire in Chapter 9.

This is how they all fit together:

- Recording: microphones and instruments > Behringer mixer > MobilePre > computer
- Playback: computer > MobilePre > Pioneer stereo amp > speakers

For recording, the Behringer mixer has a pair of RCA recording outputs. These send a stereo signal via an RCA-pair-to-two-1/4 mono TRS adapter to the two 1/4 TRS jacks on the MobilePre. The MobilePre connects with a USB cable to the computer. Everything plugged into the Behringer is then captured in Audacity.

NOTE *The terminology for connectors is a bit mixed up, so I am going to refer to the connectors on cables as* plugs *and the sockets on mixers, preamps, amps, and so forth that they plug into as* jacks*. I will also bow to convention and refer to* male *and* female *connectors even though that has always sounded weird to me.* Male *and* female *at least have the virtue of having precise meanings, unlike a lot of audio terminology.*

An Example Studio Let's start off with a photo of a basic, moderate-quality computer recording studio, which just happens to be mine all mine. Figure 2-1 shows the whole works: The computer is underneath the table. Starting from the left are various headphones; then on the desk is an external USB CD/DVD writer, a four-port powered USB hub, an LCD monitor, a color printer, the all- important hot beverage mug, a turntable, an excellent old Pioneer stereo hi-fi amplifier, a Behringer powered mixer, and a MobilePre USB preamp/ analog-to-digital converter. In front are two dynamic microphones. Not shown are a pair of nice JBL speakers mounted up on the wall. You can cram a lot of functionality into a small space.

Figure 2-2: This is the back of a treasured old but versatile Pioneer stereo amplifier. It can connect two turntables (it even has proper ground connectors for turntables), two tape decks, a microphone, a tuner, another amp, a preamp, and an Aux connector for a CD player or other input devices.

I could also cut out the Behringer entirely, for example for interviews and podcasts, and use only the MobilePre in combination with any computer.

When I'm copying vinyl LPs to CD, it goes like this:

- Recording: turntable > Pioneer amp > MobilePre > computer
- Playback: computer > MobilePre > Pioneer amp > speakers

The turntable plugs into the phono ports on the amp. The amp con- nects to the MobilePre from a pair of recording outputs. I could use an RCA-pair-to-two-1/4 TRS adapter or an RCA-pair-to-stereo-mini-plug, be- cause the MobilePre is flexible when it comes to making connections.

You can digitize any legacy media with this kind of setup, because what- ever connects to your hi-fi amp or receiver can be copied into your computer.

Both recording and playback are routed through the MobilePre, so I get to hear the playback on good speakers instead of lo-fi computer speakers. The MobilePre has a headphone port for zero-latency monitoring during recording, which is a nice thing to have. The Behringer mixer (Europower 1280S) isn't really intended to be a studio mixer; it's for powering live shows,

For playback, the MobilePre has a 1/8 stereo output. This connects with a stereo-mini-plug-to-RCA-pair adapter to the Aux input on the Pio- neer amp. The MobilePre also has a pair of 1/4 TRS outputs, so for these I would need 1/4 TRS-to-RCA adapters. Figure 2-2 shows the plug-ins on the back of the amp. A nice amp like this plus a good ADC/DAC makes a great hub for a conversion studio, because anything that connects to the amp can be recorded on your computer.
which it does most ably, because it is an integrated mixer and 1200-watt amplifier. When I'm recording a live show, I hook up to it with a laptop and the MobilePre. I also have a Zoom H2 portable digital recorder, which I can use in place of the laptop plus MobilePre. The best-quality recordings at live shows come from plugging directly into the mixer board.

NOTE *The local old-time country band that I like to listen to and record has a rather eccen- tric sound system. They have a nice PA system, but instead of plugging everyone into the mixer board, only the singers' mics are connected to the mixer. All of the musicians have to bring their own instrument amplifiers. This makes for a cluttered stage, and recording is a nightmare—plugging a recorder into the mixer means the only instru- ments it hears are whatever the mics pick up. The Zoom H2 has a neat little adapter to mount it on a mic stand so I can position it anywhere, but it's not as good as a proper setup with everything routed through the sound board.*

Let's take a closer look at the MobilePre because it is representative of a lot of USB recording interfaces. The MobilePre supplies 48v phantom power for condenser mics and has XLR and TRS jacks, a 1/8 stereo in- put, two 1/4 mono outputs, and a 1/8 zero-latency headphone port for monitoring. Both dynamic and condenser mics can plug into the XLR jacks, as long as they have XLR connectors. You could also use an XLR-to-TRS adapter to plug a dynamic mic into one of the 1/4 inputs. Its built-in ADC supports sampling rates from 8 to 48 kHz at 16 bits, and it draws its power from the USB bus of your computer, so it doesn't need its own power cord. It has physical gain control knobs and a volume control knob for headphones, so you don't need to dink around with software controls. (I'd rather twist a knob than fumble around some weirdo software interface any day.) You should be able to find one for under $150. At 16-bit/48 MHz maximum recording quality, it's becoming obsolete because comparable devices sup- port 24-bit recording. Still, it's a great little device, and because it is USB 1.1 class-compliant, it runs on any computer without special drivers.

Figures 2-3 and 2-4 show the front and back of the MobilePre.

Figure 2-3: M-Audio MobilePre, front. From left to right: Channel 1 1/4 mono TRS jack, Ch. 1 and 2 gain controls, clip LEDs, headphone port, headphone volume knob, phantom power switch, phantom power LED, and power LED le XLR plug.

Figure 2-5: One stereo 1/4 TRS plug (left) and one mono (right)

Figure 2-6: Here is a three-pin male XLR plug and a three-pin

Figure 2-7 shows a collection of plugs and adapters. You can find adapters to make anything fit anything. However, you must be careful—just because something fits doesn't mean it belongs there, so read your product manuals. Stereo TRS plugs have two black bands near the tip, and mono TRS plugs have one.

Figure 2-4: M-Audio MobilePre, backside. From left to right: USB jack, stereo line out, 1/4 mono TRS right and left outputs, stereo mic in, Ch. 2 1/4 mono TRS input, Ch. 2 XLR mic jack, Ch. 1 XLR mic jack

Sorting Out Connectors Where do the terms TRS and XLR come from? TRS is *tip-ring-sleeve*, which is the physical description of a TRS plug. Figure 2-5 is a labeled photo of a stereo and a mono TRS jack.

The origins of XLR are a little more complicated. Cannon Electric was the original manufacturer of the XLR connector, and some old-timers still call it a *cannon plug*. It started out as the "Cannon X" series of connectors. Then later versions added a latch, so there is the *L*, and then the contacts were encased in rubber, for the *R*. Figure 2-6 shows a pair of three-pin XLR plugs.

Figure 2-8 shows the three-pin male XLR connec- tor on a Behringer dynamic mic.

Multichannel Recording, PCI, USB, FireWire

Multichannel recording is done in several different ways. One way is to use a simple two-track recording interface like the MobilePre. It supports up to six inputs at once and routes them into two channels. There are no mixer controls, so this requires some finicking during recording to get a decent balance. A better way to do multichannel recording with a two- channel interface like the Pre is to record two tracks at a time, giving each instrument or performer its own individual track, rather than trying to cram them all through the Pre at once. Then Audacity is your mixer, and you have individual control of each track.

Another option for two-channel recording is a good-quality two- channel PCI sound card like the Emu 1616M PCI, which you can find used for under $200. It comes with a breakout box that supports all kinds of plug- ins, 24-bit/192 kHz recording, phantom power, and preamps.

My Behringer 1280S gets pressed into service when I have a larger group over for recording. Any analog mixer will work as long you have an ADC/ DAC to plug it into. My setup is a bit of a hack job since the Behringer is not really a studio mixer, but it works and sounds good, and it is an example of how a little ingenuity goes far in the wild world of audio. Like many mixers, the Behringer outputs to two-channel stereo, so I need to get the mix right during recording—I'll have only two channels to work with in Audacity. I

Figure 2-7: Two stereo RCA to 1/8 TRS plug, 1/4 mono TRS to two stereo RCA, 1/8 to 1/4 stereo TRS adapter, 1/4 to 1/8 mono TRS adapter

Figure 2-8: Behringer dynamic mic showing off its three-pin male XLR connector

could also record one or two tracks at a time with the Behringer and then knit them together in Audacity; there is no rule that says you have to plug in everything at one time. (Though corralling and organizing musicians can be a bit of a cat-herding experience, and sometimes you have to take what you can get.)

Audacity from version 1.3.8 supports recording as many tracks at once as your recording interface supports. Older versions max out at 16. This is where FireWire and the higher-end PCI sound cards shine, because they allow recording many tracks at once. The Focusrite Saffire Pro 40 is an ex- ample of a good value in a FireWire recording interface at about $500; this gives you 8 mic preamps, 20 total inputs and 20 outputs, 24/96 recording, blinky LEDs, and phantom power on every mic channel. The M-Audio Delta 1010 is a popular higher-end multichannel PCI sound card that connects to a rack-mount breakout box. It goes for about $600.

A cool new family of devices is USB and FireWire mixers. These give you everything in one device—preamps, phantom power, mixer board, ADC/ DAC, and direct plug-in to your computer. There are a lot of nice choices in the $300 to $1,000 range. Behringer's line of Xenyx USB mixers costs be- tween $150 to $600. They use class-compliant USB 1.1, so they plug into any computer with no special drivers needed. The M-Audio NRV10 is a nice little FireWire mixer/preamp that costs about $700.

How to choose which of these to use? USB and FireWire are portable and easy to hook up. PCI Express is the fastest. One lane of PCI-E moves about 250MBps both ways at the same time. That's 250 megabytes, not mega- bits. Plain old PCI maxes out at 133MBps. Also, unlike PCI-E, PCI uses a shared bus, so more PCI devices means more bandwidth contention. Every PCI-E device has its own dedicated data pipeline, so PCI-E devices don't have to share bandwidth. USB 1.1 is rated at

12Mbps (megabits per second), and USB 2.0 is rated at about 480Mbps, but both figures are highly theoretical, and in real life you're likely to get half that. FireWire is rated at 400Mbps. However, FireWire gives you higher sustained throughput and better perfor- mance than USB, a difference that is discussed in more detail in the follow- ing section.

A common problem with internal sound cards is picking up noise and electrical interference from hard drives, power supplies, and fans inside your computer case. This usually isn't a problem with the better sound cards like Emu, M-Audio, and RME Hammerfall, but it tends to be more of an issue with consumer-level and gamer sound cards and low-budget onboard sound. If you are getting some noise, the first thing to check is all your connections— make sure everything is hooked up correctly and anything that needs to be grounded is grounded. Sometimes moving a PCI card to a different slot makes a difference. Check your motherboard manual to see if you have shared PCI slots; you don't want to use a shared slot if the other slot is populated.

USB or FireWire?

If you like the convenience of a USB audio interface, you might also con- sider FireWire devices. How do you choose between FireWire or USB? USB devices usually cost less than FireWire, but the trade-off is you may get poorer performance because of the differences in the two protocols.

All FireWire interfaces have special controller chips, so they do not add any extra load to your computer's CPU. FireWire is a peer protocol, which means FireWire devices negotiate bus conflicts without using host CPU cy- cles. FireWire gives you two operating modes to choose from: asynchronous or isynchronous. Isynchronous mode means a device can reserve a certain portion of bandwidth all for itself that no other devices can use. So there are no collisions, which translates into high sustained throughput.

If your PC doesn't have a FireWire interface, it's easy to add one. PCI FireWire interfaces cost about $50, and many laptops include a FireWire port. When you're shopping for FireWire audio interfaces, be sure to check for hardware compatibility. As one example, the Presonus FP10 has known conflicts with certain video chipsets, and it has a limited set of FireWire inter- faces that it is known to work well with.

The Future of FireWire

"FireWire is doomed!" is a common cry of late. That may be true, though it is going to be with us for some years yet. USB 2.0 sup- posedly offers the potential to equal FireWire performance, and USB 3.0 supposedly will eclipse it. Audio hardware manufacturers have been slow to release USB 2.0 recording interfaces, though there are now a respectable number of them. A lot of them rely on custom drivers that are not USB 2.0 compliant, so shop care- fully lest you buy one that won't run on your computer. USB 3.0 is still a work in progress, and audio hardware manufacturers are not known for moving quickly. If you like FireWire recording devices, by all means buy and use them, and if FireWire ever does become obsolete, you can still use your gear because nobody is going to come and take it away from you.

USB operates only in asynchronous mode. *Asynchronous* means that any device on the same bus can send data whenever it wants to, so sometimes there are collisions, which cause latency. USB is host-dependent and puts a load on the CPU, which can also cause latency. Latency is the enemy of quality audio.

You'll see a lot of USB audio devices that still use USB 1.1. USB 1.1 has two speeds: 1.5Mbps and 12Mbps. The latter is also called *full-speed*. It's un- likely that a USB recording interface will be geared down to 1.5Mbps. The number of channels you can record at once depends on the quality level

m power when it is delivered through the microphone ca-
cause there isn't a separate power cable. They are more
than dy- namic mics. Condenser mics live mostly in
. They are also used onstage in combination with
ic mics on drum kits; the condenser mics hang overhead
ure cymbals and tran- sients, and dynamic mics are
next to the drums. Figure 2-9 shows an Audio-Technica
condenser mic, a Behringer dynamic mic, and a wind
Experienced singers know to avoid brightly colored wind
e- cause they look like clown noses.)

Figure 2-9: Audio-Technica stereo condenser mic, Behringer dynamic mic, and wind sock

Condenser mics require power. This is called

you want to record at. CD quality, two channels at 16/44.1, has a bitrate of 1,411,200Mbps. Two channels at 24/96 equals a bitrate of 4,608,000Mbps, so it seems you could record four 24/96 channels at once. However, that 12Mbps maximum is theoretical, and your real-world throughput will be half that or less. Most likely you'll be limited to two-channel 24/96 record- ing at best. Four channels at 16/44.1 or 24/48 are possible if you are careful and have a good, fast multicore PC and have it tuned for audio production. (See "Bitrate, Bit Depth, and File Size" on page 29 to learn about different bitrates.)

USB 2.0 audio devices require careful shopping, because many of them are not USB class-compliant and instead supply their own special drivers. Even Windows users have to do their homework because vendors are slow to release drivers for new Windows releases. Mac support is decent overall, and

Linux, Unix, and users on other platforms are at the back of the bus as usual. Some multitrack USB 2.0 devices are getting good reviews. For exam- ple, the M-Audio Fast Track Ultra 8R (eight in, eight out) gets high marks and works on Mac, Linux, and Windows.

Microphones

The microphones in Figure 2-9 are middle-of-the-road dynamic mics that cost less than $100 each. The microphone is very important—you won't get good recordings from low-quality microphones. There are two common types of microphones: condenser and dynamic. *Condenser* mics have a wider frequency response, are more sensitive, produce louder output, and have a faster transient response. *Transient response* is any abrupt change, such as a rim shot, a hard-strummed guitar, or a singer hitting some hard consonants (and probably spraying a bit of spit).

Condenser microphones that require phantom power typically use XLR connectors. Dynamic mics use both XLR connectors and TRS plugs. Dy- namic mics do not require phantom power, so make sure that the phantom power is turned off before you plug one in to a phantom-powered XLR jack. It shouldn't damage the mic, but it will change how it sounds. It's common to see audio gear with both types of microphone connectors, and newer de- vices have combination jacks that accept both.

The little Audio-Technica Pro 24 stereo condenser mic in Figure 2-9 is a different kind of condenser mic. It is self-powered by a little mercury battery, it has a built-in cable, and it plugs into any 1/8 TRS stereo microphone jack, like on laptops, digital recorders, and camcorders.

There are two types of condenser mics: large diaphragm (LDM) and small diaphragm (SDM). Both record sounds evenly and accurately across their entire range, though LDMs have a reputation for creating a "warmer" sound. A large diaphragm mic has a better low-frequency range than a small diaphragm, but a small diaphragm mic of the same type has a better high- frequency response. Low tones are characterized as warmer, and high tones are characterized as cooler and brighter. You're going to hear all kinds of characterizations for sound quality: warm, cold, brittle, soft, hard, bright, dull, and on and on. Trust your own perceptions, and don't worry about what other people tell you you should like.

Dynamic mics have a narrower frequency re- sponse and are less accurate than condenser mics. They are rugged, are moisture-resistant, and don't need a power supply, so dynamic mics go on stage and in the field. Dynamic mics generally cover the human vocal range plus a little bit, so this makes them good for singers.

Another type of microphone is worth consider- ation, and that is the ribbon mic, which is a type of dynamic mic. The guts of ribbon mic are a metal rib- bon suspended in a magnetic field. These are expen- sive but are prized for their clarity, spatial depth, and realism. They revolutionized the audio industries back in the 1930s; ribbon microphones set new stan- dards for realism and accuracy that the condenser mics of the day could not match. Ribbon mics fell out of favor somewhat as condenser and dynamic mics improved. They

were expensive, the metal ribbon was fragile, and thei
they needed more amplification than other types of mi
ribbon mics are more affordable and durable and prod
their ancestors, so they're definitely worth trying (Figur

Ribbon mics are natively bidirectional, which means the
sounds from both the front and the rear of the mic in a

Figure 2-10: A modern AEA R84 ribbon mic

Figure 2-11: Some common microphone polar patterns (Image Credit: Created by Wikipedia user Galak76, released under the GFDL.)

These are the common polar patterns:

Cardioid Picks up sound from the front and rejects sounds from the rear. Sub-cardioid is rather like omnidirectional, with a smaller range to the rear. Hyper-cardioid and super-cardioid have a narrower range in front, plus small lobes of rear sensitivity. This is common for stage mics and especially for vocalists. Different cardioid mics have different levels of sensitivity. Some have a wide pickup range, so they are good for a performer who moves around a lot, and some have a small area of sensitivity, so they are better at not picking up background noises.

pattern. They are effective at blocking sound from the sides. The figure- eight pattern is on the horizontal axis, so you can tip them sideways to get a different effect. A pair of matched ribbon mics placed next to each other at a 90-degree angle is called a *Blumlein pair*, or crossed figure eight. This creates a realistic stereo image. If you don't want to capture sounds from one side, for example the audience side, you'll have to block it somehow or find a ribbon mic with the capture pattern that you want.

Polar Patterns Important considerations for microphones are *polar patterns*. Polar pat- tern describes a mic's area of sensitivity, as Figure 2-11 shows. These two- dimensional diagrams don't show that polar patterns are three-dimensional, so keep in mind that they're not flat and horizontal; they encompass areas with height and depth.

Omnidirectional Picks up sound equally from all directions in a spher- ical area. Try arranging your band in a circle with an omnidirectional mic in the middle to get a spacious, natural sound. **Shotgun** The most highly directional of all with a long, narrow front and smaller rear pickup range. These are commonly used with movie cameras of all kinds, from film to digital, professional and consumer, and are favorites of wildlife photographers. **Bidirectional** Picks up sound equally well from front and back; does not pick up sounds from the side. (Not pictured in Figure 2-11.) **Half-omnidirectional, or hemispherical** Picks up about a 180-degree hemispherical area. You can get some nice live recordings with one of these because it has a wide pickup area to the front and does not pick up noises from behind. (Not pictured in Figure 2-11.)

You might be thinking, why not just use shotgun or cardioid mics for everything so you zero in on just what you want to record? Do whatever you want; let your own ears and taste be your guide and select your mics to suit the occasion. Different brands and models of mics have different levels and types of sensitivity. For example, some are forgiving of a singer who moves around a lot, while others do a good capture only up close. Wireless head- sets are wonderful for the energetic performer, and wireless mics mean no tripping over cables. Some mics emphasize bass frequencies more as you move closer, like radio DJs that boom forth with exaggerated bass. This is called the *proximity effect*.

The Cowboy Junkies *Trinity Session* album was recorded using a single *ambisonic* microphone that reportedly cost about $9,000. Ambisonics refers to surround-sound recording techniques and equipment that are supposed to produce a realistic, spatially natural sound. An ambisonic mic has multi- ple capsules to capture sound from different directions, anywhere from four to dozens. It's an interesting niche in audio production; look up *ambisonic* and *SoundField mics* if you are interested in learning more.

Which Microphones for Which Occasions? There are different microphones for all occasions, such as voice, guitar, drum, and so on. Vocalists can be especially picky, because different mics color their voices differently. You'll find plenty of passionate opinions on this subject. Keep in mind that there are a multitude of factors that affect how good a recording sounds to you: how it's edited, the type of equip- ment you're listening to, your listening location (home, friend's house, per- formance hall, outside, vehicle), and your mood and expectations. We've grown up with decades of recorded audio of all types: acetate, vinyl, differ- ent types of tape, and now digital. So what sounds "right" is heavily flavored by what we're used to hearing. Some folks still hanker for the sound of tinny '60s AM radio or boomy jukeboxes or quadrophonic eight-tracks. Some believe that vinyl sounds "warmer" than digital and that tube amps sound warmer than solid-state.

I've always wanted to do some blind tests just to see whether the picky audiophiles in my life can really hear a difference. Number one on my blind testing wish list is tube amp versus solid-state, and cold tube amp versus one that has had a 24-hour warm-up, because some of the aforementioned picky audiophiles insist that tube amps need a long warm-up period or they sound "cold." There are real differences between tube and solid-state. Tube amplifier systems drive a transformer, which in turn drives the speakers. The trans- former suppresses a lot of transients like spikes, pops, and clicks, resulting in a cleaner sound. A cold preamp has more noise than a hot preamp. Tubes also have a singular noise source, whereas semiconductor devices have mul- tiple noise sources, increasing the amount of noise you might hear. Still, on good-quality gear it's going to take some mighty fine hearing to detect the difference.

It is nearly impossible to define a "pure" experience, because even live performances are colored by their environments and equipment, ambience, and the way our brains process data. It always surprises me how much worse my favorite local band sounds on the recordings I make of its performances. During the shows, I'm having a great time and thinking they sound wonder- ful. Then later when I hear the playback, I hear all kinds of flaws: off tempo, out of key, lackluster, you name it. Maybe I'm too picky and too attuned to listening for mistakes on recordings; maybe during the live shows my brain is too busy having fun to notice flaws. Maybe I make lousy recordings.

The moral is sound quality depends on your own ears and experience— what sounds good and right to you is what matters. You can try for the most realistic fidelity or try for the best artistic and creative fidelity. It's all subjective.

Microphone Cables There are many brands of microphone cables, and you can waste a lot of money on a snooty brand name. Don't spend a mint; there are many reasonably priced, good-quality choices. You may run into some confusion over *balanced* and *unbalanced* cables. In the context of connecting micro- phones, an unbalanced cable is a coaxial cable that terminates in a tip-sleeve (TS) connector. It has a single conductor wire surrounded by a combina- tion shield and ground. These are effective at blocking external interfer- ence but are vulnerable to induced hum and noise and tend to be noisier than balanced cables. A balanced cable terminates in either a three-pin XLR connector or a TRS connector. It has two internal wires, one hot and one cold, surrounded by a shield that is not part of the signal path, so it supplies a cleaner signal. Balanced cables can run for much longer distances than unbalanced cables without picking up excessive noise.

Keep in mind that it is the signal that is balanced or unbalanced, and using a balanced cable will not make an unbalanced signal balanced. How- ever, a TS cable will convert a balanced signal to unbalanced. You need to match your cables to your mics and your preamp, amp, mixer, or what- ever you might be plugging into. Condenser mics that depend on phantom power mostly likely use three-pin XLR balanced cables, and dynamic mics

use balanced cables with both XLR and TRS connectors. A balanced signal doesn't care what carries it, so you can use XLR-to-TRS adapters as needed, provided that whatever you're plugging into sends the correct signal.

These days this shouldn't be something that you have to spend a lot of time figuring out, since most contemporary audio gear supports balanced mic connections.

Microphone cables are either stiff or flexible, depending on where they're going to be used. Flexible cables are for live performances, and stiff cables usually reside in studios where they are not moved very often.

Don’t cross electric lines with any of your audio cables if you can help it, because you may pick up interference. If you have to, cross them at right angles to reduce the overlap.

Smart Miking Placing your microphones for best results is an art in itself, and the only way to get good at it is to practice a lot. You want to be as close as possible but not so close that you pick up electronic interference or unwanted sounds such as lip smacks and spit from vocalists. Pop filters are helpful for vocalists, and windscreens are essential for outdoor recording.

The “3-to-1 rule” is a simple guide for microphone placement for live performances or in the studio when you have several mics and performers set up at the same time in the same room. When microphones are placed too close together, you may get squeals, dips and peaks, or other forms of unpleasant interference. The 3-to-1 rule means the distance between adja- cent mics should be approximately three times the distance between the mic and the sound source. If there are multiple amps present, as with my fave lo- cal band where every performer lugs their own amp onstage, these will cause problems too. Sometimes simply turning an amp to face in a different direc- tion will cure feedback problems.

Microphone stands are essential—don’t depend on hand-holding. Goosenecks take up less space and are fast to adjust, but over time they wear out. Boom stands last forever but take up more space. Some folks prefer tri- pod feet, which I always trip over, so I prefer weighted bases. Shock mounts and cages are great for isolating your mics from vibrations and don’t cost very much.

Microphone Preamp

Microphone preamps are the second most important devices in your audio chain, after ADC/DACs. As I’ve already talked about in this chapter, I have an M-Audio MobilePre and a Focusrite Saffire Pro 26. With these I don’t need an internal computer sound card or separate preamp, because they have their own built-in microphone preamps. However, even if you prefer to use an internal sound card or have good external recording interfaces, you may still want to use a separate microphone preamp. Let’s talk about why the preamp is so important.

A preamp—short for *preamplifier*—amplifies a low-level signal to *line level*. The output from a microphone, a turntable, and many instrument pickups is lower than line level. Line level is a standard analog audio signal voltage that is designed for connecting different audio components. What is this voltage? Well, that’s a good question because even though it is billed as a standard voltage, it varies depending on the manufacturer. Most are around 1 to 2 volts. A preamp has a significant effect on audio quality: A low-quality preamp will introduce noise and distortion. A good preamp amplifies the signal cleanly, without introducing defects or color.

NOTE *Audio terminology gets bent in all kinds of ways—many of the preamps you see for home hi-fi systems are not like microphone preamps because they don’t do any amplifi- cation but are just switching units where you plug everything in.*

Preamps range from bare-bones models that supply only *gain* (ampli- fication) and phantom power to gaudy delights larded up with all kinds of special effects, dials, and blinky lights. A lot of audio devices come with gobs of special effects because it costs practically nothing to add them, and they make you feel like you’re getting something special. If you like lots of special effects, this is a nice bonus; just don't let it distract you from a de- vice's real quality. At the least it is nice to have some physical knobs. Debates over which preamps are best almost take on a religious quality. Professionals might spend thousands of dollars on a single preamp. You're welcome to do this, but in my opinion you're better off starting out with inexpensive gear and investing in perfecting your recording techniques. Then when you're ready to move up to better gear, you'll appreciate the difference and know how to get the most out of it.

Speakers and Headphones

Having both speakers and headphones in your audio chain lets you hear your recordings in different ways. Studio monitor speakers are supposedly dead-flat and accurate and don't add any color of their own. They also tend to be expensive. My own studio speakers are a set of nice JBL three-way speakers. They're not real studio monitors, just nice speakers that I like. Headphones are essential—you need these for monitoring your recordings. Audio interfaces that include built-in zero-latency headphone ports are per- fect for monitoring. I seem to collect headphones: I have a nice Plantron- ics USB headset, which is great for recording podcasts; a set of Sennheiser headphones with an ordinary 1/4 TRS plug; and wireless Audio-Technica headphones. My studio speakers are powered by a nice old Pioneer SA 7500 stereo amplifier. I've had it repaired twice, and I'm going to keep it going as long as I can. It's rated at 45 watts per channel, which doesn't sound like much, but it powers those watts with some serious amperage. Amps are the real measure of power in an amplifier; that is what makes the difference be- tween a wimpy amp and a good strong clean amp. Wattage doesn't mean all that much—that's just the number salespeople like to focus on.

None of this is super-duper hi-fi, at least not according to fussy audio- phile standards, but they're all good components, and they please me.

Your Computer Must Have Muscle and Vast Drawers

Your computer should be a good modern machine with a high-powered CPU and a lot of RAM. My studio PC has an AMD Phenom triple-core CPU with 4GB RAM. Multicore CPUs make a big difference. A single-core CPU should perform fine for two-track recording and simpler recordings like podcasts and interviews. For example, I have an old ThinkPad with an 800 MHz CPU and 256MB RAM running Linux that performs nicely as a field recorder for interviews. For more than two tracks, multicore is the way to go. Don't worry about AMD versus Intel; they're both fine, so use whichever you like best.

You need as much storage as you can afford. CD-quality audio (44.1 kHz, 16-bit WAV) uses around 5MB per track per minute. Don't forget to add up all the tracks you are using and all the retakes. You can buy terabyte hard drives, and by the time you read this, they'll probably be even larger. An- other option is to combine the capacity of several hard drives in a RAID (redundant array of inexpensive disks) array. The two RAID levels that are useful for audio production are RAID 0 and RAID 10. RAID 0, also called *striping*, makes two hard disks look like one, so two 500GB drives appear as a single terabyte drive. RAID 0 is very fast but has the same weakness as a sin- gle drive—if one drive in the array fails, you'll probably lose all of your data. RAID 10 (using a good-quality hardware controller) is *mirroring* plus striping, so you get speed and redundancy. Use a good-quality hardware con- troller; you don't want some cheapie that dumps more load on your CPU but instead one that handles the load itself. It's more expensive of disks than the popular RAID 5, but it's more reliable and a lot faster—you get faster reads and writes and much faster recovery from a failed drive. I wouldn't use RAID 5 or 6 arrays for audio recording; in fact, I don't use them at all anymore because they're too fragile, they're too slow for writes, and they propagate parity errors too readily.

I wouldn't worry about building some super-duper RAID array for record- ing and editing with Audacity, unless you find yourself burning through tera- byte hard drives all the time. I use a single large hard drive on my studio PC and am ruthless with housekeeping and getting rid of unneeded files. I use a nice little four-disk Linux-powered RAID 10 server for backups.

Operating Systems

In this book I will be covering both Linux and Windows. Each one has its strengths and pitfalls. If you're a Windows user, XP is still the most reliable version even though Vista and Windows 7 have been released. You'll get the best hardware and software support and the best performance. Vista presents special problems because of its own high hardware requirements— it may bog down your system to the point that you can't comfortably use it

for audio recording, and driver support for many audio devices is immature. Audio hardware manufacturers seem a little more interested in Windows 7, but it is still a hog compared to XP. If you want to upgrade from XP, don't bother with Vista; go straight to Windows 7. If your audio hardware and soft- ware work well on XP, then keep them as long as you can.

Linux users have the usual hassles with hardware manufacturers pre- tending they don't exist no matter how many products they buy, hype, and give free support for. Appendix A will tell you what works well in Linux, and you'll also find links to sites with current information on audio hard- ware support in Linux. If it's any consolation, a lot of audio hardware ven- dors don't release very good Windows drivers either. Why? Who knows; it is a mystery that I waste too much time wondering about. Don't they want happy customers?

Latency is the enemy of quality audio, so please refer to Chapters 13 and 14 for tips on tweaking your operating system for best audio performance.

Here are system requirements per the Audacity documentation. Assume the same for Windows 7 as for Vista:

Windows 98, ME 128MB/500 MHz recommended, 64MB/300 MHz minimum **Windows 2000, XP** 512MB/1 GHz recommended, 128MB/300 MHz minimum **Windows Vista Home Basic** 2GB/1 GHz recommended, 512MB/ 1 GHz minimum **Windows Vista Home Premium/Business/Ultimate** 4GB/2 GHz recommended, 1GB/1 GHz minimum **Linux** "Audacity will run best with at least 64MB RAM and a 300 MHz processor," says the Audacity documentation for Linux users. I recommend a minimum 800 MHz CPU and 256MB RAM for podcasts, interviews, and two-track music recording and the most powerful three- or four-core CPU you can afford for multitrack recording and editing.

Portable Recording

There are several good methods for field recording. My two favorites are to fix up a netbook as a portable recording studio or to use a portable dig- ital recorder. Netbooks are so cool; I've been wishing for netbooks ever since I discovered computers. Ordinary laptops work fine too, and you get more powerful CPUs. You have all the same options as you do with a desk- top computer—your choice of preamps, mixers, and other audio interfaces, microphones, software—and you can do all of your editing on the spot and even burn CDs. You also have a nice screen and keyboard, instead of the tiny screens and buttons found on portable digital recorders.

There are a large number of them that are priced reasonably, and you can slip one in your pocket and carry it anywhere. Carry extra batter- ies and some extra storage cards, and you're ready for anything. My per- sonal favorite is the Zoom Handy H2 (Figure 2-12). It runs on two AA bat- teries and also has an AC adapter. It has four built-in good-quality micro- phones, so you can record either in two-channel stereo or in four-channel surround. It has no speakers, but it comes with earbuds, and it can also be used as a USB audio inter a computer. Its 1/8 line input can be connected directl sound board at concerts, and it also accepts an extern microphone. It uses SD cards for storage and support WAV and MP3 file formats. It costs about $150.

|ure 2-12: The Zoom H2 with an RCA-to- 1/8 stereo plug adapter for nnecting to a mixer or any other two-channel stereo RCA recording line output

Some other popular and excellent portable digital recorders are the Olympus LS10, Marantz PMD 620, Marantz PMD 660, Sony PCM-D50, Yama- ha Pocketrak 2G, and Zoom H4. All are under $600, and all have built-in mics. Ideally you'll be able to get your hands on them and give them good test drives before purchase because they share a common weakness—tiny LCD control panels with complicated menus. You'll also want to test noise levels, because some of them are nice and quiet when you use the internal mics but are scratchety with external mics. Most of them accept external mics—some have only 1/8 mini-jacks, and some accept full-sized XLR or TRS plugs. The ones that have XLR jacks don't always supply phantom pow- er for condenser mics. A nice option is to use a battery-powered condenser mic, and then you don't have to worry about phantom power.

NOTE *Devices that supply phantom power usually advertise "48v phantom power." But few mics actually use 48 volts. They usually use much less—as little as 8 to 10 volts.*

Pocket digital recorders work wonderfully well and are fun. These range from tiny keychain fobs that are the audio equivalent of sticky notes to lit- tle recorders for good-quality voice dictation to high-quality multichannel recorders. Let's see what goes into a high-quality unit.

Some other things to look at are battery life, type, and size of storage cards. Does it have any built-in storage, and what audio file formats does it support? Does it come with useful accessories such as AC adapters, wind- screen, earbuds, and stands?

An interesting variation is the M-Audio Micro Track II. This has no built-in mics but is a miniature two-channel recording studio meant to be used with high-quality external mics. It supports both dynamic and con- denser mics and supplies a full 48v phantom power for condenser mics.

I prefer to use a USB card reader for transferring files from a portable recorder to my computer. Usually it's faster, and it doesn't run down recorder batteries.

The Secret of Recording Your Own Great Audio

The "secret" behind making good-quality audio recordings isn't much of a secret: The most important factor is blocking out unwanted noises. Our brains are wonderful at ignoring sounds we don't want to pay attention to, but microphones give equal attention to all noises. Blocking unwanted noise is harder than it sounds because our modern world is very noisy: vehicle traf- fic, airplanes, appliances, televisions and stereos, fluorescent lights, $2,000 powered subwoofers roaming the streets at 120 decibels in $500 cars, con- struction, and so forth. High-frequency noises are easier to block than low- frequency sounds, as we all know from our futile attempts to escape those four-wheeled powered subwoofers, which pound their way through all barriers.

Computers add their own sounds—it's common to pick up hard drive and fan noise. So before you stock up on fancy recording equipment, job one is preparing your recording studio:

• Use a quiet room with sound-absorbing walls or wall coverings. Old car- pet and blankets work as well as spendy acoustic foam.

• Place a good directional microphone close to whatever you are record- ing and aim it carefully.

• Shield your mic from your PC.

• Adjust your sound levels carefully, neither too low nor too high.

• Mount your microphones in shock cages.

But, you may ask, why go to all that trouble? Why not just fix it later? It's all just software anyway. My dear reader, if it were as easy as silly TV shows and movies portray it, there would be no need for soundproofed music re- cording studios, and no one would ever holler, "Quiet on the set!" You can mitigate problems somewhat in Audacity, but you get better results making as good a recording as possible and saving the fix-its for problems you can't avoid. It's nothing like crime shows where ace audio techs can clean up any recording, no matter how mangled, to perfect high fidelity. That's beyond fiction and well into fantasy.

Hearing Ranges

People who live in so-called primitive societies, without all of our modern "conveniences," retain keen hearing well into old age. I suppose if they wanted to make audio recordings, they wouldn't have to work so hard at blocking unwanted background noises.

American Indians and Australia's Aborigines are reported to have hearing ranges of 10 Hz to 25 kHz. The average person is on the order of 32 Hz to 18 kHz. The excellent technical review- er for this book, Alvin Goats, has an extended hearing range of about 22 kHz. It is not as cool as it sounds because most sounds above 18 kHz are noise: fans, power supplies, speaker distortions, and such. He is sometimes accused of being hard of hearing due to all of the extra sounds masking what other people are saying. Gordon Hempton is a wonderful artist who calls himself the Sound Tracker (*http://www.soundtracker.com/*). Mr. Hempton has made a career out of recording the pure sounds of nature, with no human sounds at all. In 1992 he recorded the chorus of dawn around the globe. In later years he recorded the sounds of the Mississippi river according to the writings of Mark Twain and the sounds of Yosemite National Park according to the writings of John Muir. He uses a Neumann KU-81i Dummy Head (named Fritz) to simulate as closely as possible how we hear sounds. He has a released a number of high-quality CDs so you can hear what a world without noisy people sounds like.

Mr. Hempton began his recording career in the early 1980s. A lot of the locations that he recorded back then are now too noisy to record.

Your PC should be dedicated to the job and not used for anything else, neither gaming nor web surfing nor emailing nor anything, because you want all of your computer's power dedicated to recording. If you don't, you risk generating skips and stutters. Turn off screensavers, all power manage- ment, and any antivirus or antimalware software. (Windows users, do I need to say, don't be connected to the Internet after doing this?) Turn off all un- necessary services, scheduled jobs, and everything that is not essential.

Your microphone will pick up noise, vibrations, and interference from a surprising number of sources. If you still have an old cathode ray tube (CRT) monitor, replace it with a modern flat thin-film transistor liquid crys- tal diode (TFT-LCD) monitor, because CRTs emit radiation and noise. Some- times they even resonate to certain sounds and create echoes.

You can make an effective sound barrier between mic and computer for cheap by attaching a piece of carpet to a piece of plywood or particle

Now that you have some ideas on what you need, please visit Appendix A for a sampling of good audio gear in all price ranges. This should help you navigate the huge and splendid world of audio hardware.

Layla and Firecracker, the official studio dogs, wish you well (Figure 2-13).

Don’t Smoke

Smoking is bad for you, and it’s bad for computers and au- dio gear, especially microphones. It’s nasty for visitors to your studio—they may not appreciate breathing secondhand smoke and having to shower and change after they leave.

board. Give it feet so it stands on its own, and you have the equivalent of an expensive piece of a high-tech sound barrier. Both laptop and desktop machines should sit on nonresonating surfaces. In a pinch, you can set your laptop on a coat or a pillow, being careful to not block its cooling vents.

Although many how-tos advise making your studio as acoustically "dead" as possible, without any echoes or resonance, feel free to experiment. You might like how some things sound in a space with some hard surfaces. Psycho- acoustics come into play in professional recording studios; they don't make perfect anechoic chambers because those are so flat that they sound un- pleasant. There is no sound reflection, nothing that gives depth to sound. So, professional studios reduce the amount of random noise while preserv- ing some depth to the sound.

Visit Appendix A

Figure 2-13: Layla and Firecracker waiting for their cues

A great way to preserve and enjoy old recordings is to transfer them from any legacy medium—vinyl LPs, cassette tapes, reel-to-reel tapes, vintage 78s, videocassettes, even eight-track tapes—to CD. Or you can transfer them to a hard drive, solid-state drive, or whatever digital storage medium you prefer.

Transferring phonograph records to CD is in demand, and you might even be able to get a nice sideline going doing this. A lot of people are still hanging on to their record collections but are afraid to enjoy them because LPs are fragile. A lot of great albums have never been released on commercial CDs, or the modern CD remasters are not done well. Some people sim- ply prefer the sound of their old records.

Although you can copy any analog media and convert it to any digital audio format, in this chapter we'll talk mostly about transferring vinyl record albums and singles to CDs. Once you have converted your old analog media to a digital format, Audacity has a number of tools for cleaning up the sound quality. You may not always be able to perform perfect restorations, but you can reduce hiss, clicks, pops, and other defects to quite tolerable levels.

3

TRANSFERRING VINYL LPS (AND OTHER LEGACY MEDIA) TO CD

You can also customize dynamic range compression to suit your own needs, which is a nice thing because on modern popular CDs, dynamic range com- pression is abused to where it spoils the music. Even if they did it well, it might not be right for you, so Audacity lets you do it your way.

Finally, we'll discuss the merits of vinyl versus CD and other media, as well as the ins and out of connecting your various playback devices to your computer. If you need help setting up your hardware, skip ahead to the "Connecting Legacy Devices to Your Computer" on page 67 first.

For those of you with golden ears who are not satisfied with 16/44.1 CDs, the DVD-Audio format supports up to 24/196. That's right, 24 bits at 196,000 samples per second. DVD-Audio supports 5.1 surround sound, and so does Audacity in the 1.3.*x* releases. If you're not into surround sound, you can cram several CDs worth of music at 16/44.1 onto a DVD. We'll learn about authoring DVD-Audio disks in Chapter 6.

Okay then, let's dive into copying, editing, and then making CDs. Please review Chapter 1 if you need a refresher on the basics of using Audacity.

Preparing Vinyl LPs for Copying

First, clean up your vinyl records as well as you can. Sure, you can do a lot to clean up digital audio files, but it's not like on TV where the ace lab tech makes pristine restorations effortlessly. That's fiction, my dear readers, and we are stuck in the real world. It's better to start with the best-quality record- ing possible; it's less work, and you get better results.

I have my nice old Discwasher brushes from the olden days, and it's a good thing because the new ones are inferior. Real Discwasher brushes have a directional nap—hold them one way to clean the record, and then reverse your stroke on a clean lint-free cloth to clean the brush. There is an arrow embedded on the handle that points to the leading edge. You can use a real Discwasher brush dry or with a wet cleaning solution. The correct way to wet-clean with Discwasher is to apply the cleaning solution to the leading edge of the brush only, leaving the rest of it dry. You can clean a record on your turntable while it's rotating, but be careful you don't apply so much pressure that you damage the motor. Apply the wet leading edge of the brush for three to four turns, and then roll the brush to bring the dry part in contact with the record for another three to four turns. Give it time to dry completely before playing it, because playing a wet record can damage it. (However, on a record that is already in bad, nothing-to-lose condition, playing it wet might make it sound better. Moisten it carefully with distilled water or Discwasher D4 fluid, and give it a whirl. It won't hurt your stylus.) A nice thing to have for everyday cleaning is a carbon antistatic brush. These are always used dry and are pretty good at lifting out dust, lint, and other particles that try to make a home on your records. But they're no good for cleaning fingerprints, sticky goo, or other muck that requires a wet cleaner.

There are all kinds of cleaning solutions, microfiber cloths, brushes, and even wet power washers. The debates over the best ways to clean vinyl

The History of Record Production

The earliest recordings were made on rotating wax cylinders with a needle in the middle of a vibrating diaphragm attached to a horn, like an old-fashioned ear trumpet. The horn functioned like a microphone. The needle vibrations cut an uneven groove into the wax. Modern mono recordings were made using the same principle of a vibrating needle cutting into a softer mate- rial, except the needle was moved by a magnet.

Stereo came about when someone experimented with the angle of the magnet with respect to the needle and found that it could be controlled precisely enough to cut each side of the groove differently, creating two stereo channels. The technology for producing vinyl records, even with the advent of CDs and dig- ital audio, has continued to improve, and some record labels are still producing high-quality vinyl recordings. With all of these ad- vances, it's still a single needle doing the recording and playback for two channels, which results in a bit of *crosstalk*, so precision tuning of your turntable is required to get the best performance.

records are endless and loud; I shall leave it to you to do your own home- work and figure out what you prefer. You can pick up nasty, dirty, nothing- to-lose records to practice on at thrift shops for cheap. Given the variety of claims over what works best, I suspect that vinyl is tougher than we give it credit for.

NOTE *Never use any kind of alcohol on vintage 78 rpm records or any acetate or nonvinyl records because it will damage them. The earliest records were made of wax, and there were many different wax compositions using carnuba, beeswax, and other materials. People who know about these things recommend not using liquid cleaners at all. If you have vintage records, I recommend consulting experts who know how to handle them safely. Solutions containing alcohol are fine for modern vinyl records, and most record cleaners contain alcohol. Whatever cleaner you use, it must be something that leaves no residue.*

You should also invest in a stylus brush and cleaner, because gunk builds up on your stylus. This is less controversial; I use the Stanton SC-4 brush and cleaner, and they do the job just fine. Remember, you cannot be too care- ful when you're handling your turntable's stylus; handle it as gently as you possibly can by its mounting brace only. Never touch it with your fingers.

Use a stylus gauge to adjust the vertical tracking force of your stylus. High-quality cartridges require a mere 0.5 to 3 grams. Medium-quality styli, and those designed for DJs, go as high as 5 grams. Set the tracking force per the instructions for your particular hardware. Too light and too heavy will both cause too much wear, so you really want it just right.

Figure 3-1: Putting each song into a separate Audacity track

Depending on your turntable and tonearm, you may also have antiskate, vertical tracking angle, and azimuth adjustments. Your turntable documen- tation should tell you all about what these are and how to adjust them. The idea is to make correctly aligned contact without causing asymmetrical wear. It is worth spending some time tuning your turntable, and you may be sur- prised at how much difference tiny adjustments make.

Eight Steps to Converting Records to CDs

First let's list all the steps and then in the next sections go through them in detail. Vintage records require some special handling, which we'll get to in "Copying Vintage 78s" on page 66. If you don't know how to hook up your turntable, visit "Connecting Legacy Devices to Your Computer" on page 67 first. These are the steps to follow:

1. Set Audacity's frame rate to CDDA frames in the Selection toolbar. 2. Set the project rate to 32-bit float/44.1 or 16/44.1. 3. Copy your album into Audacity into one long track. 4. Make any fixes such as removing noise and pops, normalizing, compress-
ing, and deleting unnecessary bits. 5. Enter metadata. 6. Export the Audacity tracks to CD-ready audio files. 7. Use your favorite CD-writing software to copy your songs to a CD. 8. Pop your new CD into a player and enjoy.

The most time-consuming part is fixing defects. This chapter has some tips for common fixes, and Chapter 12 is devoted entirely to fix-its and cleanups.

I like to record singles a little differently than in step 3: I prefer to record each single into its own track, so it looks like Figure 3-1.