A CRASH COURSE TO MAKING YOUR MARK IN THE RECORDING INDUSTRY

EDITED BY KYLE P. SNYDER

CREATIVE LIVE

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ABOUT THE EDITOR

KYLE P. SNYDER

Kyle P. Snyder is an educator, engineer, and consultant dedicated to the advancement of audio education for engineers at every stage of development. He is a faculty member within the Ohio University School of Media Arts & Studies where he routinely teaches music production, recording, mixing and mastering, critical listening, sound design for film and video, and live event production. Snyder is also the Associate Director of Ohio University's High School Media Workshop, a unique opportunity which affords budding media creators the chance to test the waters for a few days while learning the latest techniques in audio & music, video, animation, and social media.

Professionally, Snyder is active in numerous organizations, including the Audio Engineering Society which he serves both internationally and locally. He is also proud to host and engineer the AES Journal Podcast, produced ten times each year in conjunction with the Journal of the Audio Engineering Society. Additionally, Snyder is a member of The National Academy of Recording Arts & Sciences, The Society of Professional Audio Recording Services, The Association for the Study of the Art of Record Production, and the Music & Entertainment Industry Educators Association.

Learn more about Kyle P. Snyder including recent writings and projects in traditional studio recording as well as classical recording, oral history, and acoustic ecology at www.kpsnyder.com.

CHAPTER

1 INTRODUCTION

INTRODUCTION

HOW TO USE THIS BOOK

We all have to start somewhere with audio but it's often not where we wind up. Asked when I first began "studying" audio (i.e., plugging things into a 16-channel mixer and hoping it didn't blow up), I'd have told you my big dream was to tour the world as a live engineer. Certainly I never would have predicted gravitating toward studio work let alone teaching! My own career path, which has been filled with varied twists and turns, has shown me the true difficulties of educating anyone about audio— no two people have quite the same collection of prior knowledge and life experiences; fewer still will require the exact same skills later in life. Thus, creating a book on audio and recording that's perfect for any one person is nearly impossible. When we break into this field (and long after), we go about it in our own unique way; that is why I'm eager to share this book with you. The resources you'll find in the following pages were designed to help you quickly understand a topic and best achieve success in our tumultuous, ever-changing world.

With this innovative eBook companion to creativeLIVE's unique online offerings, we've been able to offer you the best possible information across every topic imaginable in the recording industry in a format which finally makes sense. Think of this as a guidebook — an instruction manual, if you will.

What does that mean? Fair question! The way this book has been designed, it divides essential tasks and topics within the industry into more manageable, distinct sections:

1. INTRODUCTION / HOW TO USE THIS BOOK

2. UNDERSTANDING THE MUSIC BUSINESS

This section is a primer on the various roles found throughout the music business today and the role your project studio plays in its ever-changing success. It includes a high-level explanation of what beginners in the field need to know when it comes to licensing both their own and clients' music in the world of DIY, especially issues like fair use and sampling. It concludes with a brief discussion of what each individual working in audio needs to understand to begin monetizing his or her work, whether he or she is a trained audio professional or a garage-band enthusiast.

3. AUDIO FUNDAMENTALS

Engage here with a brief discussion of audio in various forms. It is exceedingly important that we understand the core tenants of audio, without which understanding why a microphone works becomes difficult. We'll also discuss the recording space in its various forms along with important considerations in play when creating an appropriate monitoring environment. Many individuals have made truly phenomenal recordings in nothing but a garage with sparse soundproofing while others have recorded in the most isolated rooms imaginable that they built themselves. No matter what your interest, commitment, or level of expertise is, we'll discuss the various options available for soundproofing; different ways in which to make a recording space and control room possible at different budgetary levels; and the interplay of other factors upon decision making such as the musical genre being recorded.

The latter half of this section focuses on audio equipment in one way or another. How does audio flow from device A to device B? Is it really that simple? What principles are in play, and what should the end-user be aware of? This section covers the myriad of audio connection standards that are bound to be encountered and more. Similarly, are all microphones created equal? Do you use certain microphones for specific tasks? An understandable yet comprehensive microphone selection guide will be provided. Finally, while the tools of the trade change at a drastic pace, we run through several different recommendations for studios operating in-the-box to on-location and everything in between.

4. PRE-PRODUCTION

In "Understanding The Music Business" we discuss the various roles that exist throughout the music business; which one of these hats will you be wearing? Will you be engineering as well as producing (or even playing)? We'll talk about how these seemingly disparate roles interconnect once everyone (or just one person) enters the studio and how to make it all work. Speaking of entering the studio, prior to hitting record, there's actually a great deal of work that can and should be done. In fact, the more effort that's put into pre-production the more successful your session is likely to be. So, let's evaluate what needs to be recorded. We must be familiar with the music itself, the instruments (a new one will occasionally surprise me still to this day) and certainly the recording space, if this is a remote job. Once we're familiar with the session's needs, we can make a plan: from an input list to a basic session diagram, these can make setup a breeze.

5. PRODUCTION

At long last, we're finally going to record! So, how do we actually accomplish this? A basic understanding of the various microphone techniques used for vocals, guitar, drums, and other common instruments are an excellent starting place. Along with proper microphone techniques, what is the best way to set up your room? Given that most entry-level studios have only one live room (and it's not uncommon to track in the control room), how can we control the bleed between instruments while maintaining a great vibe throughout the session, etc.? We'll discuss this along with important procedures worth following which make a session run smoothly before, during, and after the last note has been played. While every session is different from the last, these basic principles will serve you well.

6. POST PRODUCTION

With the recording session now over, it's time to explore one last substantial hurdle—editing numerous takes into one final, cohesive product. Many editing techniques are applicable across all workstations which we discuss first; however, given the ubiquity of Pro Tools as a mixing and editing environment, it seems particularly prudent to include a dedicated introduction to this digital audio workstation, an industry standard which everyone should make themselves familiar with.

Other tools and electronic instruments which can especially benefit project studios warrant a more in-depth examination within this section. We also discuss the appropriate mixing techniques for your project once the edits have been completed to ensure that everything sits correctly in the final mix before it's sent out into the world. Finally, given ideal circumstances, a mix would be sent out for mastering and then be sent on to distribution through any number of channels. However, today's project studio engineer is often saddled with this additional responsibility. We'll discuss the pros and cons of each path as well as how to prepare for and handle these and other challenges.

1

INTRODUCTION

HOW TO USE THIS BOOK

This book is divided, not necessarily by types of knowledge (analog, digital, and so on) but, by the types of tasks you're likely to face in the order you're likely to face them. We know that a predominant number of engineers learn well by actually performing the task at hand (learning by doing, if you will) which is why I was driven to design this as a guidebook; by starting at the beginning and working through to the end you're certain to find yourself well-educated on the fundamentals of audio as well as the appropriate techniques required by nearly every challenge you're likely to face. Moreover, this book was also designed as a tool for those who prefer to learn a given task when it comes up. Though some may jump straight to this text, many may be more inclined to learn through online tutorials, using this as an occasional resource. This is why we have strived to ensure that task-specific sections exist to help you through immediate questions, whether they come up while watching a creativeLIVE class or setting up for a session.

Finally, while we do start somewhere in audio, know that our journeys never end. Whether this book is your first look at the topic or yet another step toward self-improvement, I challenge you to become a lifelong learner because even the greatest engineers continue to hone their skills with every passing day. For some, this may mean studying the history of our industry, becoming involved in professional organizations, participating in online communities, or just collaborating with friends. I encourage you to seek out each of these opportunities when available as they can only further improve your education.

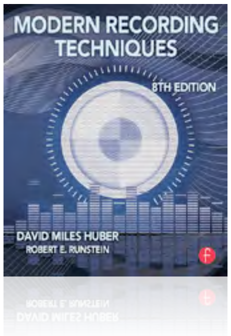
This collaboration between Focal Press and creativeLIVE is unique because it has allowed me to put before you what I feel is the *most valuable* information on a given topic, pulling from the depths of experience that Focal Press authors have to offer. If you enjoy what a particular author has to say about audio, I encourage you to read his or her entire book about which you can learn more at www.focalpress.com/audio.

KYLE P. SNYDER
www.kpsnyder.com

CHAPTER

2 UNDERSTANDING THE MUSIC BUSINESS

UNDERSTANDING THE MUSIC BUSINESS



The following is excerpted from *Modern Recording Techniques* by David Miles Huber and Robert E. Runstein.

You can purchase a copy HERE.

When you get right down to the important stuff, the recording field is built around pools of talented individuals and service industries who work together toward a common goal: producing, selling and enjoying music. As such, it's the people in the recording industry who make the business of music happen. Recording studios and other businesses in the industry aren't only known for the equipment that they have but are often judged by the quality, knowledge, vision and combined personalities of their staff. The following sections describe but a few of the ways in which a person can be involved in this multifaceted industry. In reality, the types and descriptions of a job in this techno-artistic industry are limited only by the imagination. New ways of expressing a passion for music production and sales are being created every day ... and if you see a new opportunity, the best way to make it happen is to roll up your sleeves and "just do it."

THE ARTIST

The strength of a recorded performance begins and ends with the artist. All of the technology in the world is of little use without the existence of the central ingredients of human creativity, emotion and individual technique. Just as the overall sonic quality of a recording is no better than its weakest link, it is the performer's job to see that music's main ingredient—its inner soul—is laid out for all to experience and hear. After all is said and done, a carefully planned and well-produced recording project is simply a gilded framework for the music's original drive, intention and emotion.

STUDIO MUSICIANS AND ARRANGERS

A project will often require additional musicians to add extra spice and depth to the artist's recorded performance. For example:

- An entire group of studio musicians might be called on to provide the best possible musical support for a high-profile artist or vocalist.
- A project might require musical ensembles (such as a choir, string section or background vocals) for a particular part or to give a piece a fuller sound.
- If a large ensemble is required, it might be necessary to call in a professional music contractor to coordinate all of the musicians and make the financial arrangements. The project might also require a music arranger, who can notate and possibly conduct the various musical parts.

- A member of a group might not be available or be up to the overall musical standards that are required by a project. In such situations, it's not uncommon for a professional studio musician to be called in to fit the bill. In situations like these, a project that's been recorded in a private studio might benefit from the expertise of a professional studio that has a larger recording room, an analog multitrack for that certain sound and/or an engineer that knows how to better deal with a complicated situation.

THE PRODUCER

Beyond the scheduling and budgetary aspects of coordinating a recording project, it is the job of a producer to help the artist and record company create the best possible recorded performance and final product that reflects the artist's vision. A producer can be hired for a project to fulfill a number of specific duties or might be given full, creative reign to help with any and all parts of the creative and business side of the process to get the project out to the buying public. More likely, however, a producer will act collaboratively with an artist or group to guide them through the recording process to get the best possible final product. This type of producer will often:

- Help the artist (and/or record label) create the best possible recorded performance and final product that reflects the artist's vision. This will often include a large dose of musical input, creative insight and mastery of the recording process.
- Assist in the selection of songs.
- Help to focus the artistic goals and performance in a way that best conveys the music to the targeted audience.
- Help to translate that performance into a final, salable product (with the technical and artistic help of an engineer and mastering engineer).

It's interesting to note that because engineers spend much of their working time with musicians and industry professionals with the intention of making their clients sound good, it's not uncommon for an engineer to take on the role of producer or co-producer (by default or by mutual agreement). Conversely, as producers and artists alike become increasingly more knowledgeable about recording technology, it's increasingly common to find them on the other side of the glass, sitting behind the controls of a console.

In addition, a producer might also be chosen for his or her ability to understand the process of selling a final recorded project from a business perspective to a label, film licensing entity or to the buying public. This type of producer can help with artist gain insights into the world of business, business law, budgeting and sales ... always an important ingredient in the process.

Of course, in certain circumstances, a project producer might be chosen for his or her reputation alone and/or for giving a certain cache to a project that can help put a personal "brand" on the project, thereby adding to the project's stature and hopefully help grab the public's attention.

One final thing's for certain, the artist and/or label should take time to study what type of producer is needed (if any) and then agree upon his or her creative and financial role in the project before entering into the creative process.

THE ENGINEER

The role of an engineer can best be described as an interpreter in a techno-artistic field. He or she must be able to express the artist's music and the producer's concepts and intent through the medium of recording technology. This job is actually best classified as a techno-artform, because both music and recording are totally subjective and artistic in nature and rely on the tastes, experiences and feelings of those involved. During a recording session, one or more engineers can be used on a project to:

- Conceptualize the best technological approach for capturing a performance or music experience
- Translate the needs and desires of the artists and producer into a technological approach to capturing the music
- Document the process for other engineers or future production use.
- Place the musicians in the desired studio positions
- Choose and place the microphones or pickup connections
- Set levels and balances on the recording console or DAW mixing interface
- Capture the performance (onto hard disk or tape) in the best way possible
- Overdub additional musical parts into the session that might be needed at a later time

- Mix the project into a final master recording in any number of media formats (mono, stereo, and surround-sound)
- Help in meeting the needs for archiving and/or storing the project

In short, engineers use their talent and artful knowledge of recording media technology to convey the best possible finished sound for the intended media, the client and the buying public.

ASSISTANT ENGINEER

Many studios often train future engineers (or build up a low-wage staff) by allowing them to work as assistants or interns who can offer help to staff and visiting freelance engineers. The assistant engineer might do microphone and headphone setups, run DAW setup or tape machines, help with session documentation, do session breakdowns and (in certain cases) perform rough mixes and balance settings for the engineer on the console. With the proliferation of freelance engineers (engineers who are not employed by the studio but are retained by the artist, producer or record company to work on a particular project), the role of the assistant engineer has become even more important. It's often his or her role to guide freelance engineers through the technical aspects and quirks that are peculiar to the studio … and to generally babysit the technical and physical aspects of the place.

Traditionally, this has been a no- or low-wage job that can expose a "newbie" to a wide range of experiences and situations. With hard work and luck, many assistants have worked their way into the hot seat whenever an engineer quits or is unexpectedly ill. As in life, there are no guarantees in this position—you just never know what surprises are waiting around the next corner for those who rise to the occasion.

MAINTENANCE ENGINEER

The maintenance engineer's job is to see that the equipment in the studio is maintained in top condition and regularly aligned and repaired when necessary. Of course, with the proliferation of project studios, cheaper mass-produced equipment, shrinking project budgets and smaller staffs, most studios will not have a maintenance engineer on staff. Larger organizations (those with more than one studio) might employ a full-time staff maintenance engineer, whereas outside freelance maintenance engineers and technical service companies are often called in to service smaller commercial studios in both major and non-major markets.

MASTERING ENGINEER

Often a final master recording will need to be tweaked in terms of level, equalization (EQ) and dynamics so as to present the final "master" recording in the best possible sonic and marketable light. If the project calls for it, this job will fall to a mastering engineer, whose job it is to listen to and process the recording in a specialized, fine-tuned monitoring environment. Of course, mastering is a techno-artistic field in its own right. Beauty is definitely in the ear of the beholding client and one mastering engineer might easily have a completely different approach to the sound and overall feel to a project than the next bloke. However, make no mistake about it—the mastering of a project can have a profound impact on the final sound of a project, and the task of finding the right mastering engineer for the job should never be taken lightly. Further info on mastering can be found in Chapter 19.

STUDIO MANAGEMENT

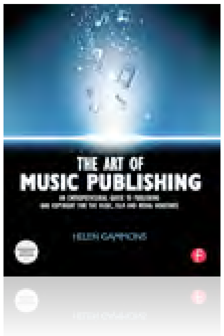
Studio management tasks include:

- *Management*: The studio manager, who might or might not be the owner, is responsible for managerial and marketing decisions for all of the inner workings of the facility and its business.
- *Bookings*: This staff person keeps track of most of the details relating to studio usage and billing.
- *Competent administration staff*: These assistants keep everyone happy and everything running as smoothly as possible.

Note, however, that some or all of these functions often vary from studio to studio. These and other equally important staff are necessary in order to successfully operate a commercial production facility on a day-to-day basis.

FOR MORE INFORMATION ON THIS TOPIC, PLEASE CHECK OUT THE FOLLOWING CREATIVELIVE VIDEO COURSE:

Check out Music Business 101: Networking HERE.



The following is excerpted from *The Art of Music Publishing: An Entrepreneurial Guide to Publishing and Copyright for the Music, Film, and Media Industries* by Helen Gammons.

You can purchase a copy HERE.

WHAT IS COPYRIGHT?

Let us begin by looking at the technical definitions and structure of copyright, before seeing how this may affect you.

Copyright is a protection that covers published and unpublished works. It exists at the point of creation, arising automatically. The copyright work, however, must exist in a material form, for example, that of a recording or sheet music.

Copyright subsists in the following works:

- original music works
- original artistic works
- original literary works
- original dramatic works
- films, sound recordings, broadcasts, cable programs, typographical arrangements of published editions.

The Copyrights Designs and Patents Act (CDPA) 1988 gives certain economic and moral rights to such works.

MASTERS AND RE-RECORDING RESTRICTIONS

Artists generally have the right to re-record their hits once their re-recording restriction has elapsed (usually somewhere between 10 and 20 years after the first release of the work, or a number of years after the conclusion of their contracted term). Many artists coming back into the marketplace are able to take full advantage of this taking control of their recordings (obviously they have to fund the new recordings themselves). If the artist is also the songwriter it puts them in control of both sides of copyright income: money derived for the copyright owners of the master and money derived for the copyright owners of the song. Artists are no longer reliant on record labels for financial support. Artists are now doing direct deals with interactive entertainment companies, telecommunications companies, equity investors, live agents and promoters. There are now many alternative-funding models to bring product into the marketplace. So while the record industry is shrinking and new competitors are entering this space who can release and distribute recordings, music publishers are well placed to handle all new areas of income stream development.

Just staying with master recordings for a second, you might also ponder on this point of view from Mick Hucknall (formerly the lead singer in Simply Red, but now a solo artist): if an artist eventually pays back the label who has invested in him for the development of the recordings to the point where that debt is repaid, surely then the ownership of that property, i.e. the recordings, should vest in the artist? I agree it's a fair position to take, but I wonder how many artists reach that point. If they do, surely their royalty should increase dramatically or they should see a reversion of those masters at some point. However, labels like having their cake and eating it.[1] Let's just go back one step and get the basics clear. Copyright exists in the song and also in the master recording. The artist and the record company control the master rights. The writer and the music publisher control the song or music rights.

[1] www.simplyred.com

FIGURE 2.1
SUMMARY OF AMERICAN COPYRIGHT.

(©2010 Peter B. Hirtle. Used with kind permission.)

NEVER PUBLISHED, NEVER REGISTERED WORKS [2]

Type of Work	Copyright Term	What was in the public domain in the U.S. as of 1 January 2010 [3]
Unpublished works	Life of the author + 70 years	Works from authors who died before 1940
Unpublished anonymous and pseudonymous works, and works made for hire (corporate authorship)	120 years from date of creation	Works created before 1890
Unpublished works when the death date of the author is not known [4]	120 years from date of creation [5]	Works created before 1890 [5]

WORKS REGISTERED OR FIRST PUBLISHED IN THE U.S.

Date of Publication [6]	Conditions [7]	Copyright Term [3]
Before 1923	None	None. In the public domain due to copyright expiration
1923 through 1977	Published without a copyright notice	None. In the public domain due to failure to comply with required formalities
1978 to 1 March 1989	Published without notice, and without subsequent registration within 5 years	None. In the public domain due to failure to comply with required formalities
1978 to 1 March 1989	Published without notice, but with subsequent registration within 5 years	70 years after the death of author. If a work of corporate authorship, 95 years from publication or 120 years from creation, whichever expires first
1923 through 1963	Published with notice but copyright was not renewed [8]	None. In the public domain due to copyright expiration
1923 through 1963	Published with notice and the copyright was renewed [8]	95 years after publication date
1964 through 1977	Published with notice	95 years after publication date
1978 to 1 March 1989	Created after 1977 and published with notice	70 years after the death of author. If a work of corporate authorship, 95 years from publication or 120 years from creation, whichever expires first
1978 to 1 March 1989	Created before 1978 and first published with notice in the specified period	The greater of the term specified in the previous entry or 31 December 2047
From 1 March 1989 through 2002	Created after 1977	70 years after the death of author. If a work of corporate authorship, 95 years from publication or 120 years from creation, whichever expires first

FIGURE 2.1
SUMMARY OF AMERICAN COPYRIGHT.

Continued.

From 1 March 1989 through 2002	Created before 1978 and first published in this period	The greater of the term specified in the previous entry or 31 December 2047
After 2001	None	70 years after the death of author. If a work of corporate authorship, 95 years from publication or 120 years from creation, whichever expires first
Anytime	Works prepared by an officer or employee of the United States Government as part of that person's offical duties [21]	None. In the public domain in the United States (17 U.S.C. § 105)

WORKS FIRST PUBLISHED OUTSIDE THE U.S. BY FOREIGN NATIONALS OR U.S. CITIZENS LIVING ABROAD [9]

Date of Publication	**Conditions**	**Copyright Term in the United States**
Before 1923	None	In the public domain (But see first special case below)

WORKS PUBLISHED ABROAD BEFORE 1978 [10]

1923 through 1977	Published without compliance with US formalities, and in the public domain in its source country as of 1 January 1996 (but see special cases) [20]	In the public domain
1923 through 1977	Published in compliance with all US formalities (i.e., notice, renewal) [11]	95 years after publication date
1923 through 1977	Solely published abroad, without compliance with US formalities or republication in the US, and not in the public domain in its home country as of 1 January 1996 (but see special cases)	95 years after publication date
1923 through 1977	Published in the US less than 30 days after publication abroad	Use the US publication chart to determine duration
1923 through 1977	Published in the US more than 30 days after publication abroad, without compliance with US formalities, and not in the public domain in its home country as of 1 January 1996 (but see special cases)	95 years after publication date

WORKS PUBLISHED ABROAD AFTER 1 JANUARY 1978

FIGURE 2.1
SUMMARY OF AMERICAN COPYRIGHT.

Continued.

After 1 January 1978	Published without copyright notice, and in the public domain in its source country as of 1 January 1996 (but see special cases) [20]	In the public domain
After 1 January 1978	Published either with or without copyright notice, and not in the public domain in its home country as of 1 January 1996 (but see special cases)	70 years after death of author, or if work of corporate authorship, 95 years from publication

SPECIAL CASES

1 July 1909 through 1978	In Alaska, Arizona, California, Hawaii, Idaho, Montana, Nevada, Oregon, Washington, Guam, and the Northern Mariana Islands ONLY. Published in a language other than English, and without subsequent republication with a copyright notice [12]	Treat as an upublished work until such date as first US-compliant publication occurred
Before 19 Aug. 1954	Published by a Laotian in Laos [18]	In the public domain
Between 18 Aug. 1954 and 3 Dec. 1975	Published by a Laotian in Laos [18]	Use the US publication chart to determine duration
Prior to 27 May 1973	Published by a national of Turkmenistan or Uzbekistan in either country [19]	In the public domain
After 26 May 1973	Published by a national of Turkmenistan or Uzbekistan in either country [19]	May be protected under the UCC
Anytime	Created by a resident of Afghanistan, Eritrea, Ethiopia, Iran, Iraq, or San Marino, and published in one of these countries [13]	Not protected by US copyright law until they become party to bilateral or international copyright agreements
Anytime	Works whose copyright was once owned or administered by the Alien Property Custodian, and whose copyright, if restored, would as of January 1, 1996, be owned by a government [14]	Not protected by US copyright law

FIGURE 2.1
SUMMARY OF AMERICAN COPYRIGHT.

Continued.

Anytime	If published in one of the following countries, the 1 January 1996 date given above is replaced by the date of the country's membership in the Berne Convention or the World Trade Organization, whichever is earlier: Andorra, Angola, Armenia, Bhutan, Cambodia, Comoros, Jordan, Democratic People's Republic of Korea, Micronesia, Montenegro, Nepal, Oman, Papua New Guinea, Qatar, Samoa, Saudi Arabia, Solomon Islands, Sudan, Syria, Tajikistan, Tonga, United Arab Emirates, Uzbekistan, Vietnam, Yemen	

SOUND RECORDINGS

(note: the following information applies only to the sound recording itself, and not to any copyrights in underlying compositions or texts.)

Date of Fixation/Publication	Conditions	What was in the public domain in the U.S. as of 1 January 2010 [3]

UNPUBLISHED SOUND RECORDINGS, DOMESTIC AND FOREIGN

Prior to 15 Feb. 1972	Indeterminate	Subject to state common law protection. Enters the public domain on 15 Feb. 2067
After to 15 Feb. 1972	Life of the author + 70 years. For unpublished anonymous and pseudonymous works and works made for hire (corporate authorship), 120 years from the date of fixation	Nothing. The soonest anything enters the public domain is 15 Feb. 2067

SOUND RECORDINGS PUBLISHED IN THE UNITED STATES

Date of Fixation/Publication	Conditions	What was the public domain in the U.S. as of 1 January 2010 [3]
Fixed prior to 15 Feb. 1972	None	Subject to state statutory and/or common law protection. Fully enters the public domain on 15 Feb. 2067
15 Feb 1972 to 1978	Published without notice (i.e., year of publication, and name of copyright owner) [15]	In the public domain

FIGURE 2.1
SUMMARY OF AMERICAN COPYRIGHT.

Continued.

15 Feb 1972 to 1978	Published with notice	95 years from publication, 2068 at the earliest
1978 to 1 March 1989	Published without notice, and without subsequent registration	In the public domain
1978 to 1 March 1989	Published with notice	70 years after death of author, or if work of corporate authorship, the shorter of 95 years from publication, or 120 years from creation. 2049 at the earliest
After 1 March 1989	None	70 years after death of author, or if work of corporate authorship, the shorter of 95 years from publication, or 120 years from creation. 2049 at the earliest

SOUND RECORDINGS PUBLISHED OUTSIDE THE UNITED STATES

Prior to 1923	None	Subject to state statutory and/or common law protection. Fully enters the public domain on 15 Feb. 2067
1923 to 1 March 1989	In the public domain in its home as of 1 Jan. 1996 or there was US publication within 30 days of the foreign publication (but see special cases)	Subject to state common law protection. Fully enters the public domain on 15 Feb. 2067
1923 to 15 Feb. 1972	Not in the public domain in its home as of 1 Jan. 1996. At least one author of the work was not a US citizen or was living abroad, and there was no US publication within 30 days of the foreign publication (but see special cases)	Enters public domain on 15 Feb. 2067
15 Feb. 1972 to 1978	Not in the public domain in its home as of 1 Jan. 1996. At least one author of the work was not a US citizen or was living abroad, and there was no US publication within 30 days of the foreign publication (but see special cases)	95 years from date of publication. 2068 at the earliest
1978 to 1 March 1989	Not in the public domain in its home as of 1 Jan. 1996. At least one author of the work was not a US citizen or was living abroad, and there was no US publication within 30 days of the foreign publication (but see special cases)	70 years after death of author, or if work of corporate authorship, the shorter of 95 years from publication, or 120 years from creation.
After 1 March 1989	None	70 years after death of author, or if work of corporate authorship, the shorter of 95 years from publication, or 120 years from creation.

FIGURE 2.1
SUMMARY OF AMERICAN COPYRIGHT.

Continued.

SPECIAL CASES

Fixed at any time	Created by a resident of Afghanistan, Eritrea, Ethiopia, Iran, or San Marino, and published in one of these countries [13]	Not protected by US copyright law because they are not party to international copyright agreements
Fixed prior to 1996	Works whose copyright was once owned or administered by the Alien Property Custodian, and whose copyright, if restored, would as of January 1, 1996, be owned by a government [14]	Not protected by US copyright law
Fixed at any time	If fixed or solely published in one of the following countries, the 1 January 1996 date given above is replaced by the date of the country's membership in the Berne Convention or the World Trade Organization, whichever is earlier: Andorra, Angola, Armenia, Bhutan, Cambodia, Comoros, Jordan, Democratic People's Republic of Korea, Micronesia, Montenegro, Nepal, Oman, Papua New Guinea, Qatar, Samoa, Saudi Arabia, Solomon Islands, Sudan, Syria, Tajikistan, Tonga, United Arab Emirates, Uzbekistan, Vietnam, Yemen	

ACHITECTURAL WORKS [16]

(Note: Architectural plans and drawings may also be protected as textual/graphics works)

Date of Design	Date of Construction	Copyright Status
Prior to 1 Dec. 1990	Not constructed by 31 Dec. 2002	Protected only as plans or drawings
Prior to 1 Dec. 1990	Constructed by 1 Dec. 1990	Protected only as plans or drawings
Prior to 1 Dec. 1990	Constructed between 30 Nov. 1990 and 31 Dec. 2002	Building is protected for 70 years after death of author, or if work of corporate authorship, the shorter of 95 years from publication, or 120 years from creation [17]
From 1 Dec. 1990	Immaterial	Building is protected for 70 years after death of author, or if work of corporate authorship, the shorter of 95 years from publication, or 120 years from creation [17]

UNDERSTANDING COPYRIGHT AND LICENSING

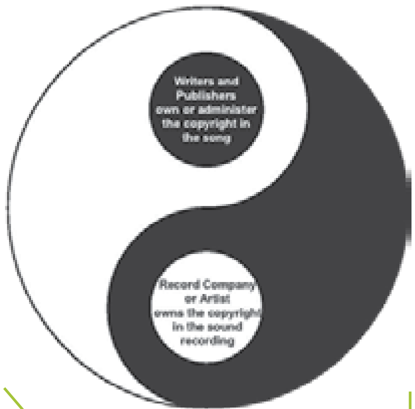


FIGURE 2.2
COPYRIGHT IN MUSIC – two parts working together. Copyright exists in the composition (song or music) and copyright also exists in the sound recording.

This is very important as there are separate organizations that control or represent either side of the industry. This is the one area that so many individuals get wrong. When music is cleared for use for a film synchronization, or a television synchronization, or another artist samples someone else's record, there is more than one set of permissions to get cleared and therefore there are twice as many contracts to enter into. Negotiations, permissions and contracts have to be entered into with all rights owners of the entire copyright (song and recording) **(Figure 2.2)**.

It follows that if you write and produce your own material (and many composers and artists now do) then you potentially own the copyright of the song and the master. If you have co-written with someone else, or you have worked together to produce and pay for the recording (or agree for this to be collectively owned) then the copyright is shared between the participants.

To clarify—if you co-wrote the song with someone then each composer would share in the copyright of the song. The writers would have to reach an agreement on how the song is split, and what percentage of the song they would each control. If you equally shared the writing then this would be 50% to each writer. Each writer can then decide independently how he or she wishes to have his or her share of the song represented.You can each select to be published by a different music publisher who will collect your share of the song for you. For example, 'Sister Sister', recorded by Beverley Knight, was written by Rod Gammons (G2 Music/Peer), David Hawk Wolinski (Rondor) and Beverley Knight (EMI). Each writer owned one-third of the song.

To verify this, go and read the label copy on your album and see for yourself who the writers are and who publishes each writer. On a lot of hip-hop titles where sampling is prevalent this can involve a very long list of publishers as each sample must recognize (and pay) each owner of the copyright involved in that track. If the samples are cleared in advance of being used there is every possibility of an amicable agreement being reached. However, if samples are not cleared then, in general, only the original copyright owners end up receiving all the money from the exploitation of the song.

OWNERSHIP OF THE MASTER RECORDING

For example, if you have created and paid for the recording as a band then you might agree, among all of you, that you all collectively own the recording and the copyright in that recording and therefore any money you generate from it will be shared equally. If an artist signs to a record label, the label will own the copyright in the master recordings as they generally front the money to pay for them.

U2 agreed from day one that irrespective of who writes which bit of the song they would share equally in the income derived from all exploitation of the songs. This has created a harmonious band relationship. This is not always the case, however, and each band or artist must decide for themselves what is best.

So you can see that you can have different owners of the song from those of the recording. It is often this point that really confuses people, but really it's quite straightforward. In the monetization of these rights all copyright owners have to supply their permission for the use of the copyright. This is usually done through the contract either between artist and record label, or between composer and music publisher. In each case the songwriter or the artist gives permission under the terms of their contract to the music publisher and/or the label, so that in simple terms they can act for the songwriter and artist in the exploitation of the copyright in the music and the recording. You can now begin to understand the foundations of the industry and the monetization of these rights.

The copyright owners are empowered to issue licences for the use of the songs or masters to whoever wishes to use them. Such areas might include advertising agencies, film companies, television companies, computer games companies, distribution companies and others. Vast sums of money are created and an industry worth billions of dollars (or pounds) exists worldwide. How the money that is created gets back to the songwriter or artist is subject to their contracts. For songwriters, music publishers and collection societies are the established route through which money flows.

COLLECTION SOCIETIES

The world of music publishing is structured with an interwoven network of collection societies. They break down into societies that collect different parts of the income stream to composers and publishers, namely income from performing rights (radio, television, film, touring) and mechanical income payable by record labels from the sale of recordings (digital, physical, streaming) containing the copyrighted works of songwriters and publishers. Their job is not just to represent local rights but to provide reciprocal protection and collection of everyone's rights when they are activated in their territory.

In other words, there is an integrated network where being a member of one society provides a common set of principles around the world for the protection of and the collection of royalties.

Collection societies have existed since 1852 with the birth of SACEM, the French Performing Right Society. MCPS and PRS (UK) were established in 1911 and 1914, respectively. The two UK societies have merged in recent years to reduce costs and find efficiencies for the benefit of its members. The UK society is now called PRS for Music. [4]

In the USA, three performing rights societies exist, which essentially do the same job, but have different characteristics. ASCAP is run and controlled by writer members and BMI is run and controlled by publisher members. SESAC grew primarily from looking after central European clients. All are proactive in supporting new composers and can assist worldwide in this regard. All have a proactive stance on education and sharing information on performing rights. Here's some information about each of them. Visit their websites and go and see their representatives in whatever country you are based—be proactive and find out more for yourself.

BMI

BMI does not charge writer members a fee to join, but there is currently (April 2010) a fee of US $150 if you wish to start an individually owned music publishing company through them, which rises to $250 for a corporation.

Formed in 1939 as a non-profit-making performing right organization, BMI was the first to offer representation to songwriters of blues, country, jazz, R&B, gospel, folk, Latin and, ultimately, rock 'n' roll.

BMI was founded by broadcasters to provide competition in the field of performing rights, specifically the songwriters and composers who were disenfranchised by the existing system. BMI was created to assure royalty payments to writers and publishers of music who were not represented by the existing performing rights organization, and to provide an alternative source of licensing for all music users.

BMI's history coincides with one of the most vibrant, evolving and challenging periods in music history. As popular music has moved from big band to rock 'n' roll and hip hop, and formats have evolved from 78 and 33$^{1}/_{3}$ rpm vinyl records to compact discs, MP3s and beyond, BMI has worked on behalf of its members to maintain a leadership position not only in the USA, but worldwide.

"Underlying everything BMI does is its philosophy; an open-door policy that welcomes songwriters, composers and music publishers of all disciplines, and helps them develop both the creative and business skills crucial to a career in music" (BMI, 2009).[5]

[4] www.prsformusic.com

[5] www.bmi.com (members include Eric Clapton, Elton John, Fall Out Boy, Daniel Powter, Colbie Caillat, Christina Aguilera, Josh Groban, Jennifer Lopez and many more).

ASCAP

There is a writer or publisher joining fee of US $25 as of 2010.

"ASCAP is a membership association of more than 350,000 US composers, songwriters, lyricists, and music publishers of every kind of music. Through agreements with affiliated international societies, ASCAP also represents hundreds of thousands of music creators worldwide. ASCAP is the only US performing rights organization created and controlled by composers, songwriters and music publishers, with a Board of Directors elected by and from the membership" (ASCAP, 2009).[6]

SESAC

"SESAC was established in 1930 and built on service, tradition and innovation. SESAC was founded in New York in 1930 by German immigrant Paul Heinecke, who, in an effort to help European publishers with their American performance royalties, established SESAC as the Society of European Stage Authors and Composers. Throughout the decades, until his death in 1972, Paul Heinecke guided SESAC with his own unique mix of old-world charm and 20th-century savvy. With an established cornerstone repertory of the finest European Classical Music, SESAC began to turn its attention to American music in the 1930s. Today, however, the company is known simply as SESAC. With an international reach and a vast repertory that spans virtually every genre of music, SESAC is the fastest growing and most technologically adept of the nation's performing rights companies".[7]

HARRY FOX

Harry Fox is the mechanical rights agency (collecting monies due to writers and publishers from the sale of each record, physical and digital) in the USA. Here's a little bit about them.

"The Harry Fox Agency represents music publishers for their mechanical and digital licensing needs. We issue licences and collect and distribute royalties on our affiliated publisher's behalf. This includes licensing for the recording and reproduction of CDs, ringtones, and internet downloads. HFA no longer issues synchronization licences for the use of music in advertising, movies, music videos, and television programs, but we do collect and distribute on synchronization licences that were granted prior to our discontinuation of synchronization service in 2002. HFA also conducts royalty examinations, investigates and negotiates new business opportunities, and pursues piracy claims".[8]

[6] www.ascap.com (Duke Ellington, Dave Matthews, George Gershwin, Stevie Wonder, Leonard Bernstein, Beyoncé, Marc Anthony, Alan Jackson, Henry Mancini, Howard Shore and many more).

[7] www.sesac.com/About/About.aspx

[8] www.harryfox.com

UNDERSTANDING COPYRIGHT AND LICENSING

SUMMARY

Copyright exists at the point of creation, but in order to enforce your rights there must be a physical form of the copyright in existence. The copyright owner has certain rights, but these are sometimes waived or adapted once contracts are issued and lawyers are involved. The waiving of these rights is usually requested, but a good lawyer will seek to bring these rights back into the contract (albeit in a diluted fashion) to ensure these rights are as protected as they can be and certain rights are correctly asserted. The waiving of the rights is often requested to ensure that the power within the contract is with the publisher and to ensure that the copyright owner (writer) cannot prevent any business they wish to conduct (i.e. if a record label failed to credit the writer, the writer could not come to the publisher and insist that they took action against the label and instruct all the copies to be removed from the stores).

The writer can ensure, however, that their right of paternity is respected by asserting these rights, so that if the label copy is incorrect the error is corrected on the next CD pressing, or on a digital download's meta-data, within a reasonable time-frame. The publisher, on behalf of the writer, would enforce this point. In addition, the writer can stipulate any areas of music and visual association that they feel are not acceptable to them or for which the publisher must seek the writer's additional approval. Such areas of concern may include an association with drugs, tobacco, animals or gratuitous violence. The writer should have the right to state whether the music can be used or not.

The thing to understand here is that the artist who has signed a contract with the record company (who owns the master) has similar rights. A film or television company must obtain clearance from both owners of the copyright (music and master). Temper this, however, with the understanding that a publisher will be trying to promote and develop income streams for the copyright work, so putting up too many obstacles may inhibit the writer's relationship with the publisher and cut back on potential income (but this is the writer's choice). If a publisher has advanced a lot of money for the copyrights then they will want to see a return on their investment. So don't be too difficult when it comes to agreeing to synchronization use.

When you start to look at and consider how you want your copyrights to be developed, ask yourself what your primary goals are.

MAKING MONEY IN THE MUSIC BUSINESS

by Jesse Cannon

Today's music business has allowed musicians to take matters into their own hands. Gone are the days of musicians waiting for a gatekeeper (someone who holds power and prevents you from being let in) at a label, late-night talk show or radio station to say they are worthy of the spotlight. In today's music business, you don't need to ask for permission to build a fanbase and you no longer need to pay thousands of dollars to do it. Every day, musicians are getting their music out to thousands of listeners without any help from industry bigwigs. They simply deliver it to the fans directly, without asking for permission or outside help, to receive exposure or connect with thousands of listeners.

THE MUSIC LANDSCAPE TODAY

While the Internet has opened new doors for musicians, many still think these opportunities are exceptions to the rule. They're not. Unlike in 2008, musicians who do it themselves have nearly identical opportunities to major label musicians who receive tens of thousands of dollars in promotional budgets. Musicians can now sell their albums in the same stores as their idols, play concerts in the same venues and get their music on humongous radio outlets like Pandora; it just requires a smart, diligent work ethic. As long as fans enjoy the music you make, you can build a fanbase just as easily as any musician with a huge budget behind them.

Every day there is more news of doom and gloom in the music business. What these reports don't always note is that the gloom is reserved for those who have traditionally held power in the industry, not those who are presently rising up in the business. In the old music business, gatekeepers (major labels, record stores, booking agents, radio, MTV, A&R, etc.) with questionable values and even more questionable methods held back musicians who didn't fit into a certain mold. These musicians then had no way to access essential tools to build a fanbase. That old music business crumbles a little more every day, simultaneously leveling the playing field between all types of musicians. Just a few years ago, there were many outlets and opportunities that a DIY musician could never gain access to. Today, there are very few opportunities that aren't open to those who do it themselves.

Half a decade ago, it seemed a musician's best hope was to put a song on Myspace, add friends, play shows and hope for the best. If they were any good, they might start to gain some fans and experience a small amount of success. But as Myspace disappeared and as major studios and record labels collapsed, we were suddenly left with a variety of blogs to dictate the cultural zeitgeist. The music business had changed and the way musicians approached building a fanbase had to change with it.

UNDERSTANDING THE MUSIC BUSINESS

MAKING MONEY IN THE MUSIC BUSINESS

by Jesse Cannon

Acts like Metric, Nine Inch Nails, A Day To Remember, Mach Miller, OK Go, Circa Survive, Coheed & Cambria, Amanda Palmer and Radiohead are now successfully selling records without a label to support them. In 2013 Macklemore & Ryan Lewis even have a Platinum selling #1 single and album (one million copies shipped) as a DIY act on their own label. New musicians are starting to wonder how they can emulate these larger acts, no longer just in sound, but also in business practice.

OBSCURITY

The biggest problem most musicians have is obscurity--not enough people know who you are. No one knows that they should listen to your music or give your song a chance to become their favorite track. Even if your music is readily available, people aren't aware that it's out there. When I'm in my studio working with musicians, I am asked two questions every single day: "How do we get more people to hear us?" and "How do we get more fans?" What musicians don't realize is that these are two different steps in the process of building a fanbase that leads to a career.

These are questions you'll be asking yourself for the rest of your life, even if you're U2, Metallica, Swedish House Mafia or some other huge act. You'll always want more people to be aware of what you do and to pay to be a part of it. Even when you're the biggest act within your genre, there are still potential fans that've never heard of you and these are people who might like your music if they did hear it. Part of keeping your fanbase growing is finding these untapped listeners, getting your music in their ears and converting them into fans.

When you're a lesser-known musician, this problem is much more daunting. No one is interested in writing about you and you can't just book gigs to get more exposure. You're fighting for every scrap of interest you can possibly get, even employing one-on-one interactions via social networks to persuade one potential fan to listen to your music.

HOW YOU GET FANS TO PAY ATTENTION TO YOU

Now that we know that you need to take matters into your own hands how do you get potential fans to pay attention to your music?

ALWAYS BE AVAILABLE

You want to have your music in as many places as possible. The more places you have your music, the more chances there are for someone to discover it. If you make merchandise, you want to have it abundantly available and worn by as many fans as possible—not only so you can fund your music, but also so you can turn your fans into a walking advertisement. If your fans want updates from you on a social network they frequent, give it to them on the network they prefer. The more people see you, the more chances you have for them to listen to your music and become a fan. Be available everywhere you possibly can.

BE EVENTFUL

Musicians who are good at marketing themselves online are well versed in doing cool promotions that get their fans excited. This is not a coincidence. This comes from putting hard work and thought into figuring out what will excite fans and how to turn every promotion into a major event that gets buzz. Being eventful means you are doing something worth talking about. Unique, personalized events will be the topic of even more talk being spread by your fans and writers alike.

DRAW ATTENTION

In order to get anyone to listen to you, you need to draw attention to yourself. This requires getting others to talk about you and actively taking part in eventful promotions. If you're not doing promotions worth talking about, there is no reason for fans to discuss you and point to your music.

FEED YOUR FANS

A constant stream of content (music, videos, blog posts, video updates or anything else that a fan can ingest and enjoy) will continue to make you an eventful act that tastemakers comment on and fans tell other potential fans about. Keep up a constant stream of eventful activity that your fans can discuss, so they always have something new to spread the word about. This helps put your name in front of more people, increasing the chance of making a new fan. If you're not creating new content on a regular basis, you lose your chance to raise awareness and you lose your chance to get discovered. **The more content you create, the more opportunities you have to be discovered.**

AFTER YOU HAVE THEIR ATTENTION PROFIT AND MAINTAIN YOUR RELATIONSHIPS

If you have ever been to a mall food court, you've probably experienced the chicken taste tactic. You walk into the food court and you may know that you want to eat at a specific place like Chipotle or you may still be undecided. Along the way, there is an employee of the local Chinese food stand, giving out free samples of their kung-pao chicken on a toothpick. They call out, inviting you to have a free taste. Inevitably, you walk over, take the free sample and then decide whether you like it. Accordingly, you decide whether to eat at the Chinese food stand or make your way to an established favorite, like Chipotle.

If you do decide to check out the Chinese stand, they're hoping for you to see all the other awesome products they have to offer: Other types of chicken, more expensive entrees, soda, cheap appetizers, even a funny cashier. You have now become aware of everything this stand has to offer, all because of your free sample. You might buy the most expensive thing on the menu or even become a regular customer. Even though the free sample cost the Chinese stand five cents, you might start spending $30 a week at their business. That works out pretty well for them, huh?

This process is used in every supermarket and food court--and even in magazine advertisements that include a free perfume sample. All types of business give away small samples for free, hoping consumers will keep coming back for more and more. If your product—in this case, your music—is good enough, you'll be able to use a free sample to create a new loyal customer—in this case, a new diehard fan.

IN MUSIC TERMS

To draw a parallel to that chicken sample, as a musician you need to offer your fans a way to ingest your music for free, enabling them to decide if they're interested in seeing what else you have available. This is the most effective way to turn a listener into a fan. If a new listener enjoys your free song, you lead them to your website—where you have more streaming music waiting for them, along with videos, free downloads or other types of content. If they like what they see and hear, you have to make sure you introduce them to other content—this time, content they have to pay for.

New listeners should be presented with a multitude of options across a few different price points. These options can include EPs (the equivalent to an appetizer), LPs (small entree), a "deluxe package" that includes maybe a T-shirt paired with music (expensive entree) and finally some stickers, coffee mugs, etc. (soda and sides). Having options helps ease the transition from new listener to financially supportive fan. This is the goal of every interaction with a new listener—and since you can't communicate directly with every single person, you need to make sure your online presence is built to do this for you.

Once this listener becomes a paying customer and a new fan, they will hopefully help expose you to more potential fans. Maybe they will bring a friend to your concert or share your music with someone—like you might bring a co-worker to the Chinese stand with you. That co-worker becomes a regular customer of the stand and then tells another friend about it, just like what can happen when someone shares your music on Facebook or Twitter. All of this exposure is possible when you make free samples of your music available. You begin a process that opens the gates for your music to be spread around and you have the opportunity to attract listeners and turn them into fans with very little friction.

THE PRICE OF A RELATIONSHIP

By eating the small cost of a free food sample, the Chinese stand created a regular customer. That customer now helps the stand make money on a regular basis by eating there and exposing the stand to other people. This is what you're trying to do with each and every listener. If your song is only available for 99 cents on iTunes, you may get some interested parties who bite the bullet and pay for it without getting to listen to it beforehand. But after Apple takes its cut and after taxes, you only see about 40 cents of that dollar anyway. The potentially random income of 40 cents from an iTunes sale isn't worth the number of potential fans it deters from listening to your music. By having that song streaming for free, you're maximizing the relationships you build—opening up the door to many more real fans than you would ever gain if you keep your music behind a pay wall. This also helps you grow a genuinely interested fanbase in an organic way—these are the people that may support you with hundreds of dollars every year by purchasing music, show tickets, merchandise, etc.

Turning a listener into a fan and then a paying customer is how you pursue your dreams and further your career. This is why it's essential to first worry about making fans, then worry about making money from them. If you're having trouble building a following, there's probably too much friction in the road from listener to fan. Tear down those borders and you'll watch your fanbase rise easier than ever before.

If you're not convinced yet, we'll delve much deeper into the nuances of this topic later in the book.

BUILDING RELATIONSHIPS

Look at it this way: You're not in the music sales business – as so many musicians think they are – you're in the relationship business. Think about how all of your relationships have grown throughout your life. You meet someone and develop an interest in them. Eventually, you might put a title on the relationship, like friend, boyfriend, girlfriend or in this case, fan. Now imagine if you proposed to a person that they should take on this new title when all they knew was your name. That would be a little uncomfortable, right?

This is what musicians are doing when they try to sell music to potential fans before they even hear it. This is a marriage proposal before the friendship has begun. It's much smarter to give your fans an introduction, showing them your awesome personality and talents. If they're impressed, you take the relationship to the next level. Take the time to impress this potential partner and fall in love before you get hitched and trade your financial information, just like you would a marriage.

Once upon a time, chunks of music sales occurred after a fan discovered a song on MTV or the radio. Now these outlets are becoming outdated and instead fans usually discover music through a streaming service like YouTube or SoundCloud—the two most popular ways to discover new music, according to MusicMetric. Usually, this exposure is the result of sharing via social networking. This opportunity is hindered when musicians hide their music in previews or don't post it online for fans to digest. It's very hard to build a relationship if you aren't active on the sites and services where people most often discover new music.

NURTURING

When musicians take advantage of the ways a listener discovers music, they're nurturing a fan relationship. Instead of trying to get discovered by writing letters and emails to A&R people at record labels, DJs at radio stations or video reps at cable networks, musicians are turning to fans that give them strong endorsements until the right people notice them.

By focusing on your relationships with your fans instead of waiting for others to give you a chance, you can build an army of fans who not only promote your music, but also financially support you. Taking the time to connect with fans – feeding them the content they want – starts a mutually beneficial relationship. The fans enjoy the content you create and you get the financial support you need to create it all. Luckily, you can now connect directly with these fans, communicate with them, find out what they want and deliver it straight to them.

Perhaps the best part about the way we communicate today is the two-way relationship between fan and artist. In the past, musicians would mostly talk to their fans, with very little response back. Maybe a fan joined their mailing list or street team, but staying in touch took a hugely concentrated effort. Sending out promos to a physical mailing list took hours of work and tons of materials and postage. That process cost record labels a significant amount of money—today, all it takes is 20 minutes and an Internet connection.

Just like what happens within a friendship, you can now exchange contact information with your fans. You're able to get in touch with each other whenever you have something to say. If you want to tell your fans what you're up to, whether it's news about a new tour or just a funny story, you can deliver that message to them immediately and directly. When your fans want to tell you it's been too long since you released music, that you should be playing this venue instead of that venue or that they just heard your song on the radio, they can do so in a heartbeat via Twitter, email or a handful of other outlets.

Best of all, this communication can start a meaningful relationship with fans that allows you to connect and create superfans like never before. Surveys show that fans are more likely to spend money on musicians they feel they have a personal connection with. Even if your biggest concern is about getting rich, as opposed to the art of music, you still need to focus on building relationships with your fans. This means talking to concert attendees, distributing content on your social networks, demonstrating your personality and most of all, giving fans what they want. If your fans want hooded sweatshirts, print them and sell them. If they want live video broadcasts, you can do it in seconds on YouTube. If they want a new record and you don't have the money for it, you can start a crowdfunding campaign and create a new record your fans fund themselves.

If you're not willing to go out of your way to make one new fan at a time, you're at a disadvantage to the thousands of musicians who are. Most fanbases are built a few fans at a time. It's very rare that you're going to see huge jumps in fans in the early stages of accumulating your following. Instead, you're going to have to concentrate on going out of your way to do more than just spam potential fans on social networks. You're going to have to put effort into one-on-one connections until you have enough fans that you can empower them to make other fans for you on a regular basis. Even then, going out of your way for your fans will ensure that they continue to become bigger fans and tell more of their friends about you.

You can build these relationships your own way. Some musicians do it through constant touring. Musicians like Justin Bieber and Lady Gaga do it through small interactions with fans on Twitter and connecting fans together on social networks based around their music—like Lady GaGa's Little Monsters network. The group Chester French takes more time to write their fans personal letters than they do pursuing press interviews. Alex Day makes countless YouTube videos and interacts with fans in the comments sections of the videos. By interacting with your fans, you strengthen your relationship and the support they give you. Studies show time and time again, **fans support musicians they feel they have the closest connections with.**

MAKING THE PROJECT STUDIO PAY FOR ITSELF

Beyond the obvious advantage of being able to record when, where and how you want to in your own project studio, there are several additional benefits to working in a personal environment. Here are ways that a project studio can help subsidize itself, at any number of levels:

- Setting your own schedule and saving money while you're at it! An obvious advantage of a project studio revolves around the idea that you can create your own music on your own schedule. Part of the expense of using a professional studio comes from having to be practiced and ready to roll on a specific date or range of dates. Having your own project studio frees you up to lay down practice tracks and/or record when the mood hits, without having to worry about punching the studio's time clock.
- For those who are in the business of music, media production or the related arts business, the equipment, building and utility payments can be written off as a tax-deductible expense (see Appendix B, "Tax Tips for Musicians").
- An individual artist or group might consider pre-producing a project in their own studio ... allowing the time and expense billings to be tax deductible.
- The same artists might consider recording part or all of their production at their own project studio. The money saved (and deducted) could be spent on a better mix down facility, professional freelance engineer, solving legal issues (such as copyright) and/or marketing.
- The "signed artist/superstar approach" refers to the mega-artist who, instead of blowing the advance royalties on lavish parties in the studio (a sure way never to see any money from your hard work), will spend the bucks on building their own professional-grade project studio. After the project has been recorded, the artist will still have a tax-deductible facility that can be operated as a business enterprise. When the next project comes along, the artist will still have a personal facility where they can record and put the saved advance bucks in the bank.

2.3

UNDERSTANDING THE MUSIC BUSINESS

MAKING MONEY IN THE MUSIC BUSINESS

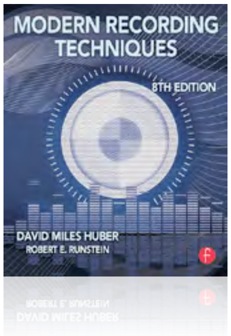
by Jesse Cannon

FOR MORE INFORMATION ON THIS TOPIC, PLEASE CHECK OUT THE FOLLOWING CREATIVELIVE VIDEO COURSE:

Check out Making Money with Music HERE.

CHAPTER

3 AUDIO FUNDAMENTALS



The following is excerpted from *Modern Recording Techniques* by David Miles Huber and Robert E. Runstein.

You can purchase a copy HERE.

When we make a recording, in effect we're actually capturing and storing sound into a memory media so that an original event can be re-created at a later date. If we start with the idea that sound is actually a concept that corresponds to the brain's perception and interpretation of a physical auditory stimulus, the study of sound can be divided into four areas:

- The basics of sound
- The characteristics of the ear
- How the ear is stimulated by sound
- The psychoacoustics of hearing

By understanding the physical nature of sound and the basics of how the ears change a physical phenomenon into a sensory one, we can discover how to best convey this science into the subjective art forms of music, sound recording and production.

THE BASICS OF SOUND

Sound arrives at the ear in the form of periodic variations in atmospheric pressure called *sound-pressure waves*. This is the same atmospheric pressure that's measured by the weather service with a barometer; however, the changes in pressure heard by the ear are too small in magnitude and fluctuate too rapidly to be observed on a barometer.

An analogy of how sound waves travel in air can be demonstrated by bursting a balloon in a silent room. Before we stick it with a pin, the molecular motion of the room's atmosphere is at a normal resting pressure. The pressure inside the balloon is much higher, though, and the molecules are compressed much more tightly together—like people packed into a crowded subway car **(Figure 2.1a)**. When the balloon is popped... KAPOW! **(Figure 2.1b)**, the tightly compressed molecules under high pressure begin to exert an outward force on their neighbors in an effort to move toward areas of lower pressure. When the neighboring set of molecules has been compressed, they will continue to exert an outward force on the next set of lower-pressured neighbors **(Figure 2.1c)** in an ongoing outward motion that continues until the molecules have used up all their energy in the form of heat.

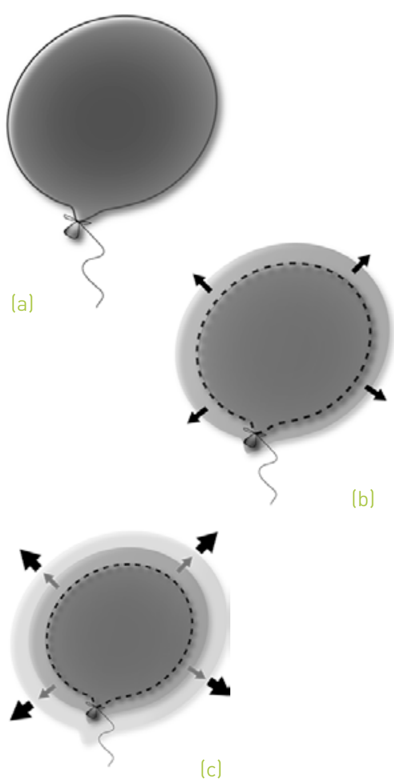


FIGURE 2.1
Wave movement in air as it moves away from its point of origin. (a) An intact balloon contains pressurized air. (b) When the balloon is popped, the compressed molecules exert a force on outer neighbors in an effort to move to areas of lower pressure. (c) The outer neighbors then exert a force on the next set of molecules in an effort to move to areas of lower pressure... and the process continues.

Likewise, as a vibrating mass (such as a guitar string, a person's vocal chords or a loudspeaker) moves outward from its normal resting state, it squeezes air molecules into a compressed area, away from the sound source. This causes the area being acted on to have a greater than normal atmospheric pressure, a process called *compression* **(Figure 2.2a)**. As the vibrating mass moves inward from its normal resting state, an area with a lower-than-normal atmospheric pressure will be created, in a process called *rarefaction* **(Figure 2.2b)**. As the vibrating body cycles through its inward and outward motions, areas of higher and lower compression states are generated. These areas of high pressure will cause the wave to move outward from the sound source in the same way waves moved outward from the burst balloon. It's interesting (and important) to note that the molecules themselves don't move through air at the velocity of sound—only the sound wave itself moves through the atmosphere in the form of high-pressure compression waves that continue to push against areas of lower pressure (in an outward direction). This outward pressure motion is known as *wave propagation*.

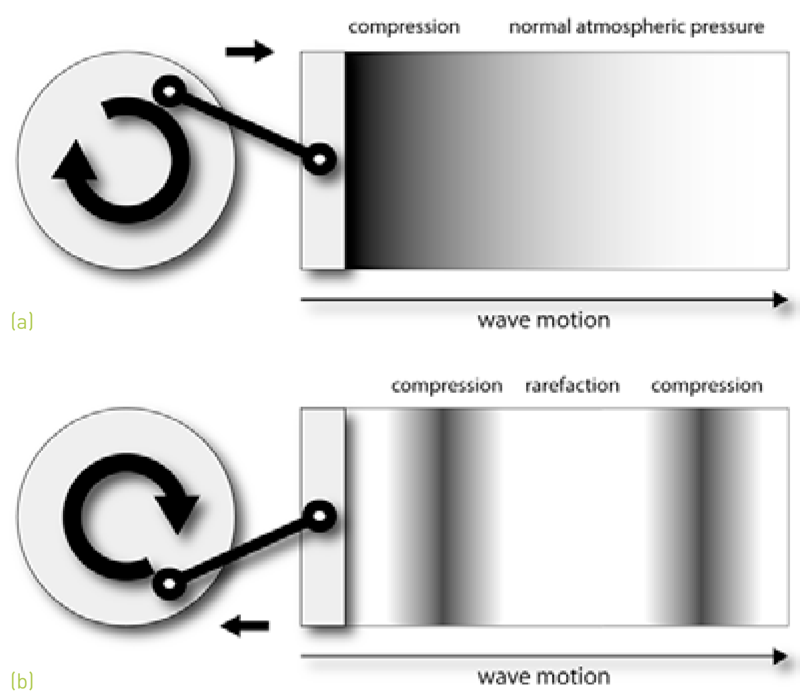


FIGURE 2.2
Effects of a vibrating mass on air molecules and their propagation. (a) Compression—air molecules are forced together to form a compression wave. (b) Rarefaction—as the vibrating mass moves inward, an area of lower atmospheric pressure is created.

WAVEFORM CHARACTERISTICS

A *waveform* is essentially the graphic representation of a sound-pressure level or voltage level as it moves through a medium over time. In short, a waveform lets us see and explain the actual phenomenon of wave propagation in our physical environment and will generally have the following fundamental characteristics:

- Amplitude
- Frequency
- Velocity
- Wavelength
- Phase
- Harmonic content
- Envelope

These characteristics allow one waveform to be distinguished from another. The most fundamental of these are amplitude and frequency **(Figure 2.3)**. The following sections describe each of these characteristics. Although several math formulas have been included, it is by no means important that you memorize or worry about them. It's far more important that you grasp the basic principles of acoustics rather than fret over the underlying math.

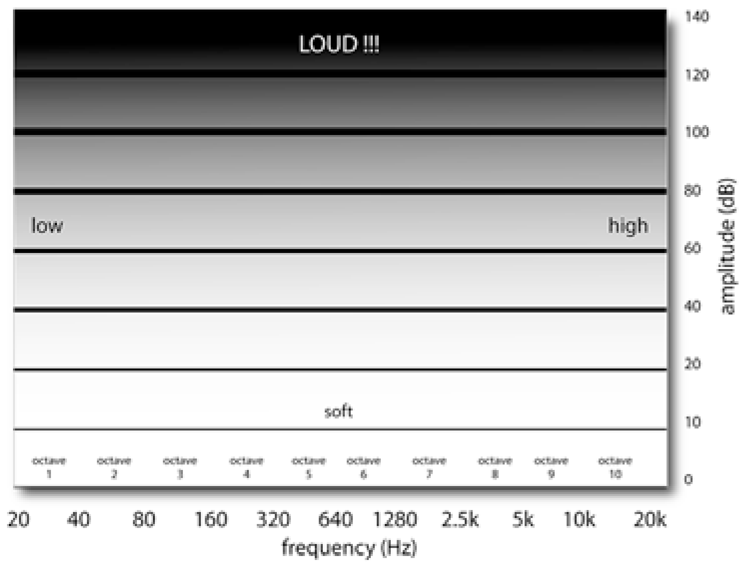


FIGURE 2.3
Amplitude and frequency ranges of human hearing.

AMPLITUDE

The distance above or below the centerline of a waveform (such as a pure sine wave) represents the *amplitude* level of that signal. The greater the distance or displacement from that centerline, the more intense the pressure variation, electrical signal level, or physical displacement will be within a medium. Waveform amplitudes can be measured in several ways **(Figure 2.4)**. For example, the measurement of either the maximum positive or negative signal level of a wave is called its *peak amplitude value* (or peak level). The total measurement of the positive and negative peak signal levels is called the peak-to-peak value. The *root-meansquare (rms) value* was developed to determine a meaningful average level of a waveform over time (one that more closely approximates the level that's actually perceived by our ears and gives a better real-world measurement of overall signal amplitudes). The rms value of a sine wave can be calculated by squaring the amplitudes at points along the waveform and then taking the mathematical average of the combined results.

The math isn't as important as the basic concept that the rms value of a perfect sine wave is equal to 0.707 times its instantaneous peak amplitude level. Because the square of a positive or negative value is always positive, the rms value will always be positive. The following simple equations show the relationship between a waveform's peak and rms values:

- rms voltage = 0.707 x peak voltage
- peak voltage = 1.414 x rms voltage

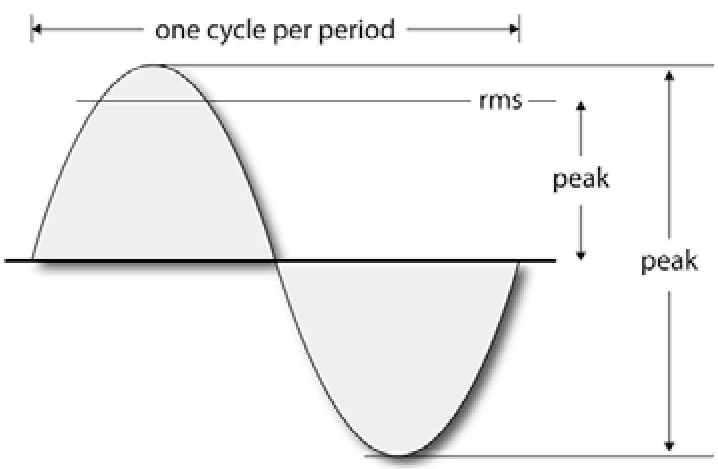


FIGURE 2.4
Graph of a sine wave showing the various ways to measure amplitude.

FREQUENCY

The rate at which an acoustic generator, electrical signal or vibrating mass repeats within a cycle of positive and negative amplitude is known as the frequency of that signal. As the rate of repeated vibration increases within a given time period, the frequency (and thus the perceived pitch) will likewise increase …and vice versa. One completed excursion of a wave (which is plotted over the 360° axis of a circle) is known as a cycle **(Figure 2.5)**. The number of cycles that occur within a second (frequency) is measured in hertz (Hz). The diagram in **Figure 2.6** shows the value of a waveform as starting at zero (0°). At time $t = 0$, this value increases to a positive maximum value and then decreases back through zero, where the process begins all over again in a repetitive fashion. A cycle can begin at any angular degree point on the waveform; however, to be complete, it must pass through a single 360° rotation and end at the same point as its starting value. For example, the waveform that starts at $t = 0$ and ends at $t = 2$ constitutes a cycle, as does the waveform that begins at $t = 1$ and ends at $t = 3$.

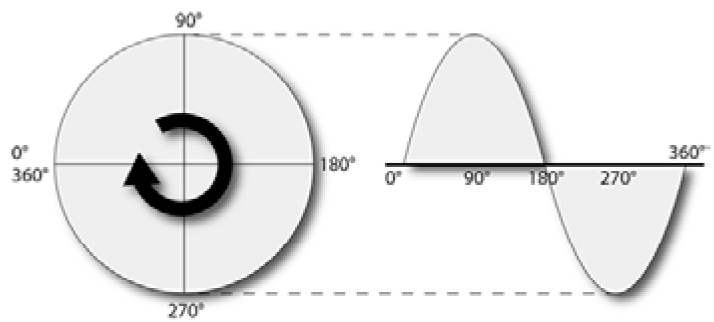


FIGURE 2.5
Cycle divided into the 360° of a circle.

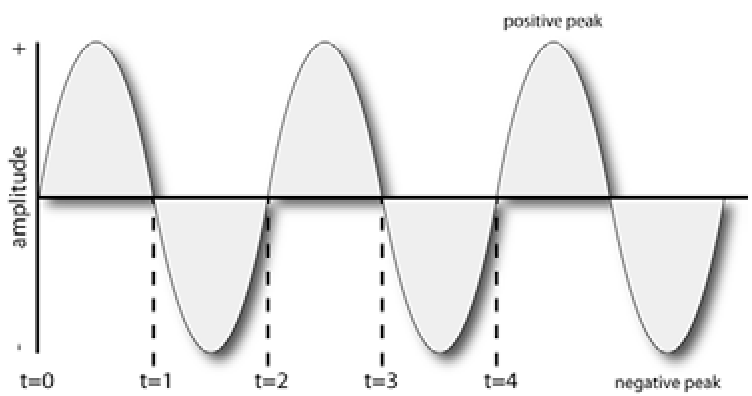


FIGURE 2.6
Graph of waveform amplitude over time.

VELOCITY

The *velocity* of a sound wave as it travels through air at 68°F (20°C) is approximately 1130 feet per second (ft/sec) or 344 meters per second (m/sec). This speed is temperature dependent and increases at a rate of 1.1 ft/sec for each Fahrenheit degree increase in temperature (2 ft/sec per Celsius degree).

WAVELENGTH

The *wavelength* of a waveform (frequently represented by the Greek letter lambda, λ close parentheses is the physical distance in a medium between the beginning and the end of a cycle. The physical length of a wave can be calculated using:

$$\lambda = V/f$$

where λ is the wavelength in the medium

V is the velocity in the medium

f is the frequency (in hertz)

The time it takes to complete 1 cycle is called the period of the wave. To illustrate, a 30-Hz sound wave completes 30 cycles each second or 1 cycle every 1/30th of a second. The period of the wave is expressed using the symbol *T*:

$$T = 1/f$$

where *T* is the number of seconds per cycle.

Assuming that sound propagates at the rate of 1130 ft/sec, all that's needed is to divide this figure by the desired frequency. For example, the simple math for calculating the wavelength of a 30-Hz waveform would be 1130/30= 37.6 feet long, whereas a waveform having a frequency of 300 Hz would be 1130/300= 3.76 feet long (Figure 2.7). Likewise, a 1000-Hz waveform would work out as being 1130/1000 = 1.13 feet long, and a 10,000-Hz waveform would be 1130/10,000= 0.113 feet long. From these calculations, you can see that whenever the frequency is increased, the wavelength decreases.

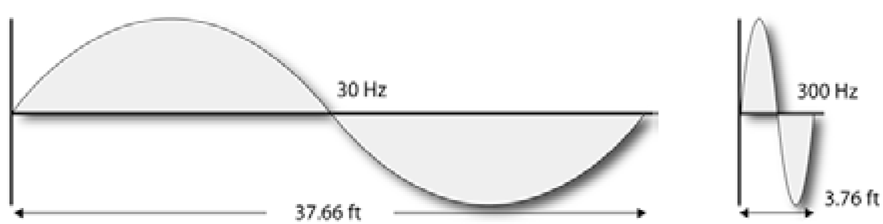


FIGURE 2.7
Wavelengths decrease in length as frequency increases (and vice versa).

REFLECTION OF SOUND

Much like a light wave, sound reflects off a surface boundary at an angle that's equal to (and in an opposite direction of) its initial angle of incidence. This basic property is one of the cornerstones of the complex study of acoustics.

For example, **Figure 2.8a** shows how a sound wave reflects off a solid smooth surface in a simple and straightforward manner (at an equal and opposite angle). **Figure 2.8b** shows how a convex surface will splay the sound outward from its surface, radiating the sound outward in a wide dispersion pattern. In **Figure 2.8c**, a concave surface is used to focus a sound inward toward a single point, while a 90° corner (as shown in **Figure 2.8d**) reflects patterns back at angles that are equal to their original incident direction. This holds true both for the 90° corners of a wall and for intersections where the wall and floor meet. These corner reflections help to provide insights into how volume levels often build up in the corners of a room (particularly at wall-to-floor corner intersections).

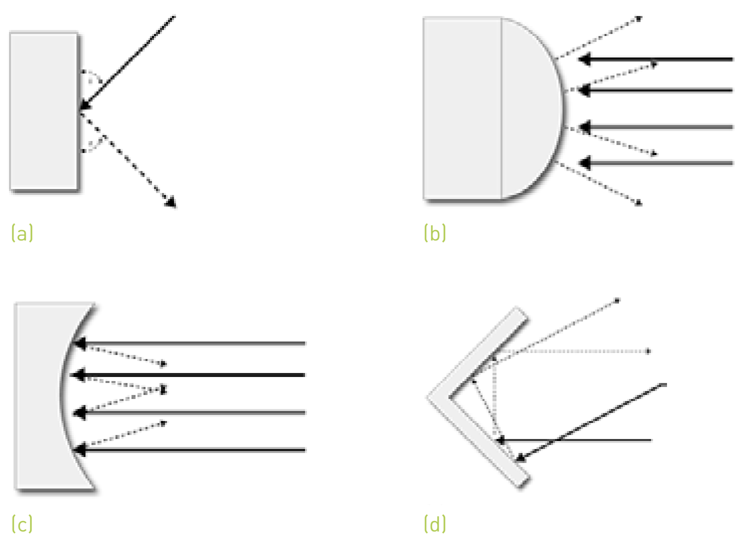


FIGURE 2.8
Incident sound waves striking surfaces with varying shapes: (a) single-planed, solid, smooth surface; (b) convex surface; (c) concave surface; (d) 90° corner reflection.

DIFFRACTION OF SOUND

Sound has the inherent ability to diffract around or through a physical acoustic barrier. In other words, sound can bend around an object in a manner that reconstructs the signal back to its original form in both frequency and amplitude.

For example, in **Figure 2.9a**, we can see how a small obstacle will scarcely impede a larger acoustic waveform. **Figure 2.9b** shows how a larger obstacle can obstruct a larger portion of the waveform; however, past the obstruction, the signal bends around the area in the barrier's wake and begins to reconstruct itself. **Figure 2.9c** shows how the signal is able to radiate through an opening in a large barrier.

Although the signal is greatly impeded (relative to the size of the opening), it nevertheless begins to reconstruct itself in wavelength and relative amplitude and begins to radiate outward as though it were a new point of origin. Finally, **Figure 2.9d** shows how a large opening in a barrier lets much of the waveform pass through relatively unimpeded.

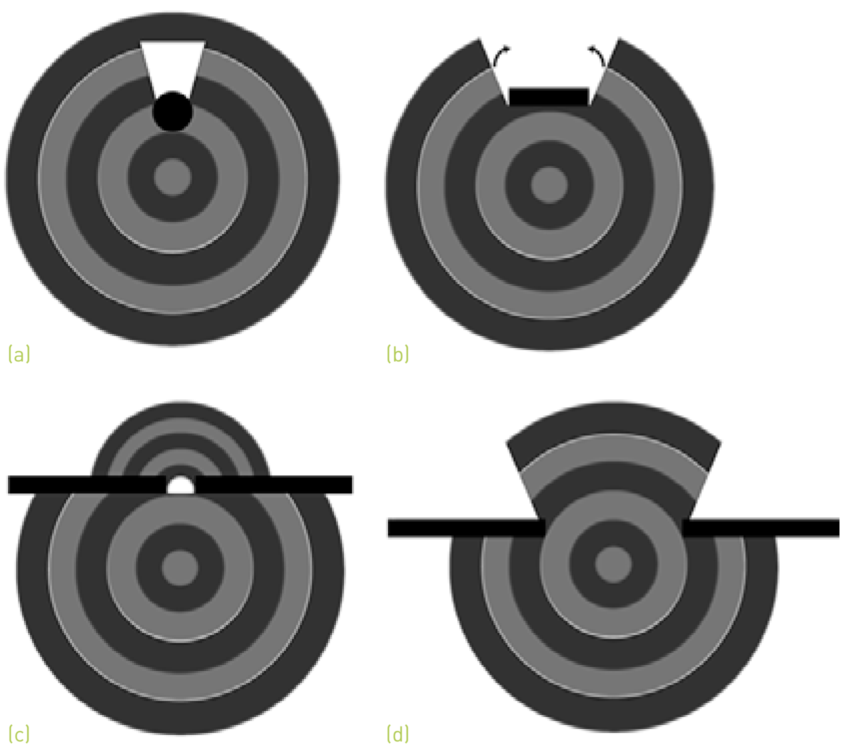


FIGURE 2.9
The effects of obstacles on sound radiation and diffraction. (a) A small obstacle will scarcely impede a longer wavelength signal. (b) A larger obstacle will obstruct the signal to a greater extent; the waveform will also reconstruct itself in the barrier's wake. (c) A small opening in a barrier will greatly impede a signal; the waveform will emanate from the opening and reconstruct itself as a new source point. (d) A larger opening allows sound to pass unimpeded, allowing it to quickly diffract back into its original shape.

FREQUENCY RESPONSE

The charted output of an audio device is known as its *frequency response* curve (when supplied with a reference input of equal level over the 20 to 20,000Hz range of human hearing). This curve is used to graphically represent how a device will respond to the audio spectrum and, thus, how it will affect a signal's overall sound. As an example, Figure 2.10 shows the frequency response of several unidentified devices. In these and all cases, the x-axis represents the signal's measured frequency, while the y-axis represents the device's measured output signal. These curves are created by feeding the input of an acoustic or electrical device with a constant-amplitude reference signal that sweeps over the entire frequency spectrum. The results are then charted on an amplitude vs. frequency graph that can be easily read at a glance. If the measured signal is the same level at all frequencies, the curve will be drawn as a flat, straight line from left to right (known as a *flat frequency* response curve). This indicates that the device passes all frequencies equally (with no frequency being emphasized or de-emphasized). If the output lowers or increases at certain frequencies, these changes will easily show up as dips or peaks in the chart.

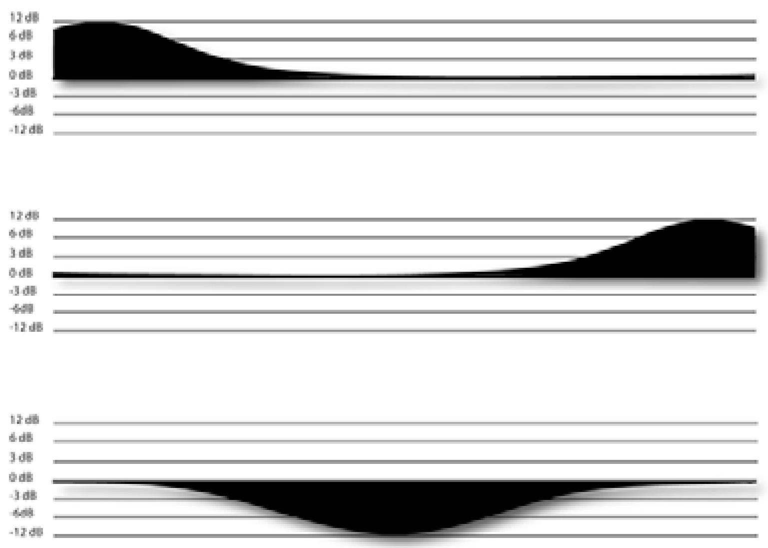


FIGURE 2.10
Frequency response curves: (a) curve showing a bass boost; (b) curve showing a boost at the upper end; (c) curve showing a dip in the midrange.

PHASE

Because we know that a cycle can begin at any point on a waveform, it follows that whenever two or more waveforms are involved in producing a sound, their relative amplitudes can (and most often will) be different at any one point in time. For simplicity's sake, let's limit our example to two pure tone waveforms (sine waves) that have equal amplitudes and frequency, but start their cyclic periods at different times. Such waveforms are said to be out of phase with respect to each other. Variations in phase, which are measured in degrees (°), can be described as a time delay between two or more waveforms. These delays are often said to have differences in relative phase degree angles (over the full rotation of a cycle, e.g., 90°, 180°, 270° or any angle between 0° and 360°). The sine wave (so named because its amplitude follows a trigonometric sine function) is usually considered to begin at 0° with an amplitude of zero; the waveform then increases to a positive maximum at 90°, decreases back to a zero amplitude at 180°, increases to a negative maximum value at 270°, and finally returns back to its original level at 360°, simply to begin all over again.

Whenever two or more waveforms arrive at a single location out of phase, their relative signal levels will be added together to create a combined amplitude level at that one point in time. Whenever two waveforms having the same frequency, shape and peak amplitude are completely in phase (meaning that they have no relative time difference), the newly combined waveform will have the same frequency, phase and shape, but will be double in amplitude (Figure 2.11a). If the same two waves are combined completely out of phase (having a phase difference of 180°), they will cancel each other out when added, which results in a relative value of zero amplitude (Figure 2.11b). If the second wave is only partially out of phase (by a degree other than 180°), the levels will be added at points where the combined amplitudes are positive and reduced in level where the combined result is negative (Figure 2.11c).

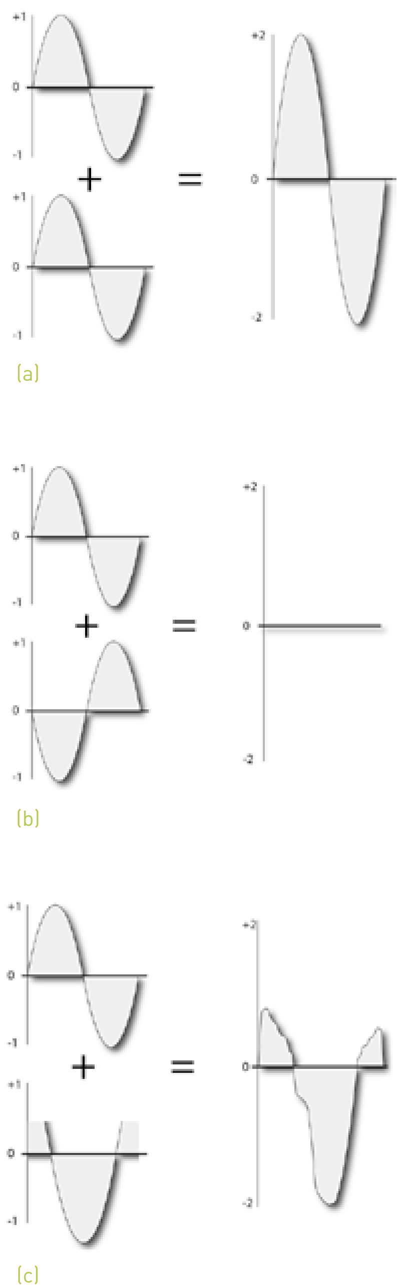


FIGURE 2.11
Combining sine waves of various phase relationships. (a) The amplitudes of in-phase waves increase in level when mixed together. (b) Waves of equal amplitude cancel completely when mixed 180º out of phase. (c) When partial phase angles are mixed, the combined signals will add in certain places and subtract in others.

PHASE SHIFT

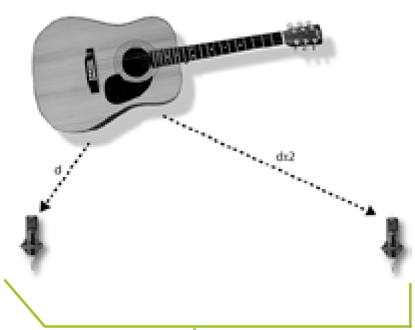


FIGURE 2.12
Cancellations can occur when a single source is picked up by two microphones.

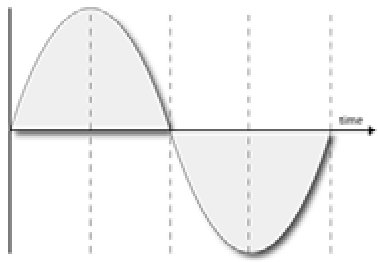
Phase shift is a term that describes one waveform's lead or lag in time with respect to another. Basically, it results from a time delay between two (or more) waveforms (with differences in acoustic distance being the most common source of this type of delay). For example, a 500 Hz wave completes one cycle every 0.002 sec. If you start with two in-phase, 500 Hz waves and delay one of them +1 by 0.001 sec (half the wave's period), the delayed wave will lag the other by one-half a cycle, or 180°. Another example might include a single source that's being picked up by two microphones that have been placed at different distances **(Figure 2.12)**, thereby creating a corresponding time delay when the mics are mixed together. Such a delay can also occur when a single microphone picks up direct sounds as well as those that are reflected off of a nearby boundary. These signals will be in phase at frequencies where the path-length difference is equal to the signal's wavelength, and out of phase at those frequencies where the multiples fall at or near the half-wavelength distance. In all the above situations, these boosts and cancellations combine to alter the signal's overall frequency response at the pickup. For this and other reasons, acoustic leakage between microphones and reflections from nearby boundaries should be kept to a minimum whenever possible.

TUTORIAL : PHASE

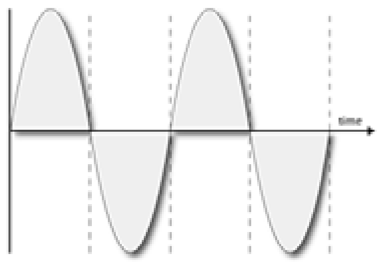
1. Go to the Tutorial section of www.modrec.com, click on Phase Tutorial and download the 0° and 180° soundfiles.
2. Load the 0° file onto track 1 of the digital audio workstation (DAW) of your choice, making sure to place the file at the beginning of the track, with the signal panned center.
3. Load the same 0° file again into track 2.
4. Load the 180° file into track 3.
5. Play tracks 1 and 2 (by muting track 3) and listen to the results. The result should be a summed signal that is 3 dB louder.
6. Play tracks 1 and 3 (by muting track 2) and listen to the results. It should cancel, producing no output.
7. Offsetting track 3 (relative to track 1) should produce varying degrees of cancellation.
8. Feel free to zoom in on the waveforms, mix them down, and view the results. Cool, huh?

HARMONIC CONTENT

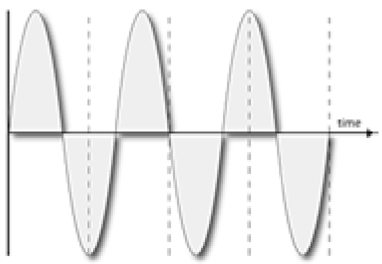
Up to this point, our discussion has centered on the sine wave, which is composed of a single frequency that produces a pure sound at a specific pitch. Fortunately, musical instruments rarely produce pure sine waves. If they did, all of the instruments would basically sound the same, and music would be pretty boring. The factor that helps us differentiate between instrumental "voicings" is the presence of frequencies (called partials) that exist in addition to the fundamental pitch that's being played. Partials that are higher than the fundamental frequency are called upper partials or overtones. Overtone frequencies that are whole-number multiples of the fundamental frequency are called harmonics. For example, the frequency that corresponds to concert A is 440 Hz **(Figure 2.13a)**. An 880 Hz wave is a harmonic of the 440-Hz fundamental because it is twice the frequency **(Figure 2.13b)**. In this case, the 440 Hz fundamental is technically the first harmonic because it is 1 times the fundamental frequency, and the 880 Hz wave is called the second harmonic because it is 2 times the fundamental. The third harmonic would be 3 times 440 Hz, or 1320 Hz **(Figure 2.13c)**. Some instruments, such as bells, xylophones and other percussion instruments, will often contain overtone partials that aren't harmonically related to the fundamental at all.



(a)



(b)

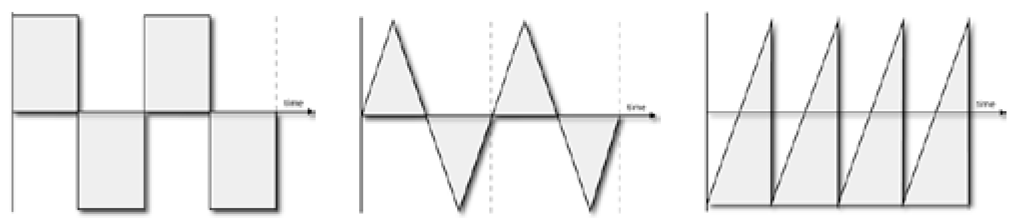


(c)

The ear perceives frequencies that are whole, doubled multiples of the fundamental as being related in a special way (a phenomenon known as the musical octave). For example, as concert A is 440 Hz (A3), the ear hears 880 Hz (A4) as being the next highest frequency that sounds most like concert A. The next related octave above that will be 1760 Hz (A5). Therefore, 880 Hz is said to be one octave above 440 Hz, and 1760 Hz is said to be two octaves above 440 Hz, etc. Because these frequencies are even multiples of the fundamental, they're known as even harmonics. Not surprisingly, frequencies that are odd multiples of the fundamental are called odd harmonics. In general, even harmonics are perceived as creating a sound that is pleasing to the ear, while odd harmonics will create a dissonant, harsher tone.

FIGURE 2.13
An illustration of harmonics: (a) first harmonic "fundamental waveform"; (b) second harmonic; (c) third harmonic.

Because musical instruments produce sound waves that contain harmonics with various amplitude and phase relationships, the resulting waveforms bear little resemblance to the shape of the single-frequency sine wave. Therefore, musical waveforms can be divided into two categories: simple and complex. Square waves, triangle waves and sawtooth waves are examples of simple waves that contain a consistent harmonic structure (Figure 2.14). They are said to be simple because they're continuous and repetitive in nature. One cycle of a square wave looks exactly like the next, and they are symmetrical about the zero line.



(a) (b) (c)

FIGURE 2.14
Simple waveforms: (a) square waves; (b) triangle waves; (c) sawtooth waves.

TUTORIAL : HARMONICS

1. Go to the Tutorial section of www.modrec.com, click on Harmonics Tutorial and download all of the soundfiles.
2. Load the first-harmonic A440 file onto track 1 of the digital audio workstation (DAW) of your choice, making sure to place the file at the beginning of the track, with the signal panned center.
3. Load the second-, third-, fourth- and fifth-harmonic files into the next set of consecutive tracks.
4. Solo the first-harmonic track, then solo the first- and second-harmonic tracks. Do they sound related in nature?
5. Solo the first-harmonic track, then solo the first- and third-harmonic tracks. Do they sound more dissonant?
6. Solo the first-, second- and third-harmonic tracks. Do they sound related?
7. Solo the first-, third- and fifth-harmonic tracks. Do they sound more dissonant?

FIGURE 2.15
Example of a complex waveform.

Complex waves, on the other hand, don't necessarily repeat and often are not symmetrical about the zero line. An example of a complex waveform **(Figure 2.15)** is one that's created by any naturally occurring sound (such as music or speech). Although complex waves are rarely repetitive in nature, all sounds can be mathematically broken down into a series of ever-changing combination of individual sine waves (or re-synthesized through a complex process known as Fourier analysis).

Regardless of the shape or complexity of a waveform that reaches the eardrum, the inner ear is able to perceive these component waveforms and then transmit the stimulus to the brain. This can be illustrated by passing a square wave through a bandpass filter that's set to pass only a narrow band of frequencies at any one time. Doing this would show that the square wave is composed of a fundamental frequency plus a number of harmonics that are made up of odd-number multiple frequencies (whose amplitudes decrease as the frequency increases). In **Figure 2.16**, we see how individual sine-wave harmonics can be combined to form a square wave.

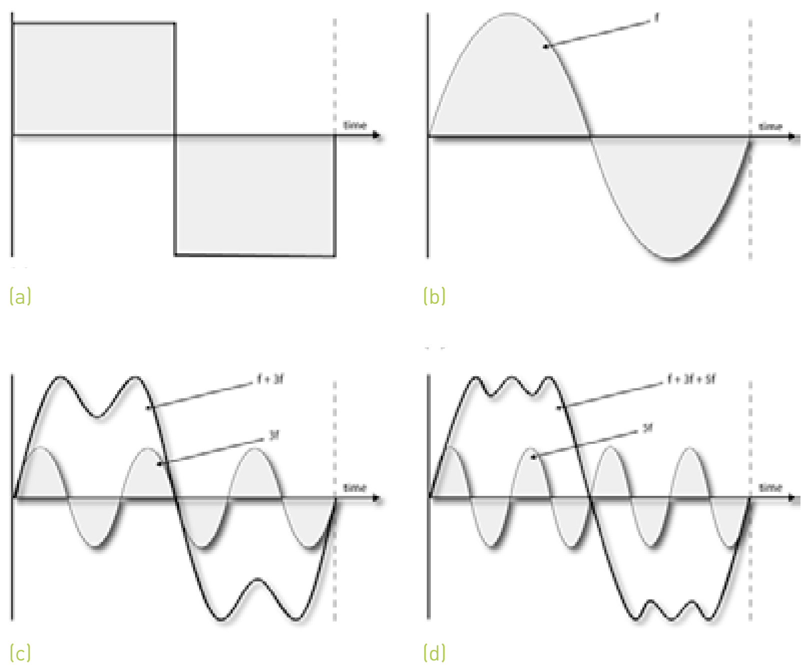
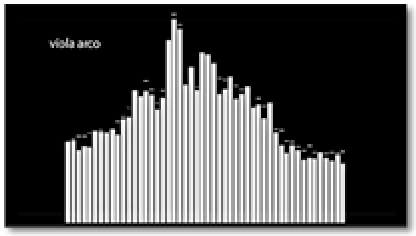
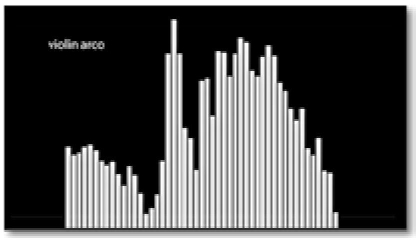


FIGURE 2.16
Breaking a square wave down into its odd-harmonic components: (a) square wave with frequency *f*; (b) sine wave with frequency *f*; (c) sum of a sine wave with frequency *f* and a lower amplitude sine wave of frequency 3*f*; (d) sum of a sine wave of frequency *f* and lower amplitude sine waves of 3*f* and 5*f*, which is beginning to resemble a square wave.



(a)



(b)

FIGURE 2.17
Harmonic structure of concert A440: (a) played on a viola; (b) played on a violin.

If we were to analyze the harmonic content of sound waves that are produced by a violin and compare them to the content of the waves that are produced by a viola (with both playing concert A440 Hz), we would come up with results like those shown in Figure 2.17. Notice that the violin's harmonics differ in both degree and intensity from those of the viola. The harmonics and their relative intensities (which determine an instrument's characteristic sound) are called the timbre of an instrument. If we changed an instrument's harmonic balance, the sonic character of the instrument would also be changed.

For example, if the violin's upper harmonics were reduced, the violin would sound more like a viola.

Because the relative harmonic balance is so important to an instrument's sound, the frequency response of a microphone, amplifier, speaker and all other elements in the signal path can have an effect on the timbral (tonal) balance of a sound. If the frequency response isn't flat, the timbre of the sound will be changed. For example, if the high frequencies are amplified less than the low and middle frequencies, then the sound will be duller than it should be. For this reason, a specific mic, equalizer or mic placement can be used as tools to vary the timbre of an instrument, thereby changing its subjective sound.

In addition to the variations in harmonic balance that can exist between instruments and their families, it is common for the harmonic balance to vary with respect to the direction that a sound wave radiates from an instrument. Figure 2.18 shows the principal radiation patterns as they emanate from a cello (as seen from both the side and top views).

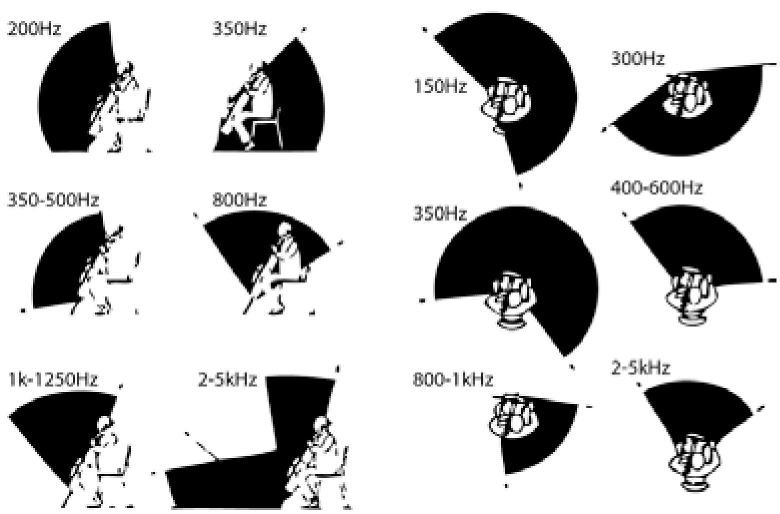


FIGURE 2.18
Radiation patterns of a cello as viewed from the side (left) and top (right).

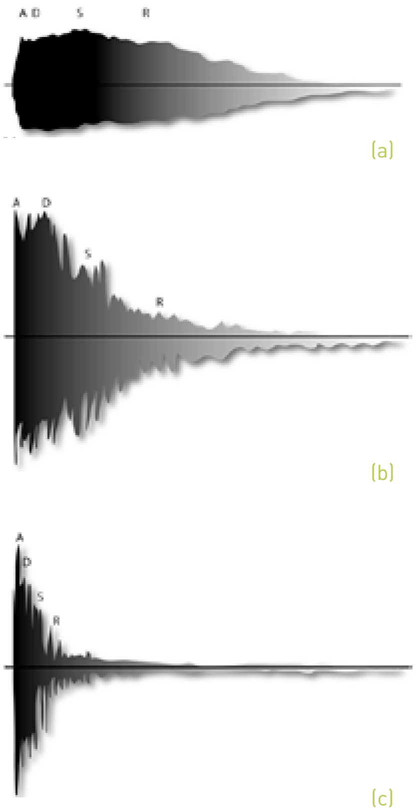


FIGURE 2.19
Various musical waveform envelopes: (a) trombone, (b) cymbal crash, and (c) snare drum, where A = attack, D = decay, S = sustain, and R = release.

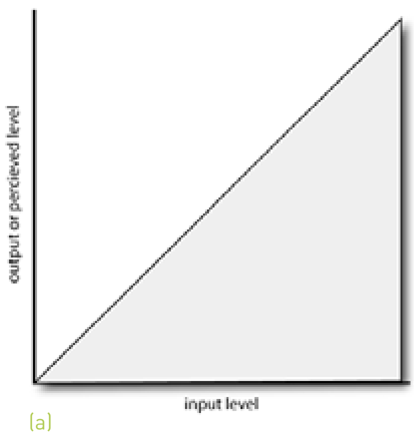
ENVELOPE

Timbre isn't the only characteristic that helps us differentiate between instruments. Each one produces a sonic amplitude *envelope* that works in combination with timbre to determine its unique and subjective sound. The envelope (ADSR) of a waveform can be described as characteristic variations in level that occur in time over the duration of a played note. The envelope of an acoustic or electronically generated signal is composed of four sections that vary in amplitude over time:

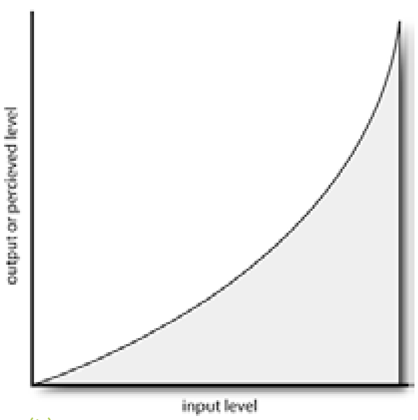
- *Attack* (A) refers to the time taken for a sound to build up to its full volume when a note is initially sounded.
- *Decay* (D) refers to how quickly the sound levels off to a sustain level after the initial attack peak.
- *Sustain* (S) refers to the duration of the ongoing sound that's generated following the initial attack decay.
- *Release* (R) relates to how quickly the sound will decay once the note is released.

Figure 2.19a illustrates the envelope of a trombone note. The attack, decay times and internal dynamics produce a smooth, sustaining sound. A cymbal crash **(Figure 2.19b)** combines a high-level, fast attack with a longer sustain and decay that creates a smooth, lingering shimmer. **Figure 2.19c** illustrates the envelope of a snare drum. Notice that the initial attack is much louder than the internal dynamics, while the final decay trails off very quickly, resulting in a sharp, percussive sound.

It's important to note that the concept of an envelope often relies on peak waveform values, while the human perception of loudness is proportional to the average wave intensity over a period of time (rms value). Therefore, high-amplitude portions of the envelope won't make an instrument sound loud unless the amplitude is maintained for a sustained period. Short high-amplitude sections tend to contribute to a sound's overall character, rather than to its loudness. By using a compressor or limiter, an instrument's character can often be modified by changing the dynamics of its envelope without changing its overall timbre.



(a)



(b)

FIGURE 2.20
Linear and logarithmic curves: (a) linear; (b) logarithmic.

LOUDNESS LEVELS: THE DECIBEL

The human ear operates over an energy range of approximately 10 10 :1 (10,000,000,000,000:1), which is an extremely wide range. Since it's difficult for us to conceptualize number ranges that are this large, a logarithmic scale has been adopted to compress the measurements into figures that are more manageable. The unit used for measuring sound-pressure level (SPL), signal level and relative changes in signal level is the *decibel* (*dB*), a term that literally means 1/10th of a Bell (an older telephone transmission loss measurement unit that was named after Alexander Graham Bell, inventor of the telephone). In order to develop an understanding of the decibel, we first need to examine logarithms and the logarithmic scale **(Figure 2.20)**. The *logarithm* (*log*) is a mathematical function that reduces large numeric values into smaller, more manageable numbers. Because logarithmic numbers increase exponentially in a way that's similar to how we perceive loudness (e.g., 1, 2, 4, 16, 128, 256, 65,536 ...), it expresses our perceived sense of volume more precisely than a linear curve can.

Before we delve into a deeper study of this important concept and how it deals with our perceptual senses, let's take a moment to understand the basic concepts and building block ideas behind the log scale, so as to get a better understanding of what examples such as "+3 dB at 10,000 Hz" really mean. Be patient with yourself! Over time, the concept of the decibel will become as much a part of your working vocabulary as ounces, gallons and miles per hour (or KG, liters and kilometers).

LOGARITHMIC BASICS

In audio, we use logarithmic values to express the differences in intensities between two levels (often, but not always, comparing a measured level to a standard reference level). Because the differences between these two levels can be really, really big, a simpler system makes use of expressed values that are mathematical exponents of 10. To begin, finding the log of a number such as 17,386 without a calculator is not only difficult ... it's unnecessary! All that's really important to help you along are three simple guidelines:

- The log of the number 2 is 0.3.
- When a number is an integral power of 10 (e.g., 100, 1000, 10,000), the log can be found simply by adding up the number of zeros.
- Numbers that are greater than 1 will have a positive log value, while those less than 1 will have a negative log value.

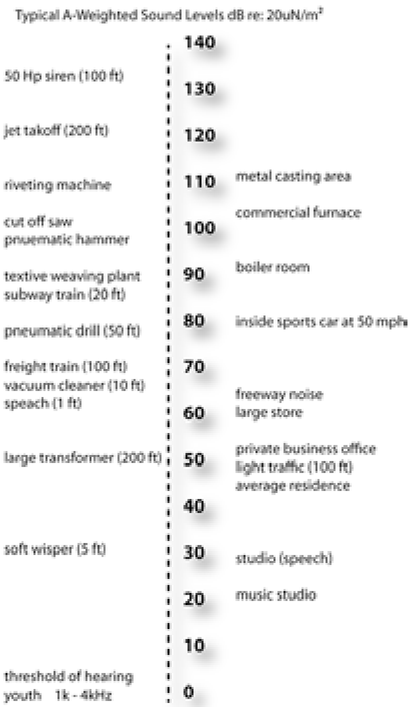


FIGURE 2.21
Chart of sound-pressure levels.
(Courtesy of General Radio Company.)

Again, the first one is an easy fact to remember: The log of 2 is 0.3 … this will make sense shortly. The second one is even easier: The logs of numbers such as 100, 1000 or 10,000,000,000,000 can be arrived at by simply counting up the zeros. The last guideline relates to the fact that if the measured value is less than the reference value, the resulting log value will be negative. For example:

log 2 = 0.3

log ½ = log 0.5 = –0.3

log 10,000,000,000,000 = 13

log 1000 = 3

log 100 = 2

log 10 = 1

log 0.1 = –1

log 0.01 = –2

log 0.01 = –3

All other numbers can be arrived at by using a scientific calculator (most computers and cell phones have one built in); however, it's unlikely that you will ever need to know any log values beyond understanding the basic concepts that are listed above.

THE DECIBEL

Now that we've gotten past the absolute bare basics, I'd like to break with tradition again and attempt an explanation of the decibel in a way that's less complex and relates more to our day-to-day needs in the sound biz. First off, the decibel is a logarithmic value that "expresses differences in intensities between two levels." From this, we can infer that these levels are expressed by several units of measure, the most common being sound-pressure level (SPL), voltage (V) and power (wattage, or W). Now, let's look at the basic math behind these three measurements.

SOUND-PRESSURE LEVEL

Sound-pressure level is the acoustic pressure that's built up within a defined atmospheric area (usually a square centimeter, or cm^2). Quite simply, the higher the SPL, the louder the sound (Figure 2.21). In this instance, our measured reference (SPL_{ref}) is the threshold of hearing, which is defined as being the softest sound that an average person can hear. Most conversations will have an SPL of about 70 dB, while average home stereos are played at volumes ranging between 80 and 90 dB

SPL. Sounds that are so loud as to be painful have SPLs of about Ref 130 to 140 dB (10,000,000,000,000 or more times louder than the 0 dB reference). We can arrive at an SPL rating by using the formula:

$$\text{dB SPL} = 20\log \text{SPL}/\text{SPL}_{ref}$$

where SPL is the measured sound pressure (in dyne/cm^2). SPL_{ref} is a reference sound pressure (the threshold limit of human hearing, 0.02 millipascals = 2 ten-billionths of our atmospheric pressure).

From this, I feel that the major concept that needs to be understood is the idea that SPL levels change with the square of the distance (hence, the 20 log part of the equation). This means that whenever a source/pickup distance is doubled, the SPL level will be reduced by 6 dB (20 log 1/2 = 20 x -0.3 = −6 dB SPL); as the distance is halved, it will increase by 6 dB (20 log 2/1 = 20 x 0.3 = 6 dB SPL), as shown in Figure 2.22.

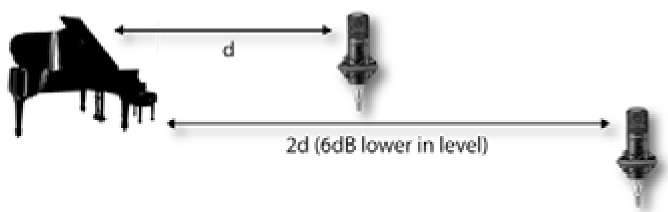


FIGURE 2.22
Doubling the distance of a pickup will lower the perceived direct signal level by 6 dB SPL.

VOLTAGE

Voltage can be thought of as the pressure behind electrons within a wire. As with acoustic energy, comparing one voltage level to another level (or reference level) can be expressed as dBv using the equation:

$$\text{dBv} = 20\log V/V_{ref}$$

where V is the measured voltage, and V_{ref} is a reference voltage (0.775 volts).

POWER

Power is usually a measure of wattage or current and can be thought of as the flow of electrons through a wire over time. Power is generally associated with audio signals that are carried throughout an audio production system. Unlike SPL and voltage, the equation for signal level (which is often expressed in dBm) is:

$$\text{dBm} = 10\log P/P_{ref}$$

where P is the measured wattage, and P_{ref} is referenced to 1 milliwatt (0.001 watt).

THE SIMPLE HEART OF THE MATTER

I am going to stick my neck out and state that, when dealing with decibels, it's far more common for working professionals to deal with the concept of power. The dBm equation expresses the spirit of the decibel term when dealing with the marking and measurements on an audio device or the numeric values in a computer dialog box. This is due to the fact that power is the unit of measure that's most often expressed when dealing with audio equipment controls; therefore, it's my personal opinion that the average working stiff only needs to grasp the following basic concepts:

- A 1 dB change is noticeable to most ears (but not by much).
- Turning something up by 3 dB will double the signal's level (believe it or not, doubling the signal level won't increase the perceived loudness as much as you might think).

  Turning something down by 3 dB will halve the signal's level (likewise, halving the signal level won't decrease the perceived loudness as much as you might think).
- The log of an exponent of 10 can be easily figured by simply counting the zeros (e.g., the log of 1,000 is 3). Given that this figure is multiplied by 10 (10 log P/P), turning something up by 10 dB will increase the signal's level 10-fold, 20 dB will yield a 100-fold increase, 30 dB will yield a 1,000-fold increase, etc.

Most pros know that turning a level fader up by 3 dB will effectively double its energy output (and vice versa). Beyond this, it's unlikely that anyone will ever ask, "Would you please turn that up a thousand times?" It just won't happen! However, when a pro asks his or her assistant to turn the gain up by 20 dB, that assistant will often instinctively know what 20 dB is... and what it sounds like. I guess I'm saying that the math really isn't nearly as important as the ongoing process of getting an instinctive feel for the decibel and how it relates to relative levels within audio production.

THE EAR

A sound source produces acoustic waves by alternately compressing and rarefying the air molecules between it and the listener, causing fluctuations that fall above and below normal atmospheric pressure. The human ear is a sensitive transducer that responds to these pressure variations by way of a series of related processes that occur within the auditory organs ... our ears. When these variations arrive at the listener, sound-pressure waves are collected in the aural canal by way of the outer ear's pinna. These are then directed to the eardrum, a stretched drum-like membrane **(Figure 2.23)**, where the sound waves are changed into mechanical vibrations, which are transferred to the inner ear by way of three bones known as the hammer, anvil and stirrup. These bones act both as an amplifier (by significantly increasing the vibrations that are transmitted from the eardrum) and as a limiting protection device (by reducing the level of loud, transient sounds such as thunder or fireworks explosions). The vibrations are then applied to the inner ear (cochlea) – a tubular, snail-like organ that contains two fluid-filled chambers. Within these chambers are tiny hair receptors that are lined up in a row along the length of the cochlea. These hairs respond to certain frequencies depending on their placement along the organ, which results in the neural stimulation that gives us the sensation of hearing. Permanent hearing loss generally occurs when these hair/nerve combinations are damaged or as they deteriorate with age.

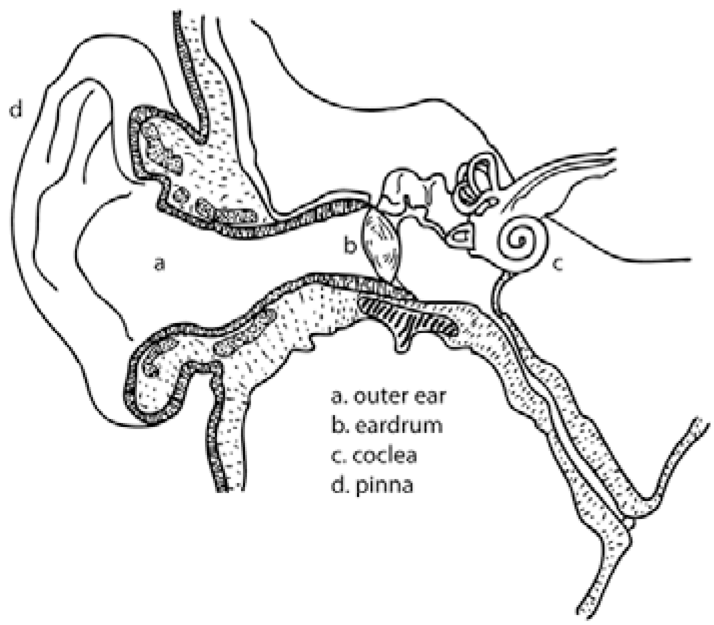


FIGURE 2.23
Outer, middle and inner ear.

THRESHOLD OF HEARING

In the case of SPL, a convenient pressure-level reference is the threshold of hearing, which is the minimum sound pressure that produces the phenomenon of hearing in most people and is equal to 0.0002 microbar. One microbar is equal to 1 millionth of normal atmospheric pressure, so you can see that the ear is an amazingly sensitive instrument. In fact, if the ear were any more sensitive, the thermal motion of molecules in the air would be audible! When referencing SPLs to 0.0002 microbar, this threshold level usually is denoted as 0 dB SPL, which is defined as the level that an average person can hear at a specific frequency only 50% of the time.

THRESHOLD OF FEELING

An SPL that causes discomfort in a listener 50% of the time is called the *threshold of feeling*. It occurs at a level of about 118 dB SPL between the frequencies of 200 Hz and 10 kHz.

THRESHOLD OF PAIN

The SPL that causes pain in a listener 50% of the time is called the *threshold of pain* and corresponds to an SPL of 140 dB in the frequency range between 200 Hz and 10 kHz.

TAKING CARE OF YOUR HEARING

During the 1970s and early 1980s, recording studio monitoring levels were often turned up so high as to be truly painful. In the mid-1990s, a small band of powerful producers and record executives banded together to successfully reduce these average volumes down to tolerable levels (85 to 95 dB) ... a practice that often continues to this day. Live sound venues and acts often continue the practice of raising house and stage volumes to chest-thumping levels. Although these levels are exciting, long-term exposure can lead to temporary or permanent hearing loss.

So what types of hearing loss are there?

- *Acoustic trauma*: This happens when the ear is exposed to a sudden, loud noise in excess of 140 dB. Such a shock could lead to permanent hearing loss.
- *Temporary threshold shift*: The ear can experience temporary hearing loss when exposed to long-term, loud noise.

A simple fact to remember: Once your hearing is gone...it's gone...for good!

- *Permanent threshold shift*: Extended exposure to loud noises in a specific or broad hearing range can lead to permanent hearing loss in that range. In short, the ear becomes less sensitive to sounds in the damaged frequency range leading to a reduction in perceived volume... What?

Here are a few hearing conservation tips (courtesy of the House Ear Institute, www.hei.org) that can help reduce hearing loss due to long-term exposure of sounds over 115 dB:

- Avoid hazardous sound environments; if they're not avoidable, wear hearing protection devices, such as foam earplugs, custom-molded earplugs, or in-ear monitors.
- Monitor sound-pressure levels at or around 85 dB. The general rule to follow is if you're in an environment where you must raise your voice to be heard, then you're monitoring too loudly and should limit your exposure times.
- Take 15-minute "quiet breaks" every few hours if you're being exposed to levels above 85 dB.
- Musicians and other live entertainment professionals should avoid practicing at concert-hall levels whenever possible.
- Have your hearing checked periodically by a licensed audiologist.

PSYCHOACOUSTICS

The area of *psychoacoustics* deals with how and why the brain interprets a particular sound stimulus in a certain way. Although a great deal of study has been devoted to this subject, the primary device in psychoacoustics is the all-elusive brain ... which is still largely unknown to present-day science.

AUDITORY PERCEPTION

From the outset, it's important to realize that the ear is a nonlinear device (what's received at your ears isn't always what you'll hear). It's also important to note that the ear's frequency response (its perception of timbre) changes with the loudness of the perceived signal. The "loudness" compensation switch found on many hi-fi preamplifiers is an attempt to compensate for this decrease in the ear's sensitivity to low- and high-frequency sounds at low listening levels.

The *Fletcher-Munson equal-loudness contour curves* **(Figure 2.24)** indicate the ear's average sensitivity to different frequencies at various levels. These indicate the sound-pressure levels that are required for our ears to hear frequencies along the curve as being equal in level to a 1000-Hz reference level (measured in phons). Thus, to equal the loudness of a 1-kHz tone at 110 dB SPL (a level typically created by a trumpet-type car horn at a distance of 3 feet), a 40-Hz tone has to be about 6 dB louder, whereas a 10-kHz tone must be 4 dB louder in order to be perceived as being equally loud. At 50 dB SPL (the noise level present in the average private business office), the level of a 40-Hz tone must be 30 dB louder and a 10-kHz tone 13 dB louder than a 1-kHz tone to be perceived as having the same volume. Thus, if a piece of music is mixed to sound great at a level of 85 to 95 dB, its bass and treble balance will actually be boosted when turned up (often a good thing). If the same piece were mixed at 110 dB SPL, it would sound both bass and treble shy when played back at lower levels … because no compensation for the ear's response was added to the mix. Over the years, it has generally been found that changes in apparent frequency balance are less apparent when monitoring at levels of 85 dB SPL.

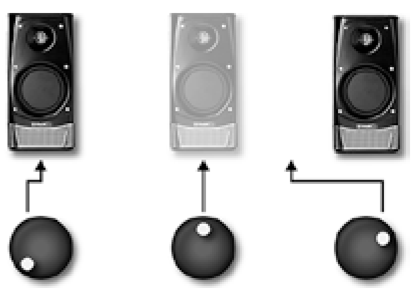


FIGURE 2.28
Pan pot settings and their relative spatial positions.

In addition to the above, whenever it is subjected to sound waves that are above a certain loudness level, the ear can produce harmonic distortion that doesn't exist in the original signal. For example, the ear can cause a loud 1-kHz sine wave to be perceived as being a combination of 1-, 2-, 3-kHz waves, and so on. Although the ear might hear the overtone structure of a violin (if the listening level is loud enough), it might also perceive additional harmonics (thus changing the timbre of the instrument). This is one of several factors that implies that sound monitored at very loud levels could sound quite different when played back at lower levels.

The loudness of a tone can also affect our ear's perception of pitch. For example, if the intensity of a 100-Hz tone is increased from 40 to 100 dB SPL, the ear will hear a pitch decrease of about 10%. At 500 Hz, the pitch will change about 2% for the same increase in sound-pressure level. This is one reason why musicians find it difficult to tune their instruments when listening through loud headphones. As a result of the nonlinearities in the ear's response, tones will often interact with each other rather than being perceived as being separate.

MASKING

Masking is the phenomenon by which loud signals prevent the ear from hearing softer sounds. The greatest masking effect occurs when the frequency of the sound and the frequency of the masking noise are close to each other. For example, a 4k-Hz tone will mask a softer 3.5-kHz tone but has little effect on the audibility of a quiet

1000-Hz tone. Masking can also be caused by harmonics of the masking tone (e.g., a 1-kHz tone with a strong 2-kHz harmonic might mask a 1900-Hz tone). This phenomenon is one of the main reasons why stereo placement and equalization are so important to the mixdown process. An instrument that sounds fine by itself can be completely hidden or changed in character by louder instruments that have a similar timbre. Equalization, mic choice or mic placement might have to be altered to make the instruments sound different enough to overcome any masking effect.

TUTORIAL : MASKING

1. Go to the Tutorial section of www.modrec.com, click on Masking Tutorial and download all of the soundfiles.
2. Load the 1000 Hz file onto track 1 of the digital audio workstation (DAW) of your choice, making sure to place the file at the beginning of the track, with the signal panned center.
3. Load the 3800- and 4000-Hz files into the next two consecutive tracks.
4. Solo and play the 1000-Hz tone.
5. Solo both the 1000- and the 4000-Hz tones and listen to their combined results. Can you hear both of the tones clearly?
6. Solo and play the 3800-Hz tone.
7. Solo both the 3800- and the 4000-Hz tones and listen to their combined results. Can you hear both of the tones clearly?]

PERCEPTION OF SPACE

In addition to perceiving the direction of sound, the ear and brain combine to help us perceive the size and physical characteristics of the acoustic space in which a sound occurs. When a sound is generated, a percentage reaches the listener directly, without encountering any obstacles.

A larger portion, however, is propagated to the many surfaces of an acoustic enclosure. If these surfaces are reflective, the sound is bounced back into the room and toward the listener. If the surfaces are absorptive, less energy will be reflected back to the listener. Three types of reflections are commonly generated within an

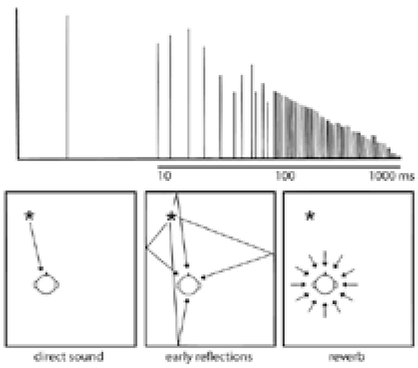


FIGURE 2.29
The three distinct soundfield types that are generated within an enclosed space.

enclosed space **(Figure 2.29)**:

- Direct sound
- Early reflections
- Reverberation

DIRECT SOUND

In air, sound travels at a constant speed of about 1130 feet per second, so a wave that travels from the source to the listener will follow the shortest path and arrive at the listener's ear first. This is called the *direct sound*. Direct sounds determine our perception of a sound source's location and size and conveys the true timbre of the source.

EARLY REFLECTIONS

Waves that bounce off of surrounding surfaces in a room must travel further than direct sound to reach the listener and therefore arrive after the direct sound and from a multitude of directions. These waves form what are called *early reflections*. Early reflections give us clues as to the reflectivity, size and general nature of an acoustic space. These sounds generally arrive at the ears less than 50 msec after the brain perceives the direct sound and are the result of reflections off of the largest, most prominent boundaries within a room. The time elapsed between hearing the direct sound and the beginning of the early reflections helps to provide information about the size of the performance room. Basically, the farther the boundaries are from the source and listener, the longer the delay before it's reflected back to the listener.

Another aspect that occurs with early reflections is called *temporal fusion*. Early reflections arriving at the listener within 30 msec of the direct sound are not only audibly suppressed, but are also fused with the direct sound. In effect, the ear can't distinguish the closely occurring reflections and considers them to be part of the direct sound. The 30-msec time limit for temporal fusion isn't absolute; rather, it depends on the sound's envelope. Fusion breaks down at 4 msec for transient clicks, whereas it can extend beyond 80 msec for slowly evolving sounds (such as a sustained organ note or legato violin passage). Despite the fact that the early reflections are suppressed and fused with the direct sound, they still modify our perception of the sound, making it both louder and fuller.

REVERBERATION

Whenever room reflections continue to bounce off of room boundaries, a randomly decaying set of sounds can often be heard after the source stops in the form of *reverberation*. A highly reflective surface absorbs less of the wave energy at each reflection and allows the sound to persist longer after the initial sound stops (and vice versa). Sounds reaching the listener 50-msec later in time are perceived as a random and continuous stream of reflections that arrive from all directions. These densely spaced reflections gradually decrease in amplitude and add a sense of warmth and body to a sound. Because it has undergone multiple reflections, the timbre of the reverberation is often quite different from the direct sound (with the most notable difference being a roll-off of high frequencies and a slight bass emphasis).

The time it takes for a reverberant sound to decrease to 60 dB below its original level is called its *decay time* or reverb time and is determined by the room's absorption characteristics. The brain is able to perceive the reverb time and timbre of the reverberation and uses this information to form an opinion on the hardness or softness of the surrounding surfaces. The loudness of the perceived direct sound increases rapidly as the listener moves closer to the source, while the reverberation levels will often remain the same, because the diffusion is roughly constant throughout the room. This ratio of the direct sound's loudness to the reflected sound's level helps listeners judge their distance from the sound source.

Whenever artificial reverb and delay units are used, the engineer can generate the necessary cues to convince the brain that a sound was recorded in a huge, stone-walled cathedral – when, in fact, it was recorded in a small, absorptive room. To do this, the engineer programs the device to mix the original unreverberated signal with the necessary early delays and random reflections. Adjusting the number and amount of delays on an effects processor gives the engineer control over all of the necessary parameters to determine the perceived room size, while decay time and frequency balance can help to determine the room's perceived surfaces. By changing the proportional mix of direct-to-processed sound, the engineer/producer can place the sound source at either the front or rear of the artificially created space.

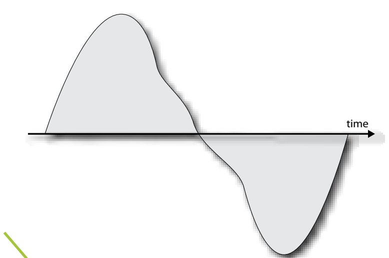


FIGURE 6.2
An analog signal is continuous in nature.

ANALOG & DIGITAL AUDIO BASICS

In the world of analog audio, signals are recorded, stored and reproduced as changes in voltage levels that continuously vary over time in a continuous fashion **(Figure 6.2)**. The digital recording process, on the other hand, doesn't operate in this manner; rather, digital recording operates by taking periodic samples of an analog audio

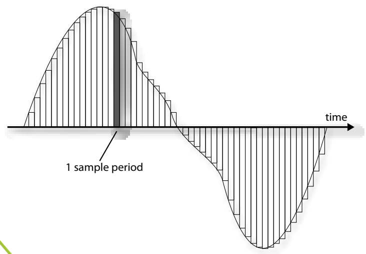


FIGURE 6.3
A digital signal makes use of periodic sampling to encode information.

waveform over time **(Figure 6.3)**, and then calculating each of these snapshot samples into grouped binary words that digitally represent these voltage levels as they change over time, as accurately as possible.

During this process, an incoming analog signal is sampled at discrete and precisely timed intervals (as determined by the sample rate). At each interval, this analog signal is momentarily "held" (frozen in time), while the converter goes about the process of determining what the voltage level actually is, with a degree of accuracy that's defined by the converter's circuitry and the chosen bit rate. The converter then generates a binary-encoded word that's numerically equivalent to the analog voltage level at that point in time **(Figure 6.4)**. Once this is done, the converter can store the representative word into a memory medium (tape, disk, disc, etc.), release its hold, and then go about the task of determining the values of the next sampled voltage. The process is then continuously repeated throughout the recording process.

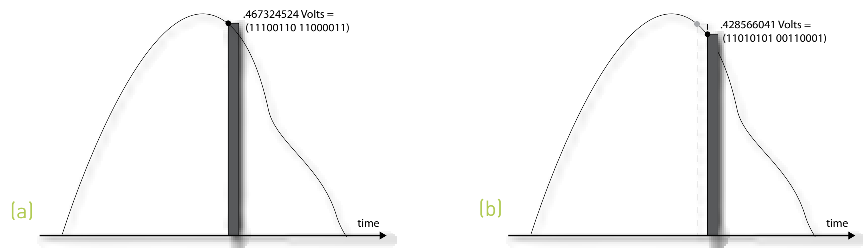


FIGURE 6.4
The sampling process. (a) The analog signal is momentarily "held" (frozen in time), while the converter goes about the process of determining the voltage level at that point in time and then converting that level into a binary-encoded word that's numerically equivalent to the original analog voltage level. (b) Once this digital information is processed and stored, the sample is released and the next sample is held, as the system again goes about the task of determining the level of the next sampled voltage ... and so forth, and so forth, and so forth over the duration of the recording.

Within a digital audio system, the sampling rate is defined as the number of measurements (samples) that are periodically taken over the course of a second. Its reciprocal (sampling time) is the elapsed time that occurs between each sampling period. For example, a sample rate of 44.1 kHz corresponds to a sample time of 1/44,100 of a second.

This process can be likened to a photographer who takes a series of action sequence shots. As the number of pictures taken in a second increases, the accuracy of the captured event will likewise increase until the resolution is so great that you can't tell that the successive, discrete pictures have turned into a continuous and (hopefully) compelling movie. Since the process of sampling is tied directly to the component of time, the sampling rate of a system determines its overall bandwidth **(Figure 6.5)**, meaning that a recording made at a higher sample rate will be capable of storing a wider range of frequencies (effectively increasing the signal's bandwidth at its upper limit).

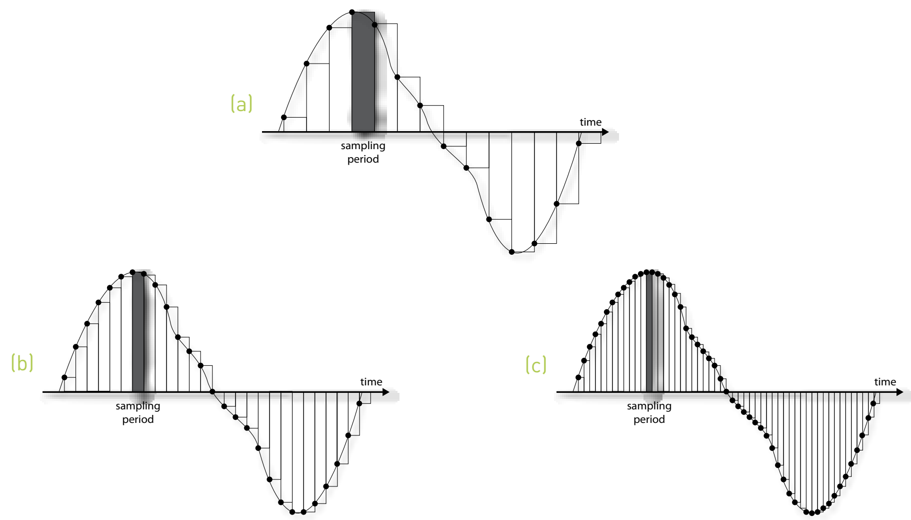


FIGURE 6.5

Discrete time sampling. (a) Whenever the sample rate is set too low, important data between sample periods will be lost. (b) As the rate is increased, more frequency-related data can be encoded.
(c) Increasing the sampling frequency further can encode the recorded signal with an even higher bandwidth range.

SOUND FILE SAMPLE RATES

The *sample rate* of a recorded digital audio bitstream directly relates to the resolution at which a recorded sound will be digitally captured. Using the film analogy, if you capture more samples (frames) of a moving image as it moves through time, you'll have a more accurate representation of that recorded event. If the number of samples are too low, the resolution will be "lossy" and will distort the event. On the other hand, taking too many picture frames might result in a recorded bandwidth that's so high that the audience won't be able to discriminate any advantages that the extra information has to offer … or the storage requirements will become increasingly large as the bandwidth and file size increase.

This analogy relates perfectly to audio because the choices of sample rate will be determined by the bandwidth (number of overall frequencies that are to be captured) versus the amount of storage needed to either save the data to a memory storage media … or the time that will be required to up/download a file through a transmission and/or online datastream. Beyond the basic adherence to certain industry sample rate standards, such are the personal decisions that must be made regarding the choice of sample rate to be used on a project. Although other sample-rate standards exist, the following are the most commonly used in the professional, project and audio production community:

- *32k*: This rate is often used by broadcasters to transmit/receive digital data via satellite. With its overall 15-kHz bandwidth and reduced data requirements, it is also used by certain devices in order to conserve on memory and is commonly used in satellite broadcast communications. Although the pro community doesn't generally use this rate, it's surprising just how good a sound can be captured at 32k (given a high-quality converter).

- *44.1k*: The long-time standard of consumer and pro audio production, 44.1 is the chosen rate of the CD-audio standard. With its overall 20-kHz bandwidth, the 44.1k rate is generally considered to be the minimum sample rate for professional audio production. Assuming that high-quality converters are used, this rate is capable of recording lossless audio, while conserving on memory storage requirements.
- *48k*: This standard was adopted early on as a standard sample rate for professional audio applications (particularly when referring to hardware digital audio devices). It's also the adopted standard rate for use within professional video and DVD production.
- *88.2k*: As a simple multiple of 44.1, this rate is often used within productions that are intended to be high-resolution products.
- *96k*: This rate has been adopted as the de facto sample rate for high-resolution recordings.
- *192k*: This high-resolution rate is uncommon within pro audio production, as the storage and media requirements are quite high.

SOUND FILE BIT RATES

The *bit rate* of a digitally recorded sound file directly relates to the number of quantization steps that are encoded into the bitstream. As a result, the bit rate (or bit depth) is directly correlated to the:

- Accuracy at which a sampled level (at one point in time) is to be encoded
- Signal-to-error figure … and thus the overall dynamic range of the recorded signal.

If the bit rate is too low to accurately encode the sample, the resolution will lead to quantization errors, which will lead to distortion. On the other hand, too high of a bit depth might result in a resolution that's so high that the resulting gain in resolution is lost on the audience's ability to discriminate it … or the storage requirements might become so high that the files become inordinately large.

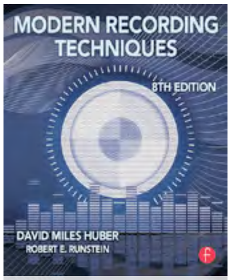
Although other sound file bit rate standards exist, the following are the most commonly used within the pro, project and general audio production community:

- *16 bits*: The long-time standard of consumer and professional audio production, 16 bits is the chosen bit depth of the CD-audio standard (offering a theoretical dynamic range of 97.8 dB). It is generally considered to be the minimum depth for high-quality professional audio production. Assuming that high-quality converters are used, this rate is capable of lossless audio recording, while conserving on memory storage requirements.
- *20 bits*: Before the 24-bit rate came onto the scene, 20 bits was considered to be the standard for high-bit-depth resolution. Although it's used less commonly, it can still be found in high-definition audio recordings (offering a theoretical dynamic range of 121.8 dB).
- *24 bits*: Offering a theoretical dynamic range of 145.8 dB, this standard bit rate is often used in high-definition audio applications, often in conjunction with the 96k sample rate (i.e., 96/24).

REGARDING DIGITAL AUDIO LEVELS

Over the past few decades, the trend toward making recordings that are as loud as possible has totally pervaded the industry to the point that it has be given the name of "The Loudness War." Not only is this practice used in mastering to make a song or project stand out in an on-air or in-the-pocket playlist ... it has also followed in the analog tradition of recording a track as hot as possible to get the best noise figures and punch. All of this is arguably well and good, except for the fact that in digital recording a track at too "hot" a level doesn't add extra punch—it just adds really nasty distortion.

Average or peak levels above full scale can easily ruin a recording. As such, since digital has a wider dynamic range than analog, it's always a good idea to reduce your levels so that they peak from -12 to -20 dB. This will accurately capture the peaks without clipping, without introducing an appreciable amount of noise into the mix. Recording at higher bit rates (i.e., 24 bits) will further reduce noise levels, allowing for increased headroom when recording at reduced levels. Of course, there are no standard guidelines or reference levels (digital meters aren't even standardized, for the most part) ... so you might want to further research the subject on your own.



The following is excerpted from *Modern Recording Techniques* by David Miles Huber and Robert E. Runstein.

You can purchase a copy HERE.

This chapter will discuss many of the basic acoustic principles and construction techniques that should be considered in the design of a music or sound production facility. I'd like to emphasize that any or all of these acoustical topics can be applied to any type of audio production facility and aren't only limited to professional music studio designs. For example, owners of modest project and bedroom studios should know the importance of designing a control room that's symmetrical and, hopefully, sounds good. It doesn't cost anything to know that if one speaker is in a corner and the other is on a wall, the perceived center image will be off balance. As with many techno-artistic endeavors, studio acoustics and design are a mixture of fundamental physics (in this case, mostly dimensional mathematics) and an equally large dose of common sense and dumb luck. More often than not, acoustics is an artistic science that melds physics with the art of intuition and experience.

THE PROJECT STUDIO

It goes without saying that the vast majority of audio production studios fall into the project studio category. This basic definition of such a facility is open to interpretation. It's usually intended as a personal production resource for recording music, audio-for-visual production, multimedia production, voiceovers ... you name it. Project studios can range from being fully commercial in nature to smaller setups that are both personal and private **(Figure 3.4)**. All of these possible studio types have been designed with the idea of giving artists the flexibility of making their art in a personal, off-the-clock environment that's both cost and time effective. The design and construction considerations for creating a privately owned project studio often differ from the design considerations for a professional music facility in two fundamental ways:

- Building constraints
- Cost

FIGURE 3.4
Gettin' it all going in the bedroom studio.
(Courtesy of Yamaha Corporation of America, www.yamaha.com)

THE RECORDING SPACE

Generally, a project studio's room (or series of rooms) is built into an artist's home or a rented space where the construction and dimensional details are already defined. This fact (combined with inherent cost considerations) often leads the owner/artist to employ cost-effective techniques for sonically treating a room. Even if the room has little or no treatment, keep in mind that a basic knowledge of acoustical physics and room design can be a handy and cost-effective tool as your experience, production needs and business abilities grow.

Modern-day digital audio workstations (DAWs) have squarely placed the Mac and PC within the ergonomics and functionality of the pro and home project studio (Figure 3.5). In fact, in many cases, the DAW is the project studio. With the advent of self-powered speaker monitors, cost-effective microphones and hardware DAW controllers, it's a relatively simple matter to design a powerful production system into almost any existing space.

FIGURE 3.5
785 Records & Publishing/
Denise Rich Songs, New York.
(Courtesy of Solid State Logic, www.solid-state-logic.com)

With regard to setting up any production/monitoring environment, I'd like to first draw your attention to the need for symmetry in any critical monitoring environment. A symmetrical acoustic environment around the central mixing axis can work wonders toward creating a balanced left/right and surround image. Fortunately, this generally isn't a difficult goal to achieve. An acoustical and speaker placement environment that isn't balanced between the left-hand and right-hand sides will allow for differing reflections, absorption coefficients and variations in frequency response that can adversely affect the imaging and balance of your final mix. Further information on this important subject can be found later in this chapter … consider this your first heads-up on an important topic.

PRIMARY FACTORS GOVERNING STUDIO AND CONTROL ROOM ACOUSTICS

Regardless of which type of studio facility is being designed, built and used, a number of primary concerns should be addressed in order to achieve the best possible acoustic results. In this section, we'll take a close look at such important and relevant aspects of acoustics as:

- Acoustic isolation
- Symmetry in control room and monitoring design
- Frequency balance
- Absorption
- Reflection
- Reverberation

Although several mathematical formulas have been included in the following sections, it's by no means necessary that you memorize or worry about them. By far, I feel that it's more important that you grasp the basic principles of acoustics rather than worry about the underlying math. Remember: More often than not, acoustics is an artistic science that blends math with the art of intuition and experience.

ACOUSTIC ISOLATION

Because most commercial and project studio environments make use of an acoustic space to record sound, it's often wise and necessary to employ effective isolation techniques into their design in order to keep external noises to a minimum. Whether that noise is transmitted through the medium of air (e.g., from nearby auto, train, or jet traffic) or through solids (e.g., from air-conditioner rumbling, underground subways, or nearby businesses), special construction techniques will often be required to dampen these extraneous sounds (Figure 3.6).

FIGURE 3.6
Various isolation, absorption, and reflective acoustical treatments for the construction of a recording/monitoring environment.
(Courtesy of Auralex Acoustics, www.auralex.com)

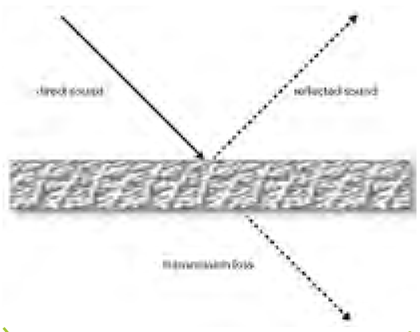


FIGURE 3.7
Transmission loss refers to the reduction of a sound signal (in dB) as it passes through an acoustic barrier.

If you happen to have the luxury of building a studio facility from the ground up, a great deal of thought should be put into selecting the studio's location. If a location has considerable neighborhood noise, you might have to resort to extensive (and expensive) construction techniques that can "float" the rooms (a process that effectively isolates and uncouples the inner rooms from the building's outer foundations). If there's absolutely no choice of studio location and the studio happens to be located next to a factory, just under the airport's main landing path or over the subway's uptown line ... you'll simply have to give in to destiny and build acoustical barriers to these outside interferences.

The reduction in the sound-pressure level (SPL) of a sound source as it passes through an acoustic barrier of a certain physical mass **(Figure 3.7)** is termed the transmission loss (TL) of a signal. This attenuation can be expressed (in dB) as:

$$TL = 14.5 \log M + 23$$

where TL is the transmission loss in decibels, and M is the surface density (or combined surface densities) of a barrier in pounds per square foot (lb/ft).

Because transmission loss is frequency dependent, the following equation can be used to calculate transmission loss at various frequencies with some degree of accuracy:

$$TL = 14.5 \log Mf - 16$$

where f is the frequency (in hertz).

Both common sense and the preceding two equations tell us that heavier acoustic barriers will yield a higher transmission loss. For example, **Table 3.1** tells us that a 12-inch-thick wall of dense concrete (yielding a surface density of 150 lb/ft^2) offers a much greater resistance to the transmission of sound than can a 4-inch cavity filled with sand (which yields a surface density of 32.3 lb/ft^2). From the second equation (TL = 14.5 log *Mf* – 16), we can also draw the conclusion that, for a given acoustic barrier, transmission losses will increase as the frequency rises. This can be easily illustrated by closing the door of a car that has its sound system turned up, or by shutting a single door to a music studio's control room. In both instances, the high frequencies will be greatly reduced in level, while the bass frequencies will be impeded to a much lesser extent. From this, the goal would seem to be to build a studio wall, floor, ceiling, window or door out of the thickest and most dense material that's available; however, expense and physical space often play roles in determining just how much of a barrier can be built to achieve the desired isolation. As such, a balance must usually be struck when using both space- and cost-effective building materials.

TABLE 3.1 ■ SURFACE DENSITIES OF COMMON BUILDING MATERIALS.

Material	Thickness (inches)	Surface Density (lb/ft²)
Brick	4	40.0
	8	80.0
Concrete (lightweight)	4	33.0
	12	100.0
Concrete (dense)	4	50.0
	12	150.0
Glass	–	3.8
	–	7.5
	–	11.3
Gypsum wallboard	–	2.1
	–	2.6
Lead	1/16	3.6
Particleboard	–	1.7
Plywood	–	2.3
Sand	1	8.1
	4	32.3
Steel	–	10.0
Wood	1	2.4

WALLS

When building a studio wall or reinforcing an existing structure, the primary goal is to reduce leakage (increase the transmission loss) through a wall as much as possible over the audible frequency range. This is generally done by:

- Building a wall structure that is as massive as is practically possible (both in terms of cubic and square foot density)
- Eliminating open joints that can easily transmit sound through the barrier
- Dampening structures, so that they are well supported by reinforcement structures and are free of resonances

The following guidelines can be helpful in the construction of framed walls that have high transmission losses:

- If at all possible, the inner and outer wallboards should not be directly attached to the same wall studs. The best way to avoid this is to alternately stagger the studs along the floor and ceiling frame, so that the front/back facing walls aren't in physical contact with each other **(Figure 3.8)**.
- Each wall facing should have a different density to reduce the likelihood of increased transmission due to resonant frequencies that might be sympathetic to both sides. For example, one wall might be constructed of two 5/8-inch gypsum wallboards, while the other wall might be composed of soft fiberboard that's surfaced with two 1/2-inch gypsum wallboards.
- If you're going to attach gypsum wallboards to a single wall face, you can increase transmission loss by mounting the additional layers (not the first layer) with adhesive caulking rather than using screws or nails.
- Spacing the studs 24 inches on center instead of using the traditional 16-inch spacing yields a slight increase in transmission loss.
- To reduce leakage that might make it through the cracks, apply a bead of non-hardening caulk sealant to the inner gypsum wallboard layer at the wall-to-floor, wall-to-ceiling and corner junctions.

Generally, the same amount of isolation is required between the studio and the control room as is required between the studio's interior and exterior environments. The proper building of this wall is important, so that an accurate tonal balance can be heard over the control-room monitors without promoting leakage between the rooms or producing resonances within the wall that would audibly color the signal. Optionally, a specially designed cavity, called a *soffit*, can be designed into the front-facing wall of the control room to house the larger studio monitors. This superstructure allows the main, far-field studio monitors to be mounted directly into the wall to reduce reflections and resonances in the monitoring environment.

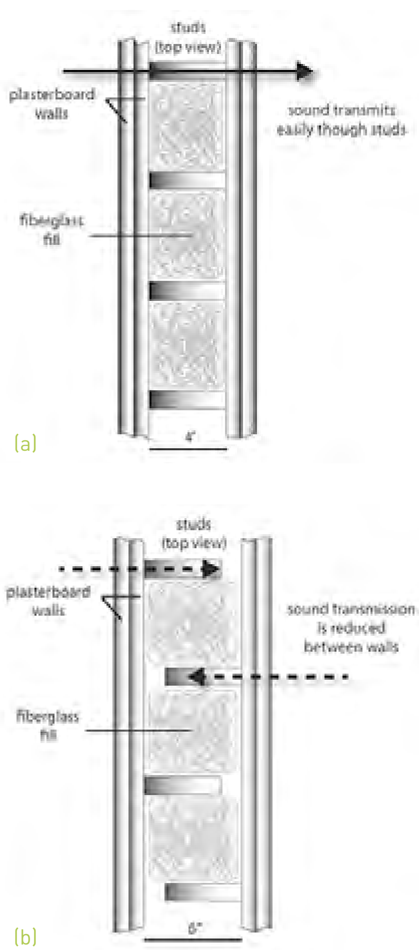


FIGURE 3.8

Double, staggered stud construction greatly reduces leakage by decoupling the two wall surfaces from each other: (a) top view showing walls that are directly tied to wall studs (allowing sound to easily pass through). (b) top view showing walls with offset, non-touching studs (so that sound doesn't easily pass from wall to wall).

It's important for the soffit to be constructed to high standards, using a multiple wall or high-mass design that maximizes the density with acoustically tight construction techniques in order to reduce leakage between the two rooms. Cutting corners by using substandard (and even standard) construction techniques in the building of a studio soffit can lead to unfortunate side effects, such as wall resonances, rattles, and increased leakage. Typical wall construction materials include:

- *Concrete*: This is the best and most solid material, but it is often expensive and it's not always possible to pour cement into an existing design.
- *Bricks* (hollow-form or solid-facing): This excellent material is often easier to place into an existing room than concrete.
- *Gypsum plasterboard*: Building multiple layers of plasterboard onto a double-walled stud frame is often the most cost- and design-efficient approach to reducing resonances and maximizing transmission loss. It's often a good idea to reduce these resonances by filling the wall cavities with rockwool or fiberglass, while bracing the internal structure to add an extra degree of stiffness.

Studio monitors can be designed into the soffit in a number of ways. In one expensive approach, the speakers' inner enclosures are cavities designed into walls that are made from a single concrete pour. Under these conditions, resonances are completely eliminated. Another less expensive approach has the studio monitors resting on poured concrete pedestals; in this situation, inserts can be cast into the pedestals that can accept threaded rebar rods (known as all-thread). By filing the rods to a chamfer or a sharper point, it's possible to adjust the position, slant and height of the monitors for final positioning into the soffit's wall framing. The most common and affordable approach uses traditional wood framing in order to design a cavity into which the speaker enclosures can be designed and positioned. Extra bracing and heavy construction should be used to reduce resonances.

FLOORS

For many recording facilities, the isolation of floor-borne noises from room and building exteriors is an important consideration. For example, a building that's located on a busy street and whose concrete floor is tied to the building's ground foundation might experience severe low-frequency rumble from nearby traffic. Alternatively, a second-floor facility might experience undue leakage from a noisy downstairs neighbor or, more likely, might interfere with a quieter neighbor's business. In each of these situations, increasing the isolation to reduce floor-borne leakage and/or transmission is essential. One of the most common ways to isolate floor-related noise is to construct a "floating" floor that is structurally decoupled from its subfloor foundation.

Common construction methods for floating a professional facility's floor uses either neoprene "hockey puck" isolation mounts, U-Boat floor floaters (Figure 3.9), or a continuous underlay, such as a rubberized floor mat. In these cases, the underlay is spread over the existing floor foundation and then covered with an overlaid plywood floor structure. In more extreme situations, this superstructure could be covered with reinforcing wire mesh and finally topped with a 4-inch layer of concrete (Figure 3.10). In either case, the isolated floor is then ready for carpeting, wood finishing, painting or any other desired surface.

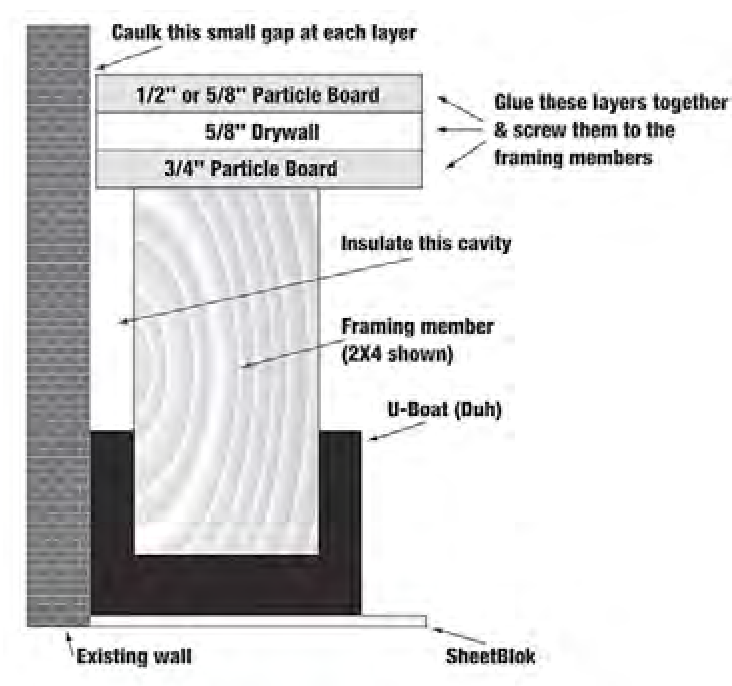


FIGURE 3.9
U-Boat™ floor beam float channels can be placed under a standard 2 x 4 floor frame to increase isolation. Floor floaters should be placed every 16 inches under a 2 x floor joist.
(Courtesy of Auralex Acoustics, www.auralex.com)

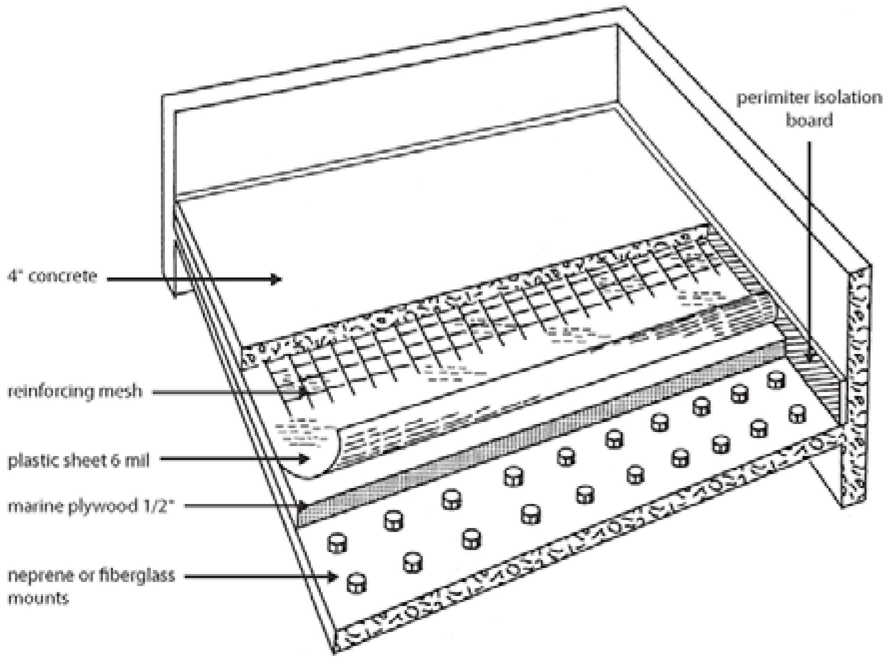


FIGURE 3.10
Basic guidelines for building a concrete floating floor using neoprene mounts.

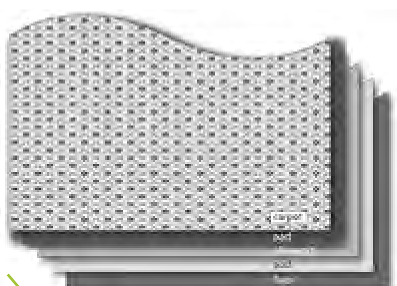


FIGURE 3.11
An alternative, cost-effective way to float an existing floor by layering relatively inexpensive materials.

An even more cost- and space-effective way to decouple a floor involves layering the original floor with a rubberized or carpet foam pad. A 1/2- or 5/8-inch layer of tongue-and-groove plywood or oriented strand board (OSB) is then laid on top of the pad. These should not be nailed to the subfloor; instead, they can be stabilized by glue or by locking the pieces together with thin, metal braces. Another foam pad can then be laid over this structure and topped with carpeting or any other desired finishing material (Figure 3.11).

It is important for the floating superstructure to be isolated from both the under-flooring and the outer wall. Failing to isolate these allows floor-borne sounds to be transmitted through the walls to the subfloor, and vice versa (often defeating the whole purpose of floating the floor). These wall perimeter isolation gaps can be sealed with pliable decoupling materials such as widths of soft mineral fiberboard, neoprene, silicone or other pliable materials.

RISERS

As we saw from the equation TL = 14.5 log *Mf* – 16, low-frequency sound travels through barriers much more easily than does high-frequency sound. It stands to reason that strong, low-frequency energy is transmitted more easily than high-frequency energy between studio rooms, from the studio to the control room or to outside locations. In general, the drum set is most likely to be the biggest leakage offender. By decoupling much of a drum set's low-frequency energy from a studio floor, many of the low-frequency leakage problems can be reduced. In most cases, the problem can be fixed by using a drum riser. Drum risers are available commercially (Figure 3.12), or they can be easily constructed. In order to reduce unwanted resonances, drum risers should be constructed using 2 x 6-inch or 2 x 8-inch beams for both the frame and the supporting joists (spaced at 16 or 12 inches on center, as shown in Figure 3.13). Sturdy 1/2- or 5/8-inch tongue-and-groove plywood panels should be glued to the supporting frames with carpenter's glue (or a similar wood glue) and then nailed or screwed down (using heavy-duty, galvanized fasteners). When the frame has dried, rubber coaster float channels or (at the very least) strips of carpeting should be attached to the bottom of the frame … and the riser will be ready for action.

FIGURE 3.12
HoverDeck™ 88 isolation riser.
(Courtesy of Auralex Acoustics, www.auralex.com)

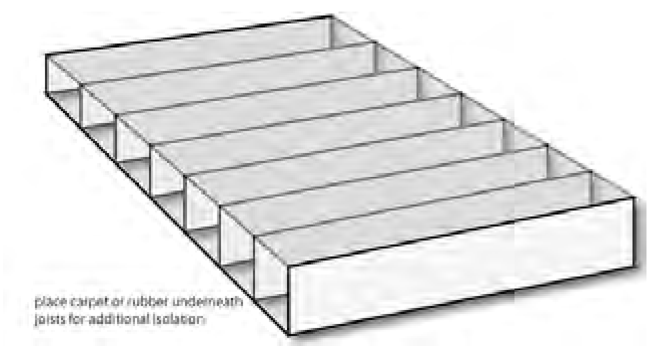
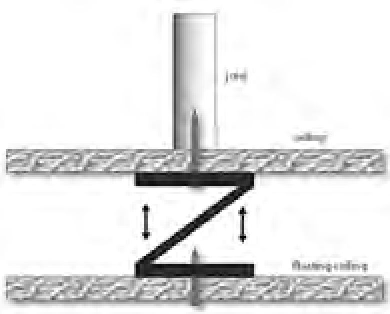


FIGURE 3.13
General construction details for a homemade drum riser.

CEILINGS

Foot traffic and other noises from above a sound studio or production room are another common source of external leakage. Ceiling noise can be isolated in a number of ways. If foot traffic is your problem and you're fortunate enough to own the floors above you, you can reduce this noise by simply carpeting the overhead hallway or by floating the upper floor. If you don't have that luxury, one approach to isolating ceiling-borne sounds is to hang a false structure from the existing ceiling or from the overhead joists (as is often done when a new room is being constructed). This technique can be fairly cost effective when spring or "Z" suspension channels are used **(Figure 3.14)**. Z channels are often screwed to the ceiling joists to provide a flexible, yet strong support to which a hanging wallboard ceiling can be attached. If necessary, fiberglass or other sound-deadening materials can be placed into the cavities between the overhead structures. Other more expensive methods use spring support systems to hang false ceilings from an existing structure.

(a)

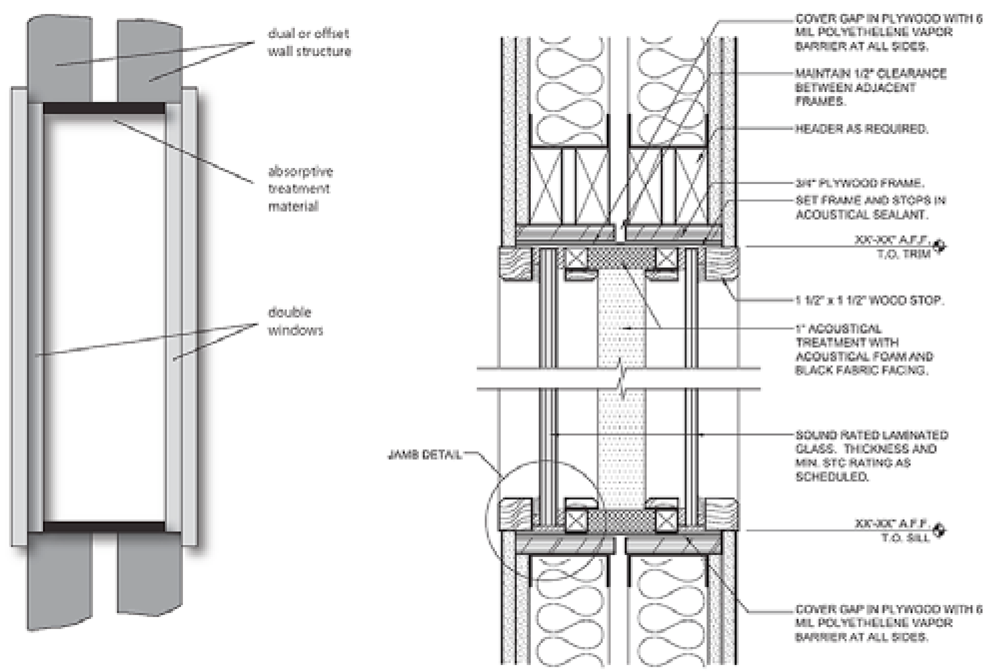


(b)

FIGURE 3.14
Ceiling isolator systems.
(a) RSIC-SI-1 (Resilient Sound Isolation Chips)
(courtesy of PAC International, Inc; www.pac-intl.com).
(b) Z channels can be used to hang a floating ceiling from an existing overhead structure.

WINDOWS AND DOORS

Access to and from a studio or production room area (in the form of windows and doors) can also be a potential source of sound leakage. For this reason, strict attention needs to be given to window and door design and construction. Visibility in a studio is extremely important within a music production environment. For example, when multiple rooms are involved, good visibility serves to promote effective communication between the producer or engineer and the studio musician (as well as among the musicians themselves). For this reason, windows have been an important factor in studio design since the beginning. The design and construction details for a window often vary with studio needs and budget requirements and can range from being deep, double-plate cavities that are built into double-wall constructions **(Figure 3.15)** to more modest prefab designs that are built into a single wall. Other more expensive designs include floor-to- ceiling windows that create a virtual "glass wall," as well as those that offer sweeping vistas, which are designed into poured concrete soffit walls.



(a) (b)

FIGURE 3.15

Detail for a practical window construction between the control room and studio.
(a) simplified drawing
(b) detailed drawing.
(Courtesy of Russ Berger Design Group, Inc., www.rbdg.com)

Access doors to and from the studio, control room, and exterior areas should be constructed of solid wood or high-quality acoustical materials (Figure 3.16), as solid doors generally offer higher TL values than their cheaper, hollow counterparts. No matter which door type is used, the appropriate seals, weatherstripping, and doorjambs should be used throughout so as to reduce leakage through the cracks. Whenever possible, double-door designs should be used to form an acoustical sound lock (Figure 3.17). This construction technique dramatically reduces leakage because the air trapped between the two solid barriers offers up high TL values.

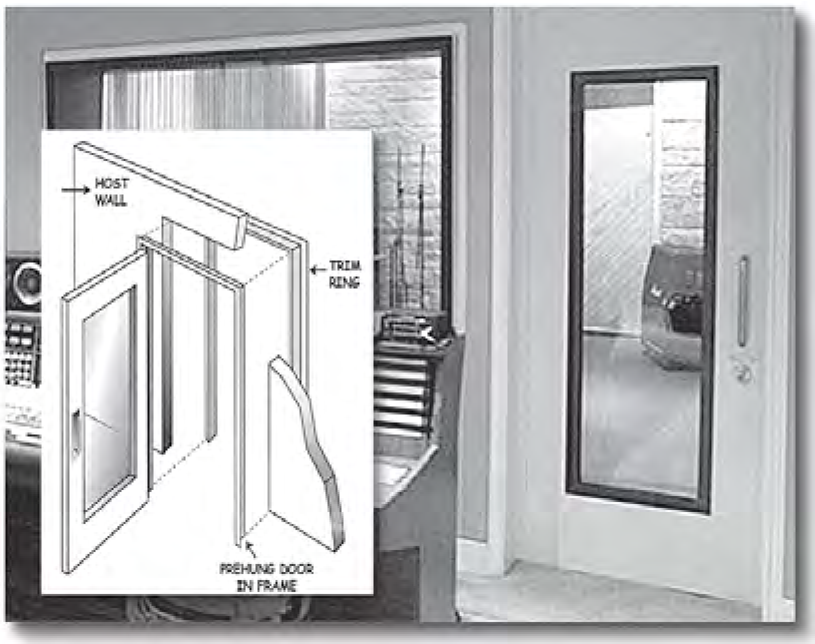


FIGURE 3.16
A SoundSecure™ studio door.
(Courtesy of ETS-Lingren, www.ets-lindgren.com)

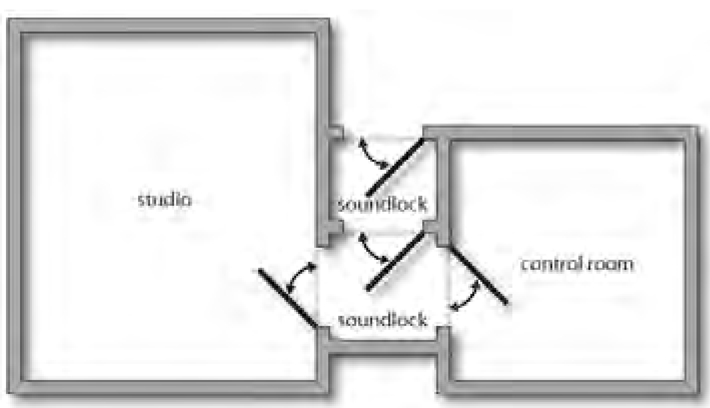


FIGURE 3.17
Example of a sound lock door system design.

ISO-ROOMS AND ISO-BOOTHS

Isolation rooms (*iso-rooms*) are acoustically isolated or sealed areas that are built into a music studio or just off of a control room **(Figure 3.18)**. These recording areas can be used to separate louder instruments from softer ones (and vice versa) in order to reduce leakage and to separate instrument types by volume to maintain control over the overall ensemble balance. For example:

- To eliminate leakage when recording scratch vocals (a guide vocal track that's laid down as a session reference), a vocalist might be placed in a small room while the rhythm ensemble is placed in the larger studio area.
- A piano or other instrument could be isolated from the larger area that's housing a full string ensemble.
- A B3 organ could be blaring away in an iso-room while backing vocals are being laid down in the main room. ... The possibilities are endless.

FIGURE 3.18
Iso-room design (located at right) at Studio Records, LLC, Ft. Worth, TX.
(Courtesy of Russ Berger Design Group, Inc., www.rbdg.com)

An iso-room can be designed to have any number of acoustical properties. By having multiple rooms and/or iso-room designs in a studio, several acoustical environments can be offered that range from being more reflective (live) to absorptive (dead) ... or a specific room can be designed to better fit the acoustical needs of a particular instrument (e.g., drums, piano or vocals). These rooms can be designed as totally separate areas that can be accessed from the main studio or control room, or they might be directly tied to the main studio by way of sliding walls or glass sliding doors. In short, their form and function can be put to use to fit the needs and personality of the session.

Isolation booths (iso-booths) provide the same type of isolation as an iso-room, but are often much smaller **(Figure 3.19)**. Often called vocal booths, these mini-studios are perfect for isolating vocals and single instruments from the larger studio. In fact, rooms that have been designed and built for the express purpose of mixing down a recording will often only have an iso-booth … and no other recording room. Using this space-saving option, vocals or single instruments can be easily overdubbed on site, and should more space be needed a larger studio can be booked to fit the bill.

FIGURE 3.19
Example of an iso-booth in action.

NOISE ISOLATION WITHIN THE CONTROL ROOM

Isolation between rooms and the great outdoors isn't the only noise-related issue in the modern-day recording or project studio. The proliferation of computers, multitrack tape machines and cooling systems has created issues that present their own Grinch-like types of noise, Noise, NOISE, NOISE!!! This usually manifests itself in the form of system fan noise, transport tape noise and computer-related sounds from CPUs, case fans, hard drives and the like.

When it comes to isolating tape transport and system fan sounds, should budget and size constraints permit, it is often wise to build an iso-room or iso-closet that's been specifically designed and ventilated for containing such equipment. An equipment room that has easy-access doors that provide for current/future wiring needs can add a degree of peace-'n'-quiet and an overall professionalism that will make both you and your clients happy.

Within a smaller studio or project studio space, such a room isn't always possible; however, with care and forethought the whizzes and whirrs of the digital era can be turned into a nonissue that you'll be proud of.

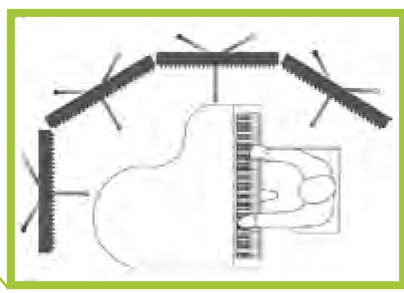
Here are a few examples of the most common problems and their solutions:

- Place the computer(s) in an isolated case, alcove or room (care needs to be taken to monitor the CPU/case temperatures so as not to harm your system).
- Connect the studio computers via a high-speed network to a remote server location.
- Replace fans with quieter ones. By doing some careful Web searching or by talking to your favorite computer salesperson, it's often possible to install CPU and case fans that are quieter than most off-the-shelf models.

ACOUSTIC PARTITIONS

Movable acoustic partitions (also known as flats or gobos) are commonly used in studios to provide on-the-spot barriers to sound leakage. By partitioning a musician and/or instrument on one or more sides and then placing the mic inside the temporary enclosure, isolation can be greatly improved in a flexible way that can be easily changed as new situations arise. Acoustic partitions are currently available on the commercial market in various design styles and types for use in a wide range of studio applications **(Figure 3.20)**. For those on a budget, or who have particular isolation needs, it's relatively simple to get out the workshop tools and make your own flats that are based around wood frames, fiberglass or other acoustically absorptive materials with your favorite colored fabric coverings— and ingenious craftsmanship **(Figure 3.21)**.

(a)



(b)

FIGURE 3.20
Acoustic partition flat examples: (a) S5–2L "Sorber" baffle system
(Courtesy of ClearSonic Mfg., Inc., www.clearsonic.com)
(b) piano panel setup
(Courtesy of Auralex Acoustics, www.auralex.com)

If you can't get a flat when you need one, you can often improvise using common studio and household items. For example, a simple partition can be easily made on the spot by grabbing a mic/boom stand combination and retracting the boom halfway at a 90° angle to make a T-shape. Simply drape a blanket or heavy coat over the T-bar and voilà—you've built a quick-'n'-dirty dividing flat.

When using a partition, it's important to be aware that musicians' need to be able to see each other, the conductor and the producer. Musicality and human connectivity almost always take precedence over technical issues.

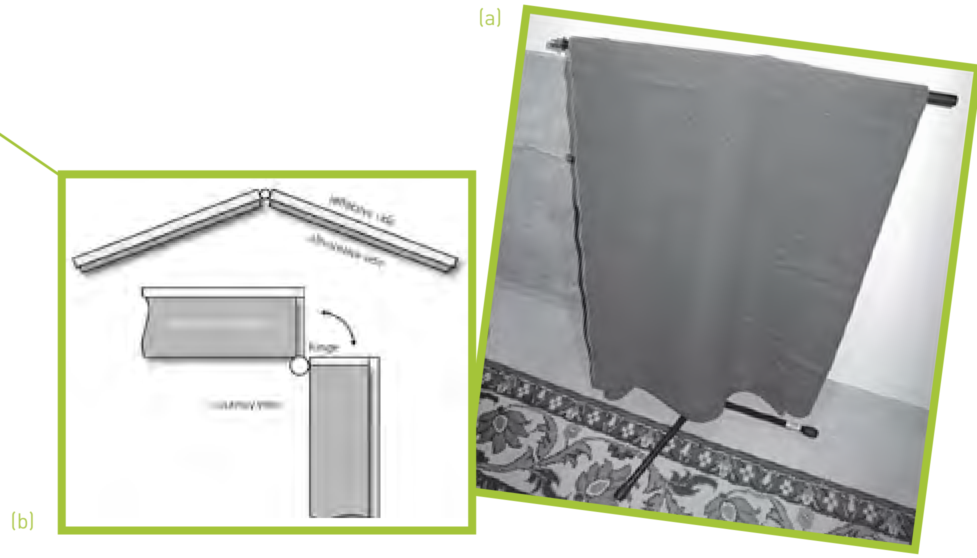
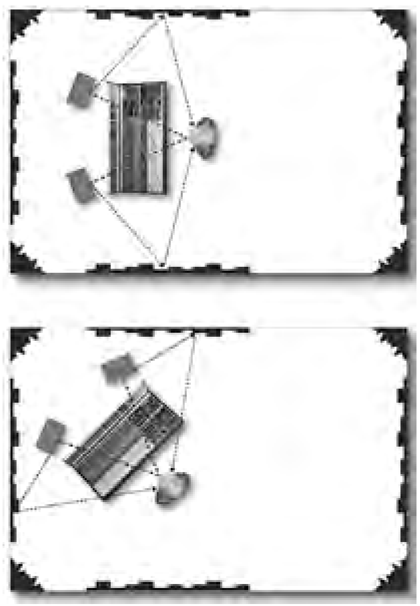


FIGURE 3.21
Examples of a homemade flat: (a) the "blanket and a boom" trick; (b) homemade flat design.

SYMMETRY IN CONTROL ROOM DESIGN

While many professional studios are built from the ground up using standard acoustic and architectural guidelines, most budget-minded production and project studios are often limited by their own unique sets of building, space and acoustic constraints. Even though the design of a budget, project or bedroom control room might not be acoustically perfect, if speakers are to be used in the monitoring environment, certain ground rules of acoustical physics must be followed in order to create a proper listening environment.

One of the most important acoustic design rules in a monitoring environment is the need for symmetrical reflections on all axes within the design of a control room or single-room project studio. In short, the center and acoustic *imaging* (ability to discriminate placement and balance in a stereo or surround field) is best when the listener, speakers, walls and other acoustical boundaries are symmetrically centered about the listener's position (often in an equilateral triangle). In a rectangular room, the best low-end response can be obtained by orienting the console and loudspeakers into the room's long dimension (Figure 3.22a). Should space or other room considerations come into play, centering the listener/monitoring position at a 45° angle within a symmetrical corner (Figure 3.22b) is another example of how the left/right imagery can be reasonably maintained.



(a)

(b)

FIGURE 3.22
Various acceptable symmetries in a monitoring environment: (a) Acoustic reflections must be symmetrical about the listener's position. In addition, orienting a control room along the long dimension can extend the room's low-end response. (b) Placing the listening environment symmetrically in a corner is another example of how the left/right imagery can be improved over an off-center placement.

Should any primary boundaries of a control room (especially wall or ceiling boundaries near the mixing position) be asymmetrical from side to side, sounds heard by one ear will receive one combination of direct and reflected sounds, while the other ear will hear a different acoustic balance (Figure 3.23). This condition can drastically alter the sound's center image characteristics, so that when a sound is actually panned between the two monitor speakers the sound will appear to be centered; however, when the sound is heard in another studio or standard listening environment the imaging may be off center. To avoid this problem, care should be taken to ensure that both the side and ceiling boundaries are largely symmetrical with respect to each other and that all of the speaker level balances are properly set.

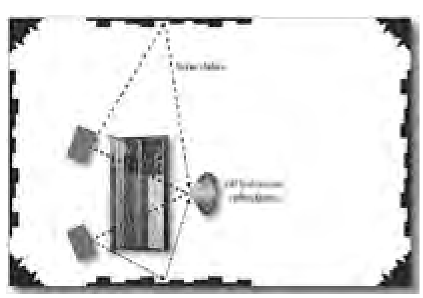


FIGURE 3.23
Placing the monitoring environment off-center and in a corner will affect the audible center image, and placing one speaker in a 90° corner can cause an off-center bass buildup and adversely affect the mix's imagery. Shifting the listener/monitoring position into the center will greatly improve the left/right imagery.

While we're on the subject of the relationship between the room's acoustic layout and speaker placement, it's always wise to place nearfield and all other speaker enclosures at points that are equidistant to the listener in the stereo and surround field. Whenever possible, speaker enclosures should be placed 1 to 2 feet away from the nearest wall and/or corner, which helps to avoid bass buildups that acoustically occur at boundary and corner locations. In addition to strategic speaker placement, homemade or commercially available isolation pads can be used to reduce resonances that often occur whenever enclosures are placed directly onto a table or flat surface.

FREQUENCY BALANCE

Another important factor in room design is the need for maintaining the original *frequency balance* of an acoustic signal. In other words, the room should exhibit a relatively flat frequency response over the entire audio range without adding its own particular sound coloration. The most common way to control the tonal character of a room is to use materials and design techniques that govern the acoustical reflection and absorption factors.

REFLECTIONS

One of the most important characteristics of sound as it travels through air is its ability to reflect off a boundary's surface at an angle that's equal to (and opposite of) its original angle of incidence **(Figure 3.24)**. Just as light bounces off a mirrored surface or multiple reflections can appear within a mirrored room, sound reflects throughout room surfaces in ways that are often amazingly complex. Through careful control of these reflections, a room can be altered to improve its frequency response and sonic character.

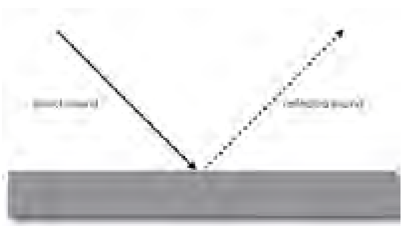


FIGURE 3.24
Sound reflects off a surface at an angle equal (and opposite) to its original angle of incidence, much as light will reflect off a mirror.

In Chapter 2, we learned that sonic reflections can be controlled in ways that disperse the sound outward in a wide-angled pattern (through the use of a convex surface) or focus them on a specific point (through the use of a concave surface). Other surface shapes, on the other hand, can reflect sound back at various other angles. For example, a 90° corner will reflect sound back in the same direction as its incident source (a fact that accounts for the additive acoustic buildups at various frequencies at or near a wall-to-corner or corner-to-floor intersection).

The all-time winner of the "avoid this at all possible cost" award goes to constructions that include opposing parallel walls in their design. Such conditions give rise to a phenomenon known as *standing waves*. Standing waves (also known as room modes) occur when sound is reflected off of parallel surfaces and travels back on its own path, thereby causing phase differences to interfere with a room's amplitude response (Figure 3.25). Room modes are expressed as integer multiples of the length, width and depth of the room and indicate which multiple is being referred to for a particular reflection.

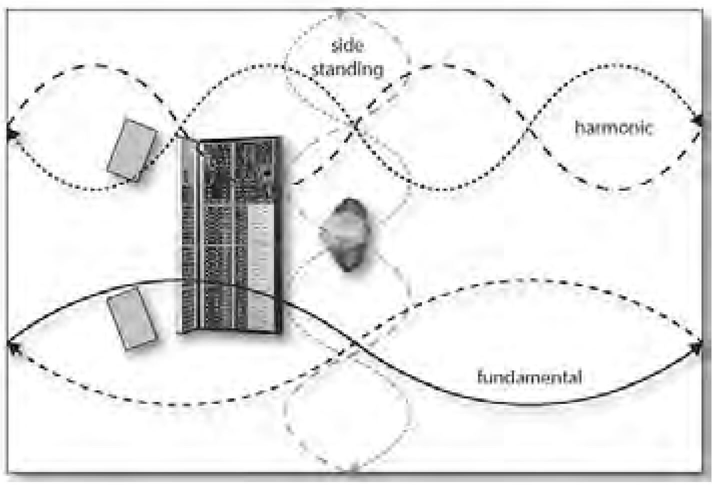


FIGURE 3.25
Standing waves within a room with reflective parallel surfaces can potentially cancel and reinforce frequencies within the audible spectrum, causing changes in its response.

Walking around a room with moderate to severe mode problems produces the sensation of increasing and/or decreasing volume levels at various frequencies throughout the area. These perceived volume changes are due to amplitude (phase) cancellations and reinforcements of the combined reflected waveforms at the listener's position. The distance between parallel surfaces and the signal's wavelength determines the nodal points that can potentially cause sharp peaks or dips at various points in the response curve (up to or beyond 19 dB) at the affected fundamental frequency (or frequencies) and upper harmonic intervals (Figure 3.26). This condition exists not only for opposing parallel walls but also for all parallel surfaces (such as between the floor and ceiling or between two reflective flats). From this discussion, it's obvious that the most effective way to prevent standing waves is to construct walls, boundaries and ceilings that are nonparallel.

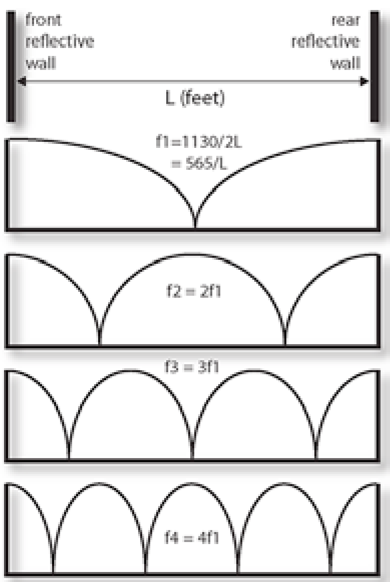


FIGURE 3.26
The reflective, parallel walls create an undue number of standing waves, which occur at various frequency intervals (f_1, f_2, f_3, f_4, and so on).

FIGURE 3.27

Commericall diffuser examples:
(a) Art Diffusor sound diffusers Models F, C & E

(Courtesy of acousticsfirst, www.acousticsfirst.com)

(b) SpaceArray sound diffusers

(Courtesy of pArtScience, www.partscience.com)

(c) open-ended view of a Primacoustic™ Razorblade quadratic diffuser

(Courtesy of Primacoustic Studio Acoustics, www.primacoustics.com)

If the room in question is rectangular or if further sound-wave dispersion is desired, diffusers can be attached to the wall and/or ceiling boundaries to help break up standing waves. Diffusers **(Figure 3.27)** are acoustical boundaries that reflect the sound wave back at various angles that are wider than the original incident angle (thereby breaking up the energy-destructive standing waves). In addition, the use of both nonparallel and diffusion wall construction can reduce extreme, recurring reflections and smooth out the reverberation characteristics of a room by building more complex acoustical pathways.

Flutter echo (also called *slap echo*) is a condition that occurs when parallel boundaries are spaced far enough apart that the listener is able to discern a number of discrete echoes. Flutter echo often produces a "boingy," hollow sound that greatly affects a room's sound character as well as its frequency response. A larger room (which might contain delayed echo paths of 50 msec or more) can have its echoes spaced far enough apart in time that the discrete reflections produce echoes that actually interfere with the intelligibility of the direct sound, often resulting in a jumble of noise. In these cases, the proper application of absorption and acoustic dispersion becomes critical.

When speaking of reflections within a studio control room, one long-held design concept relates to the concept of designing the room such that the rear of the room is largely reflective and diffuse in nature (acoustically "live"), while the front of the room is largely or partially absorptive (acoustically "dead"). This philosophy **(Figure 3.28)** argues for the fact that the rear of the room should be largely reflective providing for a balanced environment that can help reinforce positive reflections that can add acoustic "life" to the mix experience **(Figure 3.29)**. The front of the room would tend more toward the absorptive side in a way that reduces standing-wave and flutter reflections that would interfere with the overall response of the room.

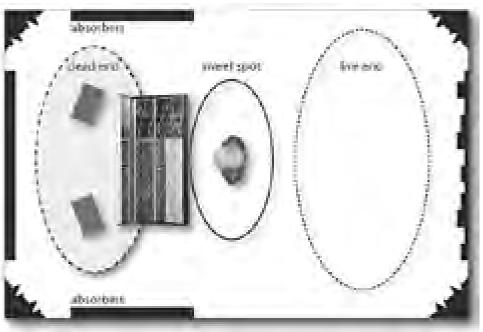


FIGURE 3.28
Control-room layout showing the live end toward the back of the room and the dead end toward the front of the room.

FIGURE 3.29
Placing bookshelves along the rear wall can provide both diffusion and a place for lots of storage.

It's important to realize that no two rooms will be acoustically the same or will necessarily offer the same design challenges. The one constant is that careful planning, solid design and ingenuity are the foundation of any good-sounding room. You should also keep in mind that numerous studio design and commercial acoustical product firms are available that offer assistance for both large and small projects. Getting professional advice is a good thing.

ABSORPTION

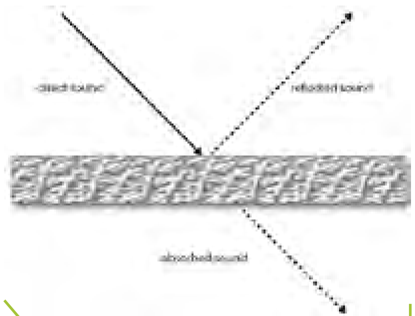


FIGURE 3.30
Absorption occurs when only a portion of the incident acoustic energy is reflected back from a material's surface.

Another factor that often has a marked effect on an acoustic space involves the use of surface materials and designs that can absorb unwanted sounds (either across the entire audible band or at specific frequencies). The *absorption* of acoustic energy is, effectively, the inverse of reflection **(Figure 3.30)**. Whenever sound strikes a material, the amount of acoustic energy that's absorbed relative to the amount that's reflected can be expressed as a simple ratio known as the material's *absorption coefficient*. For a given material, this can be represented as:

$$A = I_a/I_r$$

where I_a is the sound level (in dB) that is absorbed by the surface (often dissipated in the form of physical heat), and I_r is the sound level (in dB) that is reflected back from the surface.

The factor (1 – *a*) is a value that represents the amount of reflected sound. This makes the coefficient a decimal percentage value between 0 and 1. If we say that a surface material has an absorption coefficient of 0.25, we're actually saying that the material absorbs 25% of the original acoustic energy and reflects 75% of the total sound energy at that frequency. A sample listing of these coefficients is provided in **Table 3.2**.

To determine the total amount of absorption that's obtained by the sum of all the absorbers within a total volume area, it's necessary to calculate the average absorption coefficient for all of the surfaces together. The *average absorption coefficient* (A_{ave}) of a room or area can be expressed as:

$$A_{ave} = s_1a_1 + s_2a_2 + \ldots s_na_n/S$$

where s_1, s_2, ... , s_n are the individual surface areas; a_1, a_2, ..., a_n are the individual absorption coefficients of the individual surface areas; and *S* is the total square surface area.

On the subject of absorption, one common misconception is that the use of large amounts of sound-deadening materials will reduce room reflections and therefore make a room sound "good." In fact, the overuse of absorption will often have the effect of reducing high frequencies, creating a skewed room response that is dull and bass-heavy, as well as reducing constructive room reflections that are important to a properly designed room. In fact, with regard to the balance between reflection, diffusion and absorption, many designers agree that a balance of 25% absorption and 25% diffuse reflections is a good ratio that can help preserve the "life" of a room, while reducing unwanted buildups.

FIGURE 3.31
Various commerical absorption and diffusion wall treatments
(Courtesy of Auralex Acoustics, www.auralex.com).

FIGURE 3.32
Homemade absorber panel showing fabric that has been stretched over a wooden frame. Once the rockwool is placed inside, the frame (which can be of any size or form) can be hung on the wall, lowered from the ceiling or placed in a corner.

TABLE 3.2 ■ ABSORPTION COEFFICIENTS FOR VARIOUS MATERIALS.

	Coefficients (Hz)					
Material	125	250	500	1000	2000	4000
Brick, unglazed	0.03	0.03	0.03	0.04	0.05	0.07
Carpet (heavy, on concrete)	0.02	0.06	0.14	0.37	0.60	0.65
Carpet (with latex backing, on 40-oz hair-felt or foam rubber)	0.03	0.04	0.11	0.17	0.24	0.35
Concrete or terrazzo	0.01	0.01	0.015	0.02	0.02	0.02
Wood	0.15	0.11	0.10	0.07	0.06	0.07
Glass, large heavy plate	0.18	0.06	0.04	0.03	0.02	0.02
Glass, ordinary window	0.35	0.25	0.18	0.12	0.07	0.04
Gypsum board nailed to 2 x 4 studs on 16-inch centers	0.013	0.015	0.02	0.03	0.04	0.05
Plywood (3/8 inch)	0.28	0.22	0.17	0.09	0.10	0.11
Air (sabins/1000 ft^3)	–	–	–	–	2.3	7.2
Audience seated in upholstered seats	0.08	0.27	0.39	0.34	0.48	0.63
Concrete block, coarse	0.36	0.44	0.31	0.29	0.39	0.25
Light velour (10 oz/yd^2 in contact with wall)	0.29	0.10	0.05	0.04	0.07	0.09
Plaster, gypsum, or lime (smooth finish on tile or brick)	0.44	0.54	0.60	0.62	0.58	0.50
Wooden pews	0.57	0.61	0.75	0.86	0.91	0.86
Chairs, metal or wooden, seats unoccupied	0.15	0.19	0.22	0.39	0.38	0.30

Note: These coefficients were obtained by measurements in the laboratories of the Acoustical Materials Association. Coefficients for other materials may be obtained from Bulletin XXII of the association.

HIGH-FREQUENCY ABSORPTION

The absorption of high frequencies is accomplished through the use of dense porous materials, such as fiberglass, dense fabric and carpeting. These materials generally exhibit high absorption values at higher frequencies, which can be used to control room reflections in a frequency-dependent manner. Specially designed foam and acoustical treatments are also commercially available that can be attached easily to recording studio, production room or control-room walls as a means of taming multiple room reflections and/or dampening high-frequency reflections **(Figure 3.31)**.

In addition to buying commercial absorbers, it's very possible to put your handy shop tools to work by building your own absorber panels (of any shape, depth and style). One straightforward way to do this is to build a wooden box side-frame and then stretch a light fabric (of your favorite color or pattern) around the frame, then place commercially-available rockwool bats (or resized sections) inside. When done right, these absorbers **(Figure 3.32)** can look very pro and fit your specific needs at a fraction of their commercial equivalents … sometimes with better results.

LOW-FREQUENCY ABSORPTION

As shown in **Table 3.2**, materials that are absorptive in the high frequencies often provide little resistance to the low-frequency end of the spectrum (and vice versa). This occurs because low frequencies are best damped by pliable materials, meaning that low-frequency energy is absorbed by the material's ability to bend and flex with the incident waveform **(Figure 3.33)**. Rooms that haven't been built to the shape and dimensions to properly handle the low end will need to be controlled by using bass traps that are tuned to reduce the room's resonance frequencies.

Another absorber type can be used to reduce low-frequency buildup at specific frequencies (and their multiples) within a room. This type of attenuation device (known as a *bass trap*) is available in a number of design types:

- Quarter-wavelength trap
- Pressure-zone trap
- Functional trap.

(a)

(b)

FIGURE 3.33

Low-frequency absorption. (a) A carefully designed surface that can be "bent" by oncoming soundwaves can be used to absorb low frequencies. (b) Primacoustic™ Polyfuser, a combination diffuser and bass trap

(Courtesy of Primacoustic Studio Acoustics, www.primacoustics.com).

THE QUARTER-WAVELENGTH TRAP

The quarter-wavelength bass trap **(Figure 3.34)** is an enclosure with a depth that's one-fourth the wavelength of the offending frequency's fundamental frequency and is often built into the rear facing wall, ceiling or floor structure and covered by a metal grating to allow foot traffic. The physics behind the absorption of a calculated frequency (and many of the harmonics that fall above it) rests in the fact that the pressure component of a sound wave will be at its maximum at the rear boundary of the trap … when the wave's velocity component is at a minimum. At the mouth of the bass trap (which is at a one-fourth wavelength distance from this rear boundary), the overall acoustic pressure will be at its lowest, while the velocity component (molecular movement) will be at its highest potential. Because the wave's motion (force) is greatest at the trap's opening, much of the signal can be absorbed by placing an absorptive material at that opening point. A low-density fiberglass lining can also be placed inside the trap to increase absorption (especially at harmonic intervals of the calculated fundamental).

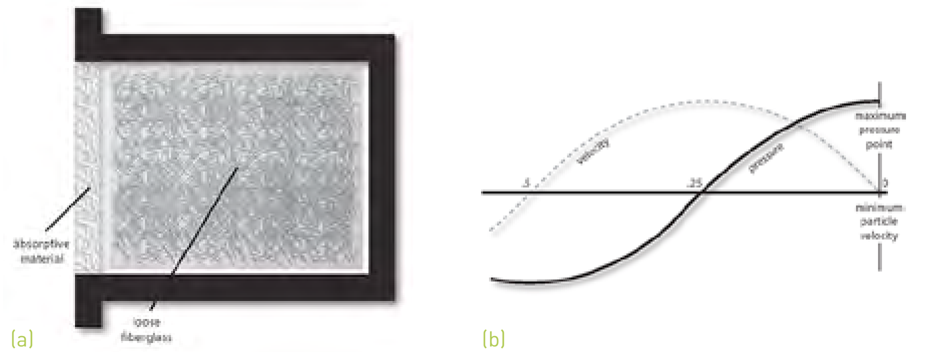


FIGURE 3.34
A quarter-wavelength bass trap: (a) physical concept design; (b) sound is largely absorbed as heat, since the particle velocity (motion) is greatest at the trap's quarter-wavelength opening.

PRESSURE-ZONE TRAP

The pressure-zone bass trap absorber **(Figure 3.35)** works on the principle that sound pressure is doubled at large boundary points that are at 90° angles (such as walls and ceilings). By placing highly absorptive material at a boundary point (or points, in the case of a corner/ceiling intersection), the built-up pressure can be partially absorbed.

FIGURE 3.35
LENRD™ bass traps.
(Courtesy of Auralex Acoustics, www.auralex.com).

FUNCTIONAL TRAP

Originally created in the 1950s by Harry F. Olson (former director of RCA Labs), the functional bass trap **(Figure 3.36)** uses a material generally formed into a tube or half-tube structure that is rigidly supported so as to reduce structural vibrations. By placing these devices into corners, room boundaries or in a freestanding spot, a large portion of the undesired bass buildup frequencies can be absorbed.

By placing a reflective surface over the portion of the trap that faces into the room, frequencies above 400 Hz can be dispersed back into the room or focal point. Figure 3.37 shows how these traps can be used in the studio to break up reflections and reduce bass buildup.

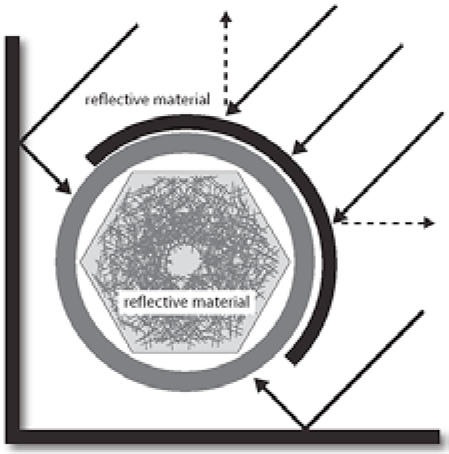


FIGURE 3.36
A functional bass trap that has been placed in a corner to prevent bass buildup.

FIGURE 3.37
Quick Sound Field.
(Courtesy of Acoustic Sciences Corporation, www.tubetrap.com)

ROOM REFLECTIONS AND ACOUSTIC REVERBERATION

Another criterion for studio design is the need for a desirable room ambience and intelligibility, which is often contradictory to the need for good acoustic separation between instruments and their pickup. Each of these factors is governed by the careful control and tuning of the reverberation constants within the studio over the frequency spectrum.

Reverberation (*reverb*) is the persistence of a signal (in the form of reflected waves within an acoustic space) that continues after the original sound has ceased. The effect of these closely spaced and random multiple echoes gives us perceptible cues as to the size, density and nature of an acoustic space. Reverb also adds to the perceived warmth and spatial depth of recorded sound and plays an extremely important role in the perceived enhancement of music.

As was stated in the latter part of Chapter 2, the reverberated signal itself can be broken down into three components:

- Direct sound
- Early reflection
- Reverb

The direct signal is made up of the original, incident sound that travels from the source to the listener. Early reflections consist of the first few reflections that are projected to the listener off of major boundaries within an acoustic space; these reflections generally give the listener subconscious cues as to the size of the room. (It should be noted that strong reflections off of large, nearby surfaces can potentially have detrimental cancellation effects that can degrade a room's sound and frequency response at the listening position.) The last set of signal reflections makes up the actual reverberation characteristic. These signals are composed of random reflections that travel from boundary to boundary in a room and are so closely spaced that the brain can't discern the individual reflections. When combined, they are perceived as a single decaying signal.

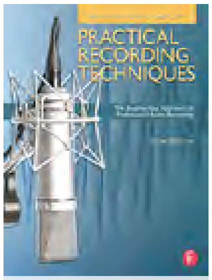
Technically, reverb is considered to be the time that's required for a sound to die away to a millionth of its original intensity (resulting in a decrease over time of 60 dB), as shown by the following formula:

$$RT_{60} = V \times 0.049/AS$$

where RT is the reverberation time (in sec), *V* is the volume of the enclosure (in ft^3), *A* is the average absorption coefficient of the enclosure, and *S* is the total surface area (in ft^2). As you can see from this equation, reverberation time is directly proportional to two major factors: the volume of the room and the absorption coefficients of the studio surfaces. A large environment with a relatively low absorption coefficient (such as a large cathedral) will have a relatively long RT_{60} decay time, whereas a small studio (which might incorporate a heavy amount of absorption) will have a very short RT_{60}.

The style of music and the room application will often determine the optimum RT_{60} for an acoustical environment. Reverb times can range from 0.25 sec in a smaller absorptive recording studio environment to 1.6 sec or more in a larger music or scoring studio. In certain designs, the RT_{60} of a room can be altered to fit the desired application by using movable panels or louvers or by placing carpets in a room. Other designs might separate a studio into sections that exhibit different reverb constants. One side of the studio (or separate iso-room) might be relatively non-reflective or dead, whereas another section or room could be much more acoustically live. The more reflective, live section is often used to bring certain instruments that rely heavily on room reflections and reverb, such as strings or an acoustic guitar, to “life.” The recording of any number of instruments (including drums and percussion) can also greatly benefit from a well-designed acoustically live environment.

Isolation between different instruments and their pickups is extremely important in the studio environment. If leakage isn’t controlled, the room’s effectiveness becomes severely limited over a range of applications. The studio designs of the 1960s and 1970s brought about the rise of the “sound sucker” era in studio design. During this time, the absorption coefficient of many rooms was raised almost to an anechoic (no reverb) condition. With the advent of the music styles of the 1980s and a return to the respectability of live studio acoustics, modern studio and control-room designs have begun to increase in size and “liveness” (with a corresponding increase in the studio’s RT_{60}). This has reintroduced the buying public to the thick, live-sounding music production of earlier decades, when studios were larger structures that were more attuned to capturing the acoustics of a recorded instrument or ensemble.



The following is excerpted from *Practical Recording Techniques* by Bruce Bartlett and Jenny Bartlett.

SETTING UP YOUR STUDIO

Once you have your equipment, you need to connect it together with cables, and possibly install equipment racks and acoustic treatment. Let's consider each step.

CABLES

Cables carry electric signals from one audio component to another. They are usually made of one or two insulated conductors (wires) surrounded by a fine-wire mesh shield that reduces hum. Outside the shield is a plastic or rubber insulating jacket. On both ends of the cable are connectors. See Companion Website.

Cables are either balanced or unbalanced. A balanced line is a cable that uses two wires (conductors) to carry the signal, surrounded by a shield **(see Figure 4.13)**. Each wire has equal impedance to ground. An unbalanced line has a single conductor surrounded by a shield **(see Figure 4.14)**. The conductor and shield carry the signal. A balanced line rejects hum better than an unbalanced line, but an unbalanced line less than 10 feet long usually provides adequate hum rejection and costs less.

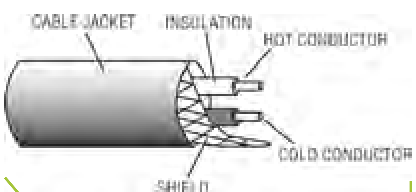


FIGURE 4.13
A 2-conductor shielded, balanced line.

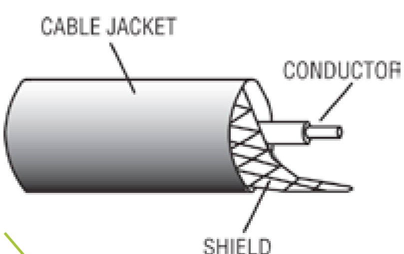


FIGURE 4.14
A 1-conductor shielded, unbalanced line.

A cable carries one of these five signal levels or voltages:

- Mic level: about 2 mV (0.002 volt) to about 1V depending on how loud the sound source is, and how sensitive the mic is
- Instrument level: typically 0.1V to 1V for passive pickups; up to 1.75V for active pickups
- Semipro or consumer line level: -10dBV (0.316 volt)
- Pro line level: +4dBu (1.23 volts)
- Speaker level: about 20 volts.

Note that "instrument level" can overlap mic level and line level.

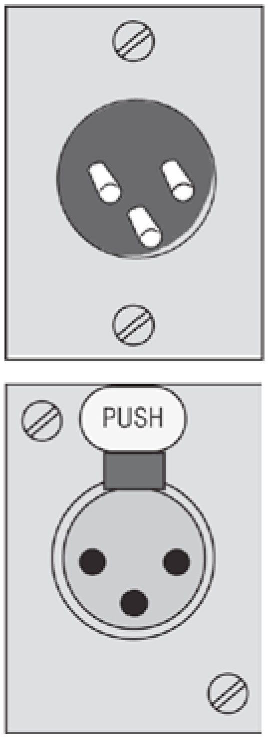
Some manufacturers use dBu instead of volts.

Mic level: about -52 dBu to +2.2 dBu

Instrument level: -17.7 dBu to +7 dBu

Semipro or consumer line level: -7.8 dBu

Pro line level: +4 dBu



EQUIPMENT CONNECTORS

Recording equipment also has balanced or unbalanced connectors built into the chassis. Be sure your cable connectors match your equipment connectors.

Balanced equipment connectors:

- 3-pin (XLR-type) connector— **Figure 4.15**
- 1/4-inch TRS (tip-ring-sleeve) phone jack— **Figure 4.16**

Unbalanced equipment connectors:

- 1/4-inch TS (tip-sleeve) phone jack— **Figure 4.16**
- Phono jack (RCA connector)— **Figure 4.17**

A jack is a receptacle; a plug inserts into a jack.

FIGURE 4.15
A 3-pin XLR-type connector used in balanced equipment.
Top: male output connector. Bottom: female input connector.

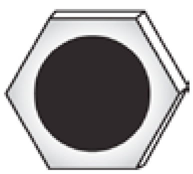


FIGURE 4.16
A 1/4-inch phone jack used in balanced and unbalanced equipment

Connectors are confusing because a single connector can have several functions (usually not at the same time). Here are some examples:

- XLR: balanced line input at +4 dBu, balanced mic input at 2 mV to 1 V, or balanced line output at +4 dBu
- TRS (stereo 1/4-inch phone jack): balanced mic input, insert send/return connectors (line level), instrument input, or stereo headphones
- TS (mono 1/4-inch phone jack): unbalanced mic input, unbalanced line-level input or output (+4 dBu or –10 dBV), instrument input, or low-cost speaker connector
- Combi connector: An XLR mic input plus a TRS input (instrument level or line level)
- RCA (phono): home stereo line-level input or output at –10 dBV, composite video input/ output, or SPDIF digital-audio input/output
- 1/8-inch (3.5 mm) mini phone jack: headphone jack, low-cost stereo mic input, line output in a portable recorder at –10 dBV; or a computer sound card's line in, line out, or speaker out

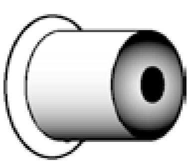


FIGURE 4.17
A phone (RCA) jack used in unbalanced equipment.

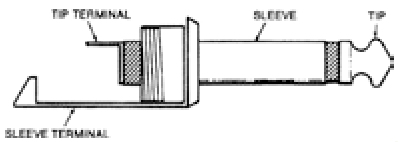


FIGURE 4.18
A mono (TS) 1/4-inch phone plug.

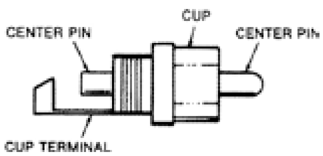


FIGURE 4.19
An RCA (phone) plug.

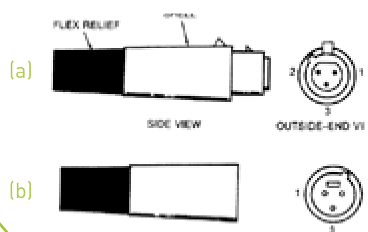


FIGURE 4.20
A 3-pin pro audio connector (XLR-type). (a) female. (b) male.

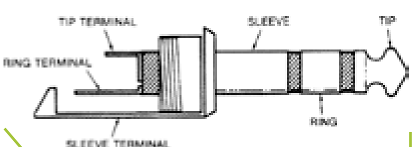


FIGURE 4.21
A stereo (TRS) phone plug.

Equipment connectors are labeled according to their function. If you see an XLR connector with the label "MIC," you know it's a balanced mic input. If it's a 1/8-inch connector on a sound card, look at the icon near the connector. It's either a mic input, line input, line output, or speaker output. You could download the manual for the sound card, which should describe the function of each connector.

CABLE CONNECTORS

Several types of cable connectors are used in audio. Figure 4.18 shows a 1/4-inch mono phone plug (or TS phone plug), used with cables for unbalanced microphones, synthesizers, and electric instruments. The tip terminal is soldered to the cable's center conductor; the sleeve terminal is soldered to the cable shield.

Figure 4.19 shows an RCA or phono plug, used to connect unbalanced line-level signals. The center pin is soldered to the cable's center conductor; the cup terminal is soldered to the cable shield.

Figure 4.20 shows a 3-pin pro audio connector (XLR-type). It is used with cables for balanced mics and balanced recording equipment. The female connector (with holes; Figure 4.20A) plugs into equipment outputs. The male connector (with pins; Figure 4.20B) plugs into equipment inputs. Pin 1 is soldered to the cable shield, pin 2 is soldered to the "hot" red or white lead, and pin 3 is soldered to the remaining lead. This wiring applies to both female and male connectors.

Figure 4.21 shows a stereo (TRS) phone plug, used with stereo headphones and with some balanced line-level cables. For headphones, the tip terminal is soldered to the left-channel lead, the ring terminal is soldered to the right-channel lead, and the sleeve terminal is soldered to the common lead. For balanced line-level cables, the sleeve terminal is soldered to the shield, the tip terminal is soldered to the hot red or white lead, and the ring terminal is soldered to the remaining lead.

Some mixers have insert jacks that are stereo phone jacks; each jack accepts a stereo phone plug. The tip is the send signal to an audio device input, the ring is the return signal from the device output, and the sleeve is the ground.

CABLE TYPES

Cables are also classified according to their function. In a studio, you'll use several types of cables: power, mic, MIDI, speaker, USB, FireWire, S/PDIF, TASCAM TDIF, Alesis Lightpipe, guitar cords, and patch cords.

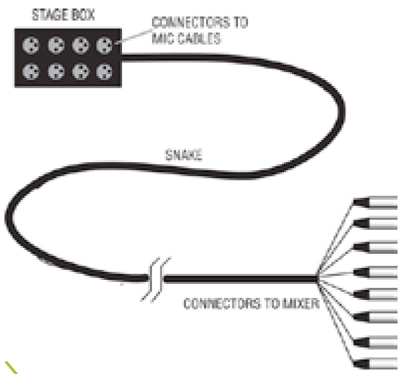


FIGURE 4.22
A stage box and snake.

A power cable, such as an AC extension cord or a power cord on a device, is made of three heavy-gauge wires surrounded by an insulating jacket. The wires are thick to handle high current without overheating.

A mic cable is usually 2-conductor shielded. It has two wires to carry the signal, surrounded by a fine-wire cylinder or shield that reduces hum pickup. On one end of the cable is a connector that plugs into the microphone, usually a female XLR-type. On the other end is either a 1/4-inch phone plug or a male XLR-type connector that plugs into your mixer or audio interface.

Rather than running several mic cables to your mixer or interface, you might consider using a snake, which is a box with multiple mic connectors, all wired to a thick multiconductor cable **(Figure 4.22)**. A snake is especially convenient if you're running long cables to recording equipment from another room. It's essential for most on-location recording.

Professional balanced equipment is interconnected with mic cable: 2-conductor shielded cable having a female XLR on one end and a male XLR on the other. Professional patch bays (described later) use balanced cables with tip-ring-sleeve phone plugs.

A MIDI cable uses a 5-pin DIN connector on each end of a 2-conductor shielded cable. The cable connects MIDI OUT to MIDI IN, or MIDI THRU to MIDI IN.

A speaker cable connects a power amp to each loudspeaker. To avoid wasting power, speaker cables should be as short as possible and should be heavy gauge (between 12 and 16 gauge). They can even be made from lamp cord (zip cord). Number 12 gauge is thicker than 14; 14 is thicker than 16.

A USB cable or a FireWire cable connects a peripheral device (like an audio interface) to a computer.

An S/PDIF cable transfers a digital signal from one device's S/PDIF output to another device's S/PDIF input. It uses a shielded unbalanced cable (ideally a 75-ohm RG59 video cable) with an RCA plug on each end.

A TASCAM TDIF cable is a multiconductor cable with a 25-pin D-sub connector on both ends. It's used to connect multiple digital-audio signals from TASCAM multitrack recorders to digital mixers or computer TDIF interfaces.

An Alesis Lightpipe cable is a fiber optic cable with a TOSLINK connector on both ends. This cable is used to connect 8 channels of digital-audio signals from an Alesis multitrack recorder to a digital mixer or computer Lightpipe interface.

A guitar cord is constructed of a 1-conductor shielded cable with a 1/4-inch phone plug on each end. It connects a guitar amp to an electric instrument (electric guitar, electric bass, synthesizer, or an acoustic guitar with a pickup). Also, it connects an electric instrument to a direct box or to an instrument input.

Patch cords connect your recorder-mixer to external devices: an effects unit, a 2-track recorder, and a power amplifier. They also connect an analog mixer to the analog inputs and outputs of a multitrack recorder, usually as a snake that combines several cables. An unbalanced patch cord is made of 1-conductor shielded cable with either a 1/4-inch phone plug or a phono (RCA) connector on each end. A stereo patch cord is two patch cords joined side-by-side.

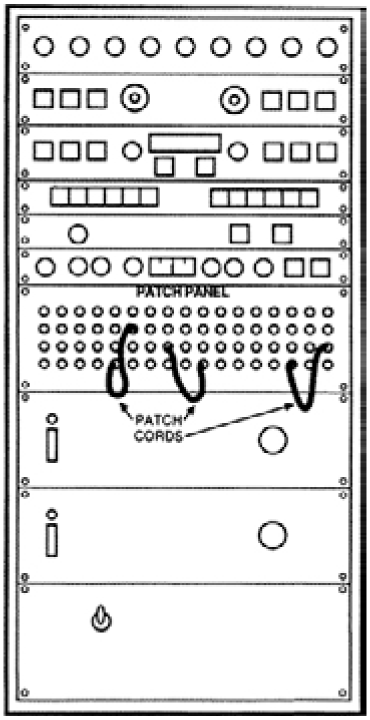


FIGURE 4.23
A rack and patch panel.

RACK/PATCH BAY

You might want to mount your signal processors in a rack, an enclosure made of wood or metal with rails of screw holes for mounting equipment (Figure 4.23). You also might want to install a patch panel or patch bay, a group of connectors that are wired to equipment inputs and outputs. Using a patch bay and patch cords, you can change equipment connections easily. You also can bypass or patch around defective equipment. Note that a patch bay increases the chance of hum pickup slightly because of the additional cables and connectors. Racks and patch bays are not essential, but they are convenient. Figure 4.24 shows some typical patch-bay assignments.

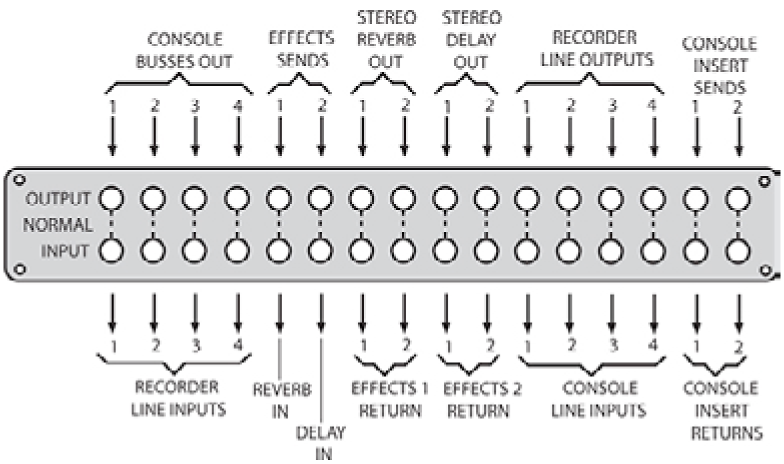


FIGURE 4.24
Some typical patch-bay assignments.

EQUIPMENT CONNECTIONS

The instruction manuals of your equipment tell you how to connect each component to the others. In general, use cables that are as short as possible to reduce hum, but that are long enough to let you make changes. Be sure to label all your cables on both ends according to what they plug into—for example, MIXER CH1 MONITOR OUT or ALESIS 3630 IN. If you change connections temporarily, or the cable becomes unplugged, you'll know where to plug it back in. A piece of masking tape folded over the end of the cable makes a stay-put label.

Let's say you have a hardware mixer in your recording setup. Here's a typical way to hook up the gear **(see Figure 4.25)**:

1. Plug the AC power cords of audio equipment and electric musical instruments into AC outlet strips fed from the same circuit breaker. Make sure that the sum of the equipment current ratings does not exceed the breaker's amp rating for that outlet. Plug the power amplifier or powered speakers into their own outlet on the same breaker so that they receive plenty of current. Consider using an AC power conditioner such as made by Furman (www.furmansound.com). It provides clean, steady AC power to sensitive electronic equipment. Surge protection and noise filtering are included.
2. Connect mic cables to mics. Use mic cables with a male XLR connector on one end and a female XLR connector on the other end.
3. Connect mic cables to the female XLR connectors in either the snake junction box, or directly into mic inputs on a mixer or mic preamps. Plug the snake connectors into the mic inputs. If your mixer has phone-jack mic inputs, you may need to use an impedance-matching adapter (female XLR to phone) between the mic cable and the mic input jack **(Figure 4.26)**.
4. Set the output volume of synthesizers and sound modules about three-quarters up. Using a guitar cord, connect their audio outputs to instrument or line inputs on your mixer. If this causes hum, use a direct box. Using a MIDI cable, connect the MIDI OUT of a MIDI controller to the MIDI IN of your audio interface or MIDI interface.
5. If you are recording a guitar direct, connect its cord either to (1) an instrument input on your mixer or audio interface (1/4-inch phone jack), or (2) a direct box. Connect the XLR output of the direct box to a mic input on your mixer or audio interface.

6. If the mixer is a standalone unit (not part of a recorder-mixer), connect the mixer's stereo line outputs to the inputs of an audio interface. Use a stereo RCA-to-RCA cable or two phone-to-phone cables. If the mixer has a USB or FireWire connector, connect that to the mating connector in your computer—you don't need an audio interface.

7. Connect the audio interface line outputs to the mixer's 2-track or tape inputs, or directly to powered speakers. Use a stereo RCA-to-RCA cable or two phone-to-phone cables. Again, if the mixer has a USB or FireWire connector, connect that to the mating computer connector and omit the audio interface.

8. Connect the mixer's monitor outputs to the power-amp inputs. Connect the power-amp outputs to loudspeakers. Or if you are using powered (active) monitors, connect the mixer monitor outputs to the monitor-speaker inputs.

9. If the mixer does not have internal effects, connect the mixer aux-send connectors to effects inputs (not shown). Connect the effects outputs to the mixer aux-return or bus-in connectors. Use phone-to-phone cables.

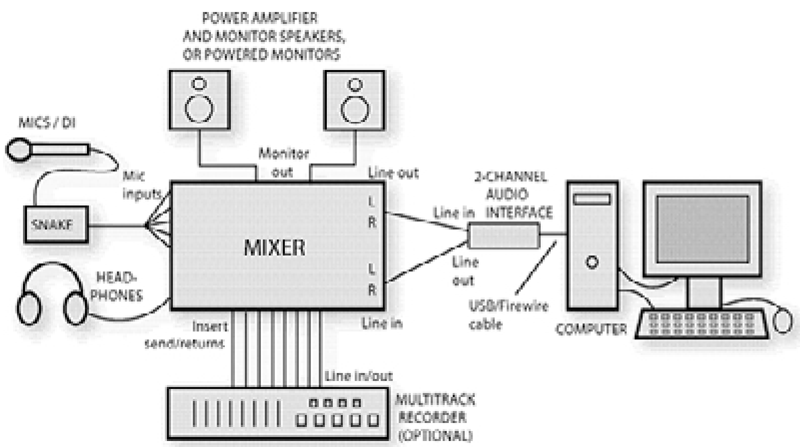


FIGURE 4.25
Typical connections for a mixer-based studio.

FIGURE 4.26
A female XLR to phone plug adapter, such as Radio Shack #274-016.

Check your equipment manuals to determine their input and output levels. When you connect devices that run at different levels, set the +4/–10 switch on each unit to match the levels. If there is no such switch on either device, connect between them a +4/–10 converter box such as the Whirlwind LM2U line level converter (www.whirlwindusa.com).

Figure 4.27 shows typical connections in a DAW recording studio with a multichannel audio interface. Figure 4.28 shows a typical layout for a DAW recording studio.

As shown in Figure 4.27, you might connect the equipment like this:

1. Using a guitar cord, connect electric instruments to instrument inputs on the audio interface. If an instrument is more than about 15 feet from the interface, connect its output to a direct box (using guitar cords), and connect the direct box XLR output to a snake or to an audio interface mic input.
2. Using an XLR mic cable, connect each mic to a mic input on the audio interface. If the mics are more than about 15 feet from the interface, connect each mic to a snake box, and connect the snake XLR connectors to the interface mic inputs. If you prefer to use a separate mic preamp and A/D converter, plug the mic into the preamp, and connect the preamp's line output to the A/D converter's line input using an XLR or phone-to-phone cable.
3. Using a MIDI cable, connect the MIDI OUT of a MIDI controller to the MIDI IN of the audio interface.
4. Using two phone-to-phone cables (stereo or mono), connect the stereo output of the interface to two powered monitors.

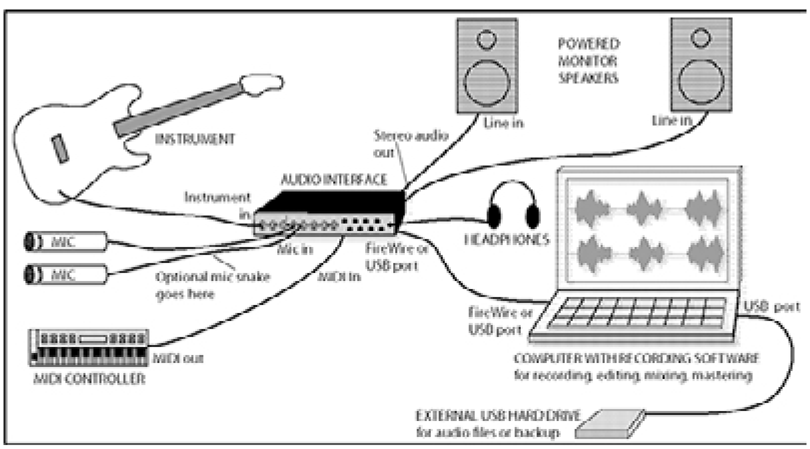


FIGURE 4.27
Typical connections in a DAW recording studio.

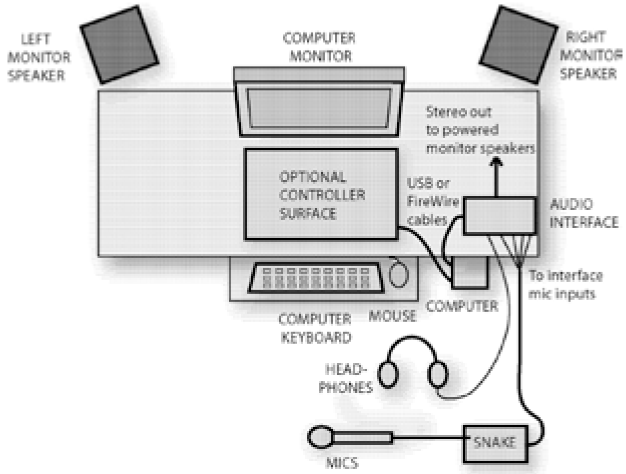


FIGURE 4.28
Typical layout of a DAW recording studio.

- To prevent ground loops, plug all equipment into outlet strips powered by the same AC outlet or AC circuit.
- Do not use an AC (electrical) 3-to-2 adapter to disconnect the power ground—it causes a safety hazard.
- Some power amps create hum if they don't get enough AC current. So connect the power amp (or powered speakers) AC plug to its own wall outlet socket—the same outlet that feeds the outlet strips for the recording equipment.
- If possible, use balanced cables going into balanced equipment. Balanced cables have XLR or TRS connectors and two conductors surrounded by a shield. At both ends of the cable, connect the shield to a screw in the chassis, not to XLR pin 1. Or use audio gear whose XLR connectors are wired with pin 1 to chassis ground, not to signal ground.
- Transformer-isolate unbalanced connections.
- Don't use conventional SCR dimmers to change the studio lighting levels. Use Luxtrol® variable-transformer dimmers or multiway incandescent bulbs instead.

Even if your system is wired properly, a hum or buzz may appear when you make a connection. Follow these tips to stop the hum:

- If the hum is coming from a direct box, flip its ground-lift switch.
- Check cables and connectors for broken leads and shields.

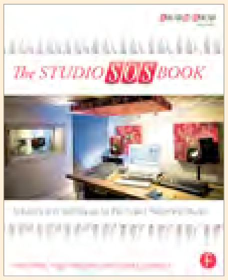
- Unplug all equipment from each other. Start by listening just to the powered monitor speakers. Connect a component to the system one at a time, and see when the hum starts.
- Remove audio cables from your devices and monitor each device by itself. It may be defective.
- Lower the volume on your power amp (or powered speakers), and feed them a higher-level signal.
- Use a direct box instead of a guitar cord between instrument and mixer.
- To stop a ground loop when connecting two devices, connect between them a 1:1 isolation transformer, direct box, or hum eliminator (such as the Jensen CI-2RR, Behringer HD400, Rolls HE18, or Ebtech He2PKG). See Figures A-3 and A-4 in Appendix A.
- Make sure that the snake box is not touching metal.
- To prevent accidental ground loops, do not connect XLR pin 1 to the connector shell except for permanent connections to equipment inputs and outputs.
- Try another mic.
- If you hear a hum or buzz from an electric guitar, have the player move to a different location or aim in a different direction. You might also attach a wire between the player's body and the guitar strings near the tailpiece to ground the player's body.
- Turn down the high-frequency EQ on a buzzing bass guitar track.
- To reduce buzzing between notes on an electric-guitar track, apply a noise gate.
- Route mic cables and patch cords away from power cords; separate them vertically where they cross. Also keep recording equipment and cables away from computer monitors, power amplifiers, and power transformers.
- See Rane's excellent article on sound system interconnections at www.rane.com.

By following all these tips, you should be able to connect audio equipment without introducing any hum. Good luck!

REDUCING RADIO FREQUENCY INTERFERENCE (RFI)

RFI is heard as buzzing, clicks, radio programs, or "hash" in the audio signal. It's caused by CB transmitters, computers, lightning, radar, radio and TV transmitters, industrial machines, auto ignitions, stage lighting, and other sources. Many of the following techniques are the same used to reduce hum from other sources. To reduce RFI:

- If you think that a speaker cable, mic cable, or patch cord is picking up RFI, wrap the cable several times around an RFI choke (available at Radio Shack). Put the choke near the device that is receiving audio.
- Install high-quality RFI filters in the AC power outlets. The cheap types available from local electronics shops are generally ineffective.
- If the shield is disconnected in a balanced cable connector, solder it to pin 1.
- If a mic is picking up RFI, solder a 0.047 microfarad capacitor between pin 1 and 2, and between pin 1 and 3, in the female XLR connector of the mic cable. Caution: this might cause high-frequency distortion with some mics.
- At equipment XLR mic inputs, use a cable connector with a tab that is connected to the metal shell. Use a wide strip of metal braid to connect this tab to pin 1. Make sure the connector makes metal-to-metal contact with the chassis.
- Periodically clean connector contacts with Caig Labs DeoxIT, or at least unplug and plug them in several times.



The following is excerpted from *The Studio SOS Book* by Paul White, Hugh Robjohns and Dave Lockwood.

You can purchase a copy HERE.

CASE STUDY ■ SOUND ON SOUND STUDIO SOS

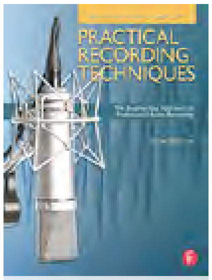
We were asked to sort out a hum problem in a very simple recording setup comprising a Zoom hard disk recorder and an old Joe Meek compressor. The problem here was that whenever the owner plugged his guitar into the compressor to record it onto the Zoom multitrack machine he was greeted by a huge level of hum and other unwanted interference noises.

The problem turned out to be almost the opposite of a ground loop: there was no ground at all! Everything in his recording system was running from double-insulated (Class II) power supplies (wall warts and line lumps!), which incorporate isolating transformers, without a mains-ground connection anywhere. This meant that the whole system was 'floating' and the screens of all the connecting cables were effectively acting as radio aerials, picking up all sorts of unwanted rubbish from the ether and passing it all directly into the audio electronics.

If everything in your recording system uses wall-wart or line-lump double-insulated (Class II) power supplies, which incorporate isolating transformers, you can end up without a mains-ground connection at all!, resulting in excessive hum. The problem can be solved using a dedicated grounding mains plug, such as the one opposite (note the model shown is for a UK 13A socket).

The solution was to attach a length of wire to the metalwork of the Zoom recorder via one of the screws on the underside, and to ground the other end to the metal chassis of a piece of household equipment that was properly grounded – in this case, a nearby hi-fi system. As soon as we did that the hum problem vanished entirely because the cable screens were then able to act as proper screens, trapping unwanted interference and passing it to ground rather than into the audio circuitry!

In theory you could create a ground cable by wiring only a ground wire to the earth pin of a standard three-pin plug, but we certainly don't recommend this as tripping over the cable might just disconnect it from the ground pin in the plug and allow it to touch the live pin, which would then cause all the connected equipment to become electrically live. A far safer solution is to buy a dedicated grounding mains plug fitted with a metal ground pin and plastic dummy pins for the live and neutral. Although these are not stocked by typical DIY stores, an on-line search for "grounding mains plug" brought up several sources.



The following is excerpted from *Practical Recording Techniques* by Bruce Bartlett and Jenny Bartlett.

You can purchase a copy HERE.

What microphone is best for recording an orchestra? What's a good snare mic? Should the microphone be a condenser or dynamic, omni or cardioid?

You can answer these questions more easily once you know the types of microphones and understand their specs. First, it always pays to get a high-quality microphone. The mic is a source of your recorded signal. If that signal is noisy, distorted, or tonally colored, you'll be stuck with those flaws through the whole recording process. Better get it right up front.

Even if you have a MIDI studio and get your sounds from samples or synthesizers, you still might need a good microphone for sampling, or to record vocals, sax, acoustic guitar, and so on.

A microphone is a transducer—a device that changes one form of energy into another. Specifically, a mic changes sound into an electrical signal. Your mixer amplifies and modifies this signal.

TRANSDUCER TYPES

Mics for recording can be grouped into three types depending on how they convert sound to electricity: dynamic, ribbon, or condenser. A dynamic mic capsule, or transducer, is shown in Figure 6.1. A coil of wire attached to a diaphragm is suspended in a magnetic field. When sound waves vibrate the diaphragm, the coil vibrates in the magnetic field and generates an electrical signal similar to the incoming sound wave. Another name for a dynamic mic is moving-coil mic, but this term is seldom used. In a ribbon mic capsule, a thin metal foil or ribbon is suspended in a magnetic field (Figure 6.2). Sound waves vibrate the ribbon in the field and generate an electrical signal.

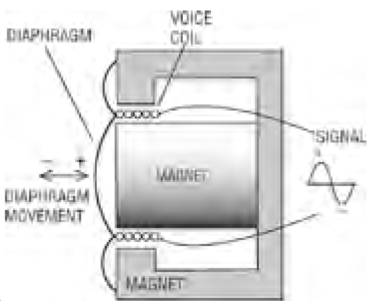


FIGURE 6.1
A dynamic transducer.

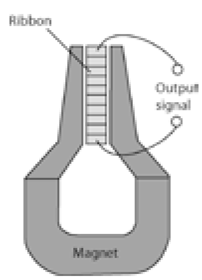


FIGURE 6.2
A ribbon transducer.

A condenser or capacitor mic capsule has a conductive diaphragm and a metal backplate placed very close together (Figure 6.3). They are charged with static electricity to form two plates of a capacitor. When sound waves strike the diaphragm, it vibrates. This varies the spacing between the plates. In turn, this varies the capacitance and generates a signal similar to the incoming sound wave. Because of its lower diaphragm mass and higher damping, a condenser mic responds faster than a dynamic mic to rapidly changing sound waves (transients).

Two types of condenser mic are true condenser and electret condenser. In a true condenser mic (externally biased mic), the diaphragm and backplate are charged with a voltage from a circuit built into the mic. In an electret condenser mic, the

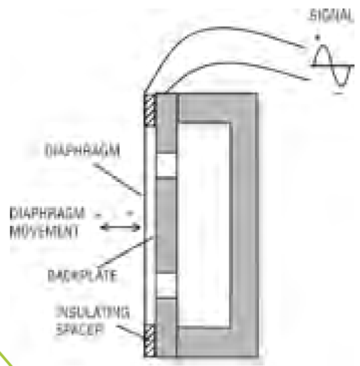


FIGURE 6.3
A condenser transducer.

diaphragm and backplate are charged by an electret material, which is in the diaphragm or on the backplate. Electrets and true condensers can sound equally good, although some engineers prefer true condensers, which tend to cost more.

A condenser mic needs a power supply to operate, such as a battery or phantom power supply. Phantom power is 12 to 48 volts DC applied to pins 2 and 3 of the mic connector through two equal resistors. The microphone receives phantom power and sends audio signals on the same two conductors. Ground for the phantom power supply is through the cable shield. Nearly all mixing consoles and audio interfaces supply phantom power at their mic input connectors. You simply plug the mic into the mixer to power it. Phantom power is explained in detail in Appendix D.

Dynamics and ribbons need no power supply. You can plug these types of mics into a phantom supply without damage because the voice coil or ribbon is not connected to ground (unless they are accidentally shorted to the mic housing, or one cable conductor is shorted to the shield). However, because damage is possible in theory, most ribbon mic manufacturers recommend that phantom power be turned off unless you're using one of the newer ribbon mics that have a built-in preamp that is phantom powered.

Figure 6.4 shows a cutaway view of a typical dynamic vocal mic and condenser instrument mic.

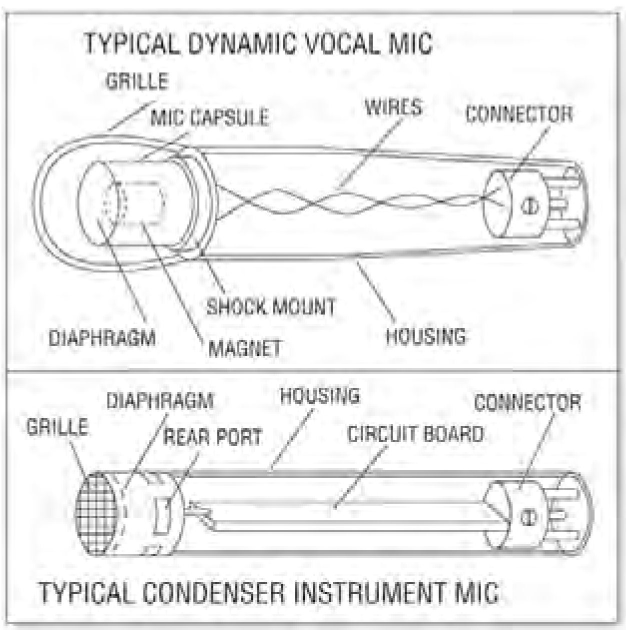


FIGURE 6.4
Inside a typical dynamic vocal mic and condenser instrument mic.

GENERAL TRAITS OF EACH TRANSDUCER TYPE

Condenser

- Wide, smooth frequency response
- Detailed sound, extended highs
- Omni type has excellent low-frequency response
- Transient attacks sound sharp and clear
- Preferred for acoustic instruments, cymbals, studio vocals
- Can be miniaturized

Dynamic

- Tends to have rougher response, but still quite usable
- Rugged and reliable
- Handles heat, cold, and high humidity
- Handles high volume without distortion
- Preferred for guitar amps and drums
- If flat response, can take the "edge" off woodwinds and brass

Ribbon

- Prized for its warm, smooth tone quality
- Delicate
- Complements digital recording
- Preferred for horns and guitar amps

There are exceptions to the tendencies listed above. Some dynamics have a smooth, wide-range frequency response. Some condensers are rugged and handle high SPLs. It depends on the specs of the particular mic.

Audio clip 13 at www.taylorandfrancis.com/cw/bartlett-9780240821535 demonstrates the sound of each transducer type.

POLAR PATTERN

Microphones also differ in the way they respond to sounds coming from different directions. An omnidirectional microphone is equally sensitive to sounds arriving from all directions. A unidirectional mic is most sensitive to sound arriving from one direction—in front of the mic—but softens sounds entering the sides or rear of the mic. A bidirectional mic is most sensitive to sounds arriving from two directions—in front of and behind the mic—but rejects sounds entering the sides.

There are three types of unidirectional patterns: cardioid, supercardioid, and hypercardioid. A mic with a cardioid pattern is sensitive to sounds arriving from a broad angle in front of the mic. It is about 6 dB less sensitive at the sides, and about 15 to 25 dB less sensitive in the rear. The supercardioid pattern is 8.7 dB less sensitive at the sides and has two areas of least pickup at 125 degrees away from the front. The hypercardioid pattern is 12 dB less sensitive at the sides and has two areas of least pickup at 110 degrees away from the front.

To hear how a cardioid pickup pattern works, talk into a cardioid mic from all sides while listening to its output. Your reproduced voice is loudest when you talk into the front of the mic, and softest when you talk into the rear.

Play audio clip 14 at www.taylorandfrancis.com/cw/bartlett-9780240821535.

The super- and hypercardioid reject sound from the sides more than the cardioid. They are more directional, but they pick up more sound from the rear than the cardioid does.

A microphone's polar pattern is a graph of its sensitivity versus the angle at which sound approaches the mic. The polar pattern is plotted on polar graph paper. Sensitivity is plotted as distance from the origin. **Figure 6.5** shows various polar patterns.

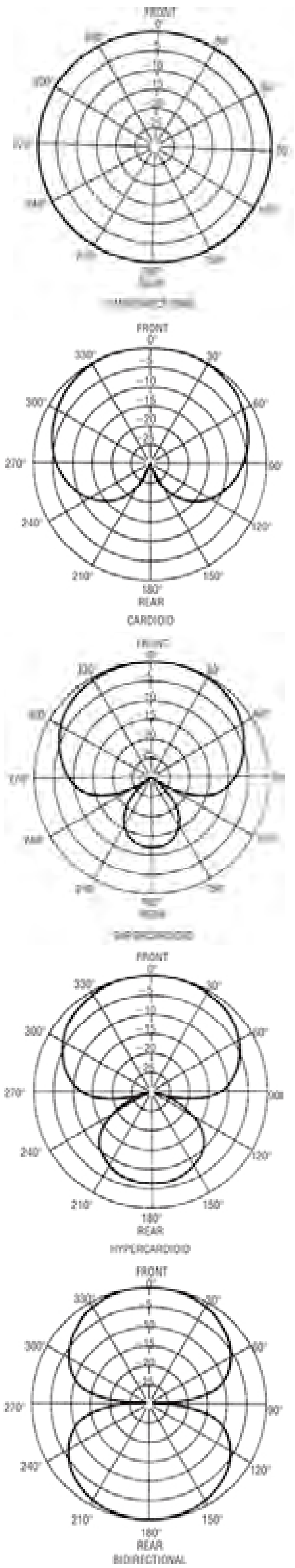


FIGURE 6.5
Various polar patterns. Sensitivity is plotted vs. angle of sound incidence.

TRAITS OF EACH POLAR PATTERN

Omnidirectional

- All-around pickup
- Most pickup of room reverberation (*play audio clip 14 at www.taylorandfrancis.com/cw/bartlett-9780240821535*)
- Not much isolation unless you mike close
- Low sensitivity to pops (explosive breath sounds)
- Low handling noise
- No up-close bass boost (proximity effect)
- Extended low-frequency response in condenser mics—great for pipe organ or bass drum in an orchestra or symphonic band
- Lower cost in general

Unidirectional (cardioid, supercardioid, hypercardioid)

- Selective pickup
- Rejection of room acoustics, background noise, and leakage
- Good isolation—good separation between tracks
- Up-close bass boost (except in mics that have holes in the handle)
- Better gain-before-feedback in a sound-reinforcement system
- Coincident or near-coincident stereo miking

Cardioid

- Broad-angle pickup of sources in front of the mic
- Maximum rejection of sound approaching the rear of the mic
- Most popular pattern

Supercardioid

- Maximum difference between front hemisphere and rear hemisphere pickup (good for stage-floor miking)
- More isolation than a cardioid
- Less reverb pickup than a cardioid

Hypercardioid

- Maximum side rejection in a unidirectional mic
- Maximum isolation—maximum rejection of reverberation, leakage, feedback, and background noise

Bidirectional

- Front and rear pickup, with side sounds rejected (for across-table interviews or two-part vocal groups, for example)
- Maximum isolation of an orchestral section when miked overhead
- Blumlein stereo miking (two bidirectional mics crossed at 90 degrees)

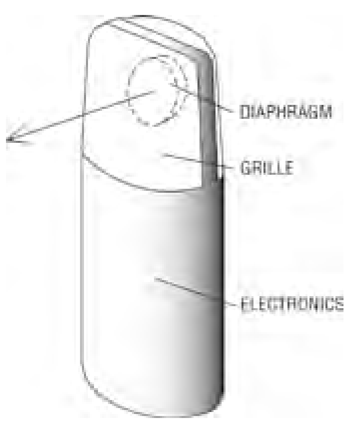


FIGURE 6.6
A typical multipattern mic that is side-addressed.

In a good mic, the polar pattern should be about the same from 200 Hz to 10 kHz. If not, you'll hear off-axis coloration: the mic will have a different tone quality on and off axis. Small-diaphragm mics tend to have less off-axis coloration than large-diaphragm mics. You can get either the condenser or dynamic type with any kind of polar pattern (except bidirectional dynamic). Ribbon mics are either bidirectional or hypercardioid.

Some condenser mics come with switchable patterns. Note that the shape of a mic does not indicate its polar pattern.

If a mic is end-addressed, you aim the end of the mic at the sound source. If a mic is side-addressed, you aim the side of the mic at the sound source. **Figure 6.6** shows a typical side-addressed condenser mic with switchable polar patterns.

Boundary mics that mount on a surface have a pattern that is half-omni (hemispherical), half-supercardioid, or half-cardioid (like an apple sliced in half through its stem). The boundary mounting makes the mic more directional so it picks up less room acoustics.

FREQUENCY RESPONSE

As with other audio components, a microphone's frequency response is the range of frequencies that it will reproduce at an equal level (within a tolerance, such as ±3 dB).

The following is a list of sound sources and the microphone frequency response that is adequate to record the source with high fidelity. A wider range response works, too.

- Most instruments: 80 Hz to 15 kHz
- Bass instruments: 40 Hz to 9 kHz
- Brass and voice: 80 Hz to 12 kHz
- Piano: 40 Hz to 12 kHz
- Cymbals and some percussion: 300 Hz to 15 or 20 kHz
- Orchestra or symphonic band: 40 Hz to 15 kHz

If possible, use a mic with a response that rolls off below the lowest fundamental frequency of the instrument you're recording. For example, the frequency of the low-E string on an acoustic guitar is about 82 Hz. A mic used on the acoustic guitar should roll off below that frequency to avoid picking up low-frequency noise such as rumble from trucks and air conditioning. Some mics have a built-in low-cut switch for this purpose. Or you can filter out the unneeded lows at your mixer.

A frequency-response curve is a graph of the mic's output level in dB at various frequencies. The output level at 1 kHz is placed at the 0 dB line on the graph, and the levels at other frequencies are so many decibels above or below that reference level.

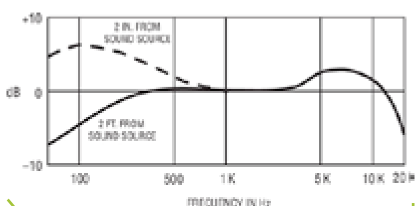


FIGURE 6.7
An example of the frequency response of a microphone with proximity effect and a presence peak around 5 kHz.

The shape of the response curve suggests how the mic sounds at a certain distance from the sound source. (If the distance is not specified, it's probably 2 to 3 feet.) For example, a mic with a wide, flat response reproduces the fundamental frequencies and harmonics in the same proportion as the sound source. So a flat-response mic tends to provide accurate, natural reproduction at that distance.

A rising high end or a "presence peak" around 5 to 10 kHz sounds more crisp and articulate because it emphasizes the higher harmonics **(Figure 6.7)**. Play audio clip 15 at www.taylorandfrancis.com/cw/bartlett-9780240821535. Sometimes this type of response is called tailored or contoured. It's popular for guitar amps and drums because it adds punch and emphasizes attack. Some microphones have switches that alter the frequency response.

Most uni- and bidirectional mics boost the bass when used within a few inches of a sound source. You've heard how the sound gets bassy when a vocalist sings right into the mic. This low-frequency boost related to close mic placement is called the proximity effect, and it's often plotted on the frequency-response graph. Omni mics have no proximity effect; they sound tonally the same at any distance.

The warmth created by proximity effect adds a pleasing fullness to drums. In most recording situations, though, the proximity effect lends an unnatural boomy or bassy sound to the instrument or voice picked up by the mic. Some mics—multiple-D or variable-D types—are designed to reduce it. These types have holes or slots in the mic handle. Some mics have a bass-rolloff switch to compensate for the bass boost. Or you can roll off the excess bass with your mixer's equalizer until the sound is natural. By doing so, you also reduce low-frequency leakage picked up by the microphone.

Note that mic placement can greatly affect the recorded tone quality. A flat-response mic does not always guarantee a natural sound because mic placement has such a strong influence.

IMPEDANCE (Z)

This spec is the mic's effective output resistance at 1 kHz. A mic impedance between 150 and 600 ohms is low; 1000 to 4000 ohms is medium; and above 25 kilohms is high.

Always use low-impedance mics. If you do, you can run long mic cables without picking up hum or losing high frequencies. The input impedance of a mixer mic input is about 1500 ohms. If it were the same impedance as the mic, about 250 ohms, the mic would "load down" when you plug it in. Loading down a mic makes it lose level, distort, or sound thin. To prevent this, a mic input has an impedance much higher than that of the microphone. But it's still called a low-Z input.

MAXIMUM SPL

To understand this spec, first we need to understand sound pressure level (SPL). It is a measure of the intensity of a sound. The quietest sound we can hear, the threshold of hearing, is 0 dBSPL. Normal conversation at 1 foot measures about 70 dBSPL; painfully loud sound is above 120 dBSPL.

If the maximum SPL spec is 125 dBSPL, the mic starts to distort when the instrument being miked is putting out 125 dBSPL at the mic. A maximum SPL spec of 120 dB is good, 135 dB is very good, and 150 dB is excellent.

Dynamic mics tend not to distort, even with very loud sounds. Some condensers are just as good. Some have a pad you can switch in to prevent distortion in the mic circuitry. Because a mic pad reduces signal-to-noise ratio (S/N), use it only if the mic distorts.

SENSITIVITY

This spec tells how much output voltage a mic produces when driven by a certain SPL. A high-sensitivity mic puts out a stronger signal (higher voltage) than a low-sensitivity mic when both are exposed to an equally loud sound.

A low-sensitivity mic needs more mixer gain than a high-sensitivity mic. More gain usually results in more noise. When you record quiet music at a distance (classical guitar, string quartet), use a mic of high sensitivity to override mixer noise. When you record loud music or mike close, sensitivity matters little because the mic signal level is well above the mixer noise floor. That is, the S/N is high. Listed below are typical sensitivity specs for three transducer types:

- Condenser: 5.6 mV/Pa (high sensitivity)
- Dynamic: 1.8 mV/Pa (medium sensitivity)
- Ribbon or small dynamic: 1.1 mV/Pa (low sensitivity)

The louder the sound source, the higher the signal voltage the mic puts out. A very loud instrument, such as a kick drum or guitar amp, can cause a microphone to generate a signal strong enough to overload the mic preamp in your mixer. That's why most mixers have pads or input-gain controls—to prevent preamp overload from hot mic signals.

SELF-NOISE

Self-noise or equivalent noise level is the electrical noise or hiss a mic produces. It's the dBSPL of a sound source that would produce the same output voltage that the noise does.

Usually the self-noise spec is A-weighted. That means the noise was measured through a filter that makes the measurement correlate more closely with the annoyance value. The filter rolls off low and high frequencies to simulate the frequency response of the ear.

An A-weighted self-noise spec of 14 dBSPL or less is excellent (quiet), 21 dB is very good, 28 dB is good, and 35 dB is fair—not good enough for quality recording.

A mic with a self-noise spec around 30 dB can sound noise-free if it is used very close to a sound source—for example, clipped onto an acoustic guitar or fiddle.

Because a dynamic mic has no active electronics to generate noise, it has very low selfnoise (hiss). So most spec sheets for dynamic mics do not specify self-noise.

SIGNAL-TO-NOISE RATIO

This is the difference in decibels between the SPL in dB and a mic's self-noise in dB. The higher the SPL of the sound source at the mic, the higher the S/N. Given an SPL of 94 dB, an S/N spec of 74 dB is excellent; 64 dB is good. The higher the S/N ratio, the cleaner (more noise-free) the signal, and the greater the "reach" of the microphone.

Reach is the clear pickup of quiet, distant sounds due to high S/N. Reach is not specified in data sheets because any mic can pick up a source at any distance if the source is loud enough. For example, even a cheap mic can reach several miles if the sound source is a thunderclap.

POLARITY

The polarity spec relates the polarity of the electrical output signal to the acoustic input signal. The standard is "pin 2 hot." That is, the mic produces a positive voltage at pin 2 with respect to pin 3 when the sound pressure pushes the diaphragm in (positive pressure).

Be sure that your mic cables do not reverse polarity. On both ends of each cable, the wiring should be pin 1 shield, pin 2 red, pin 3 white or black. Or the wiring on both ends should be pin 1 shield, pin 2 white, pin 3 black. If some mic cables are correct polarity and some are reversed, and you mix their mics to mono, the bass may cancel.

MICROPHONE TYPES

The following sections describe several types of recording mics.

LARGE-DIAPHRAGM CONDENSER MICROPHONE (LDC)

This is a condenser microphone, usually side-addressed, with a diaphragm 1 inch or larger in diameter **(Figure 6.6)**. It generally has very good low-frequency response and low self-noise. Common uses are studio vocals and acoustic instruments. Examples are the AKG C12VR; C414; Perception 100 and 200; C2000B and C3000B; Apex 530; Audio-Technica AT2020/3035/4040; Audix SCX25; Blue Blueberry; Blue Reactor; CAD Equitek Series, GXL Series and M177; DPA 4041; Karma Mics Unity; Lawson L47MP MKII and L251; Mojave Audio MA-301 fet; Manley Gold Reference; Neumann U87; U47, and TLM 103; Sennheiser MK 4; Soundelux Elux 251; ADK various models; SE Electronics various models; Shure KSM Series; Shure Beta 181; MXL V67G, V69, 900, 2001, 2003, and 2006; Rode NT1A; Studio Projects B and C Series; Samson CL7 and C01; Nady SCM 950 and 100; Violet Flamingo; M-Audio Solaris, Sputnik, Luna, and Nova; and Behringer B1 and B2.

SMALL-DIAPHRAGM CONDENSER MICROPHONE (SDC)

This is a stick-shaped or "pencil" condenser microphone, usually cardioid and end-addressed, with a diaphragm under 1 inch in diameter **(Figure 6.4)**. It generally has very good transient response and detail, making it a fine choice for close miking acoustic instruments—especially cymbals, acoustic guitar, and piano. Examples are the AKG C 451 B; Audio-Technica AT 3031 and AT 4051a; Audix SCX1, ADX50, and ADX51; Berliner CM-33, CAD Equitek e60; M-Audio Pulsar; Samson C02; Crown CM-700; DPA 4006; Mojave MA-100 and MA-101SP; Neumann KM 184; Sennheiser e614 and MKH50; Shure KSM109/SL, KSM137/Sl, and SM81; MXL 600 and 603S; Behringer B5 with cardioid capsule; and Studio Projects C4.

DYNAMIC INSTRUMENT MICROPHONE

This is a stick-shaped dynamic microphone, end-addressed **(Figure 6.4)**. Although it may have a fl at response, it generally has a presence peak and some low-frequency rolloff to prevent boominess when used up close. It's often used on drums and guitar amps. Examples include the Shure SM57; AKG D112 (kick drum); Audio-Technica AT AE2500 (kick); Beyerdynamic M-99; Electro-Voice N/D868 (kick); Heil PR40 (kick); Audix D2, D3, D6, and i5; and Sennheiser MD421, e604 and e602 (kick). A drum microphone kit is a collection of microphones for a drum set, such as the Audix DP QUAD mic pack.

LIVE-VOCAL MICROPHONE

This unidirectional mic is shaped like an ice-cream cone because of its large grille used to reduce breath pops. It can be a condenser, dynamic, or ribbon type, and it usually has a presence peak and some low-frequency rolloff. Examples are the Shure SM58; Beta 58A, SM85, SM86, SM87A, and KSM9; AKG D3800; C 535 EB, C5, and D 5/D 5S; Audix OM5 and OM7; Beyerdynamic M88 TG and TG V70d; CAD D189; EV N-Dym Series; Sennheiser e945; and Neumann KMS104 and KMS105.

RIBBON MICROPHONE

This mic can be side- or end-addressed. It generally is used wherever you want a warm, smooth tone quality (sometimes with reduced highs). Examples include models by Beyerdynamic, Coles, Royer, Cascade, Audio-Technica, Shure, Blue, and AEA.

BOUNDARY MICROPHONE

Boundary mics are designed to be used on surfaces. Tape them to the underside of a piano lid, or tape them to the wall for pickup of room ambience. They can be used on hard baffles between instruments, or on panels to make the mics directional. A boundary mic uses a mini condenser mic capsule mounted very near a sound-reflecting plate or boundary **(Figure 6.8)**. Because of this construction, the mic picks up direct sound and reflected sound at the same time, in phase at all frequencies. So you get a smooth response free of phase cancellations. A conventional mic near a surface sounds colored; a boundary mic on a surface sounds natural. Other benefits are a wide, smooth frequency response free of phase cancellations, excellent clarity and reach, and the same tone quality anywhere around the mic.

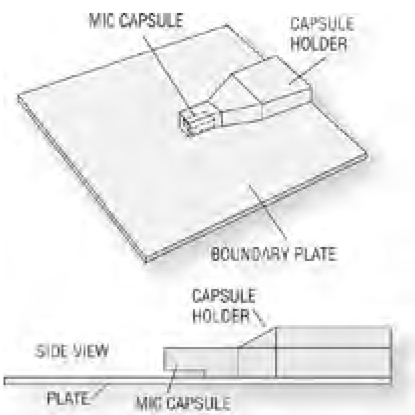


FIGURE 6.8
Typical PZM construction.

Examples of half-omnidirectional or hemispherical boundary mics include the AKG CBL 99, Audio-Technica models, Beyerdynamic models, Crown PZM-30D and PZM-6D, and Shure Beta 91.

Some boundary mics have a half-cardioid or half-supercardioid polar pattern. They work great on a conference table, or near the front edge of a stage floor to pick up drama or musicals. Examples are the Crown PCC-170, and the Bartlett TM-125, TM-125C, and Floor Recording Mic.

MINIATURE MICROPHONE

Mini condenser mics can be attached to drum rims, flutes, horns, acoustic guitars, fiddles, and so on. Their tone quality is about as good as larger studio microphones

FIGURE 6.9
A mini mic is the size of a penny.

and the price is relatively low. With these tiny units you can mike a band in concert without cluttering the stage with boom stands **(Figure 6.9)**, or you can mike a whole drum set with two or three of these. Although you lose individual control of each drum in the mix, the cost is low and the sound is quite good with some bass and treble boost. Compared to large mics, mini mics tend to have more noise (hiss) in distant-miking applications. A lavalier mic is a mini mic worn on the chest to pick up speech from a newscaster or a wandering lecturer. Examples are the AKG C 516, C 518, and C 519; Bartlett Spark Mini Mic, Guitar Mic, and Fiddle Mic; Shure Beta 98S; Audix M1245 and Micro-D; Countryman Isomax B6; DPA 4060; and Sennheiser e608.

STEREO MICROPHONE

A stereo microphone combines two directional mic capsules in a single housing for convenient stereo recording **(Figure 6.10)**. Simply place the mic a suitable distance and height from the sound source, and you'll get a stereo recording with little fuss. Examples include the Audio-Technica AT2022, AT4050ST, and AT8022; Neumann SM69; Shure VP88; AEA R88 MK2; and Royer SF-12.

FIGURE 6.10
Audio-Technica AT8022, an example of a stereo microphone.

Because there is no spacing between the mic capsules, there also is no delay or phase shift between their signals. Coincident stereo microphones are mono-compatible—the frequency response is the same in mono and stereo—because there are no phase cancellations if the two channels are combined.

DIGITAL MICROPHONE

This condenser microphone has a built-in analog-to-digital converter. It is usually side-addressed, has a large diaphragm, has a flat response, and very low self-noise. Its output is a digital signal, which is immune to picking up hum. An example is the Neumann Solution-D.

HEADWORN MICROPHONE

This microphone is used for a live performance that might be recorded. It is a small condenser mic worn on the head, either omni- or unidirectional. The headworn mic allows the performer freedom of movement on stage. Some models provide excellent gain before feedback and isolation. Examples include the AKG C 520, Audio-Technica BP892 MicroSet, Samson SE50, Sennheiser ME 3-EW, Shure WBH53, Rode HS1-P, Countryman Isomax E6, Crown CM-311A, and DPA 4066F/4088F.

MICROPHONE SELECTION

Table 6.1 is a guide to choosing a mic based on your requirements.

TABLE 6.1 ■ MIC APPLICATION GUIDE

Requirement	Characteristic
Natural, smooth tone quality	Flat frequency response
Bright, present tone quality	Presence peak (emphasis around 5 kHz)
Extended lows	Omni condenser or dynamic with good low-frequency response
Extended highs (detailed sound)	Condenser
Reduced "edge" or detail	Dynamic or ribbon
Boosted bass up close	Directional mic
Flat bass response up close	Omni mic, or directional mic with sound holes in the handle
Reduced pickup of leakage, feedback, and room acoustics	Directional mic, or omni mic up close
Enhanced pickup of room acoustics	Omni mics
Miking close to a surface, even coverage of moving sources or large sources, inconspicuous mic	Boundary mic
Coincident or near-coincident stereo	Stereo mic
Extra ruggedness	Dynamic mic
Reduced handling noise	Omni mic, or unidirectional with shock mount
Reduced breath popping	Omni mic, or unidirectional with pop filter
Distortion-free pickup of very loud sounds	Condenser with high maximum SPL spec, or dynamic
Low self-noise, high sensitivity, noise-free pickup of quiet sounds	Large-diaphragm condenser mic

Suppose you want to record a grand piano playing with several other instruments. You need the microphone to reduce leakage. Table 6.1 recommends a unidirectional mic or an omni mic up close. For this particular piano, you also want a natural sound, for which the table suggests a mic with a flat response. You want a detailed sound, so a condenser mic is the choice. A microphone with all these characteristics is a flatresponse, unidirectional condenser mic. If you're miking close to a surface (the piano lid), a boundary mic is recommended.

Now suppose you're recording an acoustic guitar on stage, and the guitarist roams around. This is a moving sound source, for which the table recommends a mini mic attached to the guitar. Feedback and leakage might not be a problem because you're miking close, so you can use an omni mic. Thus, an omni condenser mic is a good choice for this application.

For a home studio, a suggested first choice is a cardioid condenser mic with a flat frequency response. This type of mic is especially good for studio vocals, cymbals, percussion, and acoustic instruments. Remember that the mic needs a power supply to operate, such as a battery or phantom power supply. Your second choice of microphone for a home studio is a cardioid dynamic microphone with a presence peak in the frequency response. This type is good for drums and guitar amps. I recommend cardioid over omni for a home studio. The cardioid pattern rejects the leakage, background noise, and room reverb often found in home studios. An omni mic, however, can do that, too, if you mike close enough. Also, omni mics tend to provide a more natural sound at lower cost, and they have no proximity effect.

MIC ACCESSORIES

There are many devices used with microphones to route their signals or to make them more useful. These include pop filters, stands and booms, shock mounts, cables and connectors, stage boxes and snakes, and splitters.

POP FILTER

A much needed accessory for a vocalist's microphone is a pop filter or windscreen. It usually is a foam "sock" that you put over the mic. Some microphones have pop filters or ball-shaped grilles built in.

Why is it needed? When a vocalist sings a word starting with "p," "b," or "t" sounds, a turbulent puff of air is forced from the mouth. A microphone placed close to the mouth is hit by this air puff, resulting in a thump or little explosion called a pop. The windscreen reduces this problem.

The best type of pop filter is a nylon screen in a hoop, or a perforated-metal disk, placed a few inches from the mic.

You can also reduce pop by placing the mic above or to the side of the mouth, or by using an omni mic.

Audio clip 16 at www.taylorandfrancis.com/cw/bartlett-9780240821535 demonstrates how a pop filter or mic placement can prevent breath pops.

STANDS AND BOOMS

Stands and booms hold the microphones and let you position them as desired. A mic stand has a heavy metal base that supports a vertical pipe. At the top of the pipe is a rotating clutch that lets you adjust the height of a smaller telescoping pipe inside the large one. The top of the small pipe has a standard 5/8-inch 27 thread, which screws into a mic stand adapter.

A boom is a long horizontal pipe that attaches to the vertical pipe. The angle and length of the boom are adjustable. The end of the boom is threaded to accept a mic stand adapter, and the opposite end is weighted to balance the weight of the microphone.

SHOCK MOUNT

A shock mount holds a mic in a resilient suspension to isolate the mic from mechanical vibrations, such as floor thumps and mic-stand bumps. Many mics have an internal shock mount which isolates the mic capsule from its housing; this reduces handling noise as well as stand thumps.

FOR MORE INFORMATION ON THIS TOPIC, PLEASE CHECK OUT THE FOLLOWING CREATIVELIVE VIDEO COURSE:

Check out Recording Rock Vocals HERE.

Check out Recording Rock Guitars HERE.

FOR MORE INFORMATION ON THIS TOPIC, PLEASE CHECK OUT THE FOLLOWING CREATIVELIVE VIDEO COURSE:

Check out Advanced Drum Production HERE.

The following is excerpted from *Practical Recording Techniques* by Bruce Bartlett and Jenny Bartlett.

You can purchase a copy HERE.

You want to set up a recording system that's affordable, easy to use, and sonically excellent. With today's wide array of user-friendly sound tools, you can do just that. This chapter is a guide to equipment for a recording studio: what it does, what it costs, and how to set it up.

What is the bare-bones equipment you need to crank out a decent demo CD? How much does it cost? Thanks to the new breed of affordable equipment, you can put together a complete home recording studio affordably. That includes powered speakers, two mics and mic stands, recording software, an audio interface, headphones, and cables.

EQUIPMENT

Let's examine each piece of equipment in a recording studio. You'll need some sort of recording device, headphones, cables, mics, direct box, monitor speakers, audio interface, and effects.

RECORDING DEVICE

Six types of recording devices to choose from are a 2-track recorder, an iOS recording system, a recorder-mixer, a separate multitrack recorder and mixer, a computer, and a keyboard workstation. We'll look at each one.

PORTABLE 2-TRACK RECORDER

This device records in stereo on two tracks, either with two external mics or two built-in mics. Prices are $200 to $1800. Three types of 2-track recorder are:

- Handheld flash-memory recorder **(Figure 4.1)**
- Apple iOS device with recording software
- Laptop computer with recording software

FIGURE 4.1
Roland R-09HR, an example of a portable 2-track recorder.

A 2-track recorder works well for on-location stereo recording of classical music: an orchestra, symphonic band, string quartet, pipe organ, or soloist.

You can also use a 2-track recorder to capture practice sessions or gigs of pop, folk, or jazz groups. Recordings made this way sound sort of distant, as if you are listening from out in the audience. They might have background noise and not the best balance. But the recording process is very simple: just place the recorder, set a level, and hit Record.

Some handheld recorders are the Sony PCM-D50 and PCM-M10; Alesis PalmTrack; Zoom H4n; H2N, and H1; Tascam DR-07mkII, DR-05, DR-100, and DR-08; Yamaha PocketTrak; Korg MR-2 and SOS; Roland R05 and R26; Marantz PMD620 and PMD661; Edirol (Roland) R -09HR (Figure 4.1); M-Audio MicroTrack II 24/96; and Olympus LS-7 and LS-100.

iOS RECORDING SYSTEM

Apple's iOS devices including the iPad have loads of recording app options. On the iPad, simply tap the App Store Icon and search for recording apps. Check out GarageBand, MultiTrack DAW by Harmonic Dog, Sonoma Wire Works' StudioTrack **(Figure 4.2)**, StudioMini XL, Wizdom Music Morphwiz, and n-Track Studio (http://ntrack.com). On the iPhone and iPod touch, be sure to check out Audiofile Engineering's FiRe.

Also available are audio and audio/MIDI interfaces that plug into iOS devices. An example is the Tascam iU2. The Alesis iO Dock acts as an interface for an iPad, providing multiple I/O options.

Some stereo microphones are designed to plug into an iPad or other iOS devices. Three examples are the IK Multimedia iRig Mic, Blue Mikey Digital, and Tascam iM2.

FIGURE 4.2
Sonoma Wire Works StudioTrack app for the Apple iPad.

MINI STUDIO

Also called a portable studio, notebook studio, personal studio, or pocket studio, this piece of gear combines a 4-track or 8-track recorder with a mixer—all in one portable chassis **(Figure 4.3)**. A similar option is an Apple iPad with recording software, mentioned earlier.

A mini studio records in MP3 or WAV format to a flash memory card. It lets you record several instruments and vocals, then mix them to stereo. That is, you "bounce" the tracks to a stereo file in the memory card. You can copy the stereo mix file to your computer via USB (Universal Serial Bus, an industry standard that defines certain cables and communications protocols for high-speed data transfer). Pocket studios sound good enough to make demos, or to use as a musical scratchpad for ideas, but not to release commercial albums. Some examples are the Boss BR-800 and Micro BR, TASCAM DP-004, and Zoom R16. Costing about $200 to $599, a mini studio can be a good choice for beginning recordists.

FIGURE 4.3

Fostex R16, an example of a mini studio.

Some features include:

- An internal MIDI sound module (synthesizer) that plays back MIDI sequences and includes MIDI files or rhythm patterns to jam with.
- Built-in effects.
- Built-in mic (in some models).
- Autopunch (automatically "punch" into and out of record mode to correct mistakes).
- Battery or AC-adapter powered.
- Virtual tracks let you record multiple takes of a single performance, then select your favorite take during mixdown.

- Guitar-amp modeling simulates various guitar amps; mic modeling simulates mic models.
- 2 mic inputs; records up to 2 tracks at a time.
- Plays back 4 or 8 tracks at once.

DIGITAL MULTITRACKER (RECORDER-MIXER)

Other names for this device are standalone digital audio workstation (DAW), portable digital studio, or personal digital studio.

Like the mini studio, the digital multitracker combines a multitrack recorder with a mixer in a single chassis **(Figure 4.4)**. It's convenient and portable. Offering better sound than a mini studio, the digital multitracker provides CD sound quality because it can record wave files. It records 8 to 32 tracks onto a built-in hard drive or flash memory card. The mixer includes faders (volume controls) for mixing, EQ or tone controls, and aux sends for effects (such as reverb). An LCD screen displays recording levels, waveforms for editing, and other functions.

The price for this device ranges from $700 to $3600. Some manufacturers of mini studios or digital multitrackers are TASCAM, Boss, Fostex, Korg, Roland, Yamaha, and Zoom (Samson).

FIGURE 4.4
Korg D3200, an example of a digital multitracker (recorder-mixer).

SEPARATE MULTITRACK RECORDER AND MIXER

Ideal for on-location recording, a multitrack hard-disk recorder (HD recorder) records up to 48 tracks reliably on a built-in hard drive (Figure 4.5), just like the hard drive in your computer. The recorder does not have mic preamps, so it must be used with a mixer or mic preamps. Multiple recorders can be linked to get more tracks. Some examples are the Alesis ADAT HD24XR, TASCAM X-48MKII, iZ Technology RADAR V, and Fostex D2424L Mk2. Prices range from $1,500 to $20,000.

FIGURE 4.5
Alesis HD24XR, an example of a multitrack hard-disk recorder.

Current units record with 24-bit resolution and up to 96 kHz sampling rate. Some record 24 tracks of 24-bit programs at sample rates of 44.1 and 48 kHz. They record 12 tracks of 24-bit recording at 88.2 and 96 kHz. Some HD recorders allow track editing, either on a built-in LCD screen or on a plug-in computer monitor. A few models have built-in removable hard drives for backing up projects, while others include network or high-speed data connections for data transfer.

FIGURE 4.6
Mackie Onyx 1620i, an example of a mixer. It has a FireWire port for connecting to a computer.

A mixer (Figure 4.6) is an electronic device with several mic preamplifiers, volume and tone controls for each mic, monitor volume controls, and outputs for effects devices. To use it, plug mics and electric instruments into the mixer, which amplifies their signals. Connect the output signal of each mic preamp to a multitrack recorder. While recording, you use the mixer to send those signals to the desired recorder tracks, and to set recording levels. During mixdown, the mixer combines (mixes) the tracks to stereo. It also lets you adjust each track's volume, tone quality, effects, and stereo position. A large, complex mixer is called a mixing console (board or desk) and costs upwards of $3000.

The normal price range for mixers is about $180 to $1500. Mixers are made by such companies as TASCAM, Alesis, Yamaha, Mackie, Allen & Heath, Soundcraft, M-Audio, Tapco, Peavey, Samson, and many others.

COMPUTER DIGITAL AUDIO WORKSTATION (DAW)

This low-cost recording setup has three parts: a personal computer, an audio interface, and recording software **(Figure 4.7)**. Some software records MIDI data (explained later) as well as audio. The audio interface converts audio from your mics, mic preamps, or mixer into a signal that is recorded on the computer's hard drive as a magnetic pattern. Dozens of tracks can be recorded with professional quality. You mix the tracks with a mouse by adjusting the "virtual" controls that appear on your computer screen. Then you record the mix on your computer's hard drive.

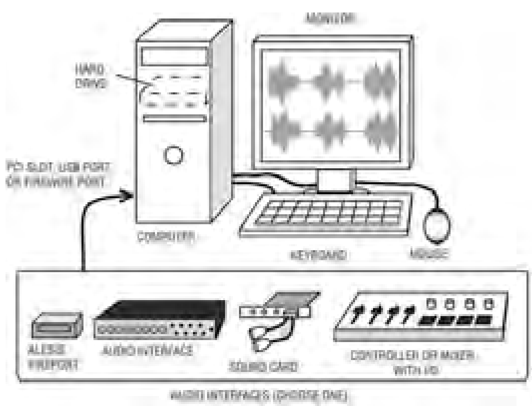


FIGURE 4.7
Computer with recording software and a choice of audio interface.

A computer studio costs about the same as a mini studio and is more powerful. It's a bargain. But since software requires computer skills, it's a little harder to learn and use than a hardware multitracker. Software recordings are at least CD quality—better than the MP3 recordings you get with a mini studio.

Recording software costs anywhere from $0 (freeware) to $1800, with $150-$500 being typical prices. Software examples include Cakewalk Home Studio, Sonar Studio, and X1 Producer; Avid Pro Tools 10 and HD Native; Apple Logic Studio and Logic Pro; Reaper; Steinberg Cubase and Nuendo; PreSonus Studio One and Studio One Pro; FL Studio Producer; Propellerhead Record; Sony Vegas and Sound Forge; BIAS Deck; Pro Tracks Plus; Magix Samplitude and Sequoia; SADiE; Pyramix; Mackie Tracktion 3; N Track Studio; RML Labs SAW Studio; Adobe Audition; and MOTU Digital Performer.

Free multitrack recording programs are available for download. Although they lack extensive features, they offer a chance to practice your skills at no cost. Examples are Audacity (from http://audacity.sourceforge.net) and GarageBand (supplied free with any new Apple computer or $79 with an iLife software upgrade).

Related to recording software is music creation software, which comes with samples of musical instruments. You set up loops (repeating drum riffs and musical patterns), play along with them using a MIDI keyboard, and record that MIDI sequence. Examples include Ableton Live, Propellerhead Reason, Sony ACID Pro, and Spectrasonics Stylus RMX.

A sound card (PCIe audio interface) costs $100 to $400. Some makers of sound cards include Echo, Event, Frontier Design, M-Audio, RME, Lynx, and E-MU.

An audio interface is a chassis with 2 to 16 mic preamps, A/D convertors, and a USB, FireWire, or Thunderbolt port that sends digital audio on a single cable to your computer. Some units have MIDI ins and out, and a high-impedance instrument input for an electric guitar, electric bass, acoustic guitar pickup, or synth.

Audio interface prices: 2-channel costs $150 to $500, 4-channel costs about $150 to $450 **(Figure 4.8)**, and 8-channel starts at $400. If you link (connect together) two 8-channel interfaces, you can record up to 16 mic signals at once.

Manufacturers of audio interfaces include Akai, Avid, Metric Halo, MOTU, PreSonus, Apogee, Focusrite, M-Audio, TC Electronic, ART, Prism, Mackie, E-MU, Alesis, Edirol, TASCAM, and many others.

FIGURE 4.8
M-Audio Fast Track Ultra, an example of a 4-channel USB 2.0 audio interface. It offers low-latency direct monitoring with reverb, and can create two headphone mixes.

An audio interface usually has built-in mic preamps. Advanced users might prefer to use separate mic preamps connected to an A/D converter (analog-to-digital converter). Some converter manufacturers are Lynx, Mitek, Crane Song, Millenia Media, RME, Universal Audio, and Apogee.

Another type of audio interface is a mixer that connects to a computer via a FireWire or USB connection. It sends the output signal of each mic preamp to a separate track in your recording software, and returns the stereo mix back to the mixer for monitoring. This can be a good system for recording a band because it offers easy

FIGURE 4.9
Fontier Design AlphaTrack, an example of a controller for recording software.

setup of monitor mixes with effects. Some examples are the Mackie Onyx 1640i (Figure 4.6) with 16 recording channels, PreSonus StudioLive (16 recording channels and 16 mic inputs), Alesis Multimix 16 USB 2.0 (18 recording channels and 8 mic inputs), and Behringer Xenyx X2442USB (2 recording channels). Generally these mixers are not controllers for recording software—you still need to use a mouse.

A USB microphone has a USB interface built in, so you just plug the mic directly into your computer's USB port. No audio interface is needed. Examples are the Audio Technica AT2020 USB and AT2005 USB; Samson C01U; Blue Snowball, Yeti Pro, and Spark Digital; Rode Podcaster USB microphone; Apogee ONE and MicC; and MXL USB.009, 990 USB, and 990 USB Stereo mics.

An alternative to a USB microphone is a standard microphone plugged into an XLR to-USB mic adapter. The MXL MicMate, Shure X2U, and the Blue Icicle connect to any mic and convert its output to USB. Phantom power and headphone monitoring are included. Its digital-audio format is 16-bit, 44.1 or 48 kHz. The Centrance MicPort Pro is similar but records up to 24-bit/96 kHz audio.

Using a mouse can be fatiguing and can lead to repetitive stress syndrome. An alternative to the mouse is a control surface or controller. It looks like a mixer with faders, but it adjusts the virtual controls you see on the computer screen. That way you can use your computer for recording, and control the software with knobs and faders instead of a mouse. Two controllers with one fader are the Frontier Design AlphaTrack ($200, Figure 4.9) and PreSonus FaderPort ($150). Multifader controllers are made by such companies as Mackie, TASCAM, M-Audio, Avid, and Alesis ($350 to $1300).

You might want to get a wireless QWERTY keyboard so you can control your DAW from a mic location in the studio. Or add a USB extender cable to the keyboard.

KEYBOARD WORKSTATION

Here's another device to record music. This is a synthesizer/sampler with a pianostyle keyboard, a built-in multitrack sequencer (MIDI recorder) and effects. Examples include the Korg Kronos and M50; Kurzweil PC3K8 and PC3LE; Yamaha Motif XF8; and Roland Fantom-G8 and Juno-Gi. The workstation lets you create a tune with several instrument sounds. Then you record the audio from the keyboard into a computer. If you want to add a vocal, use MIDI/audio recording software. An arranger keyboard workstation automatically creates backing tracks (drums, bass, and chords) by following your left hand's notes and right hand's melodies. Examples include the Yamaha Tyros4 and PSR-S910.

A similar device is a beat box or groove box, which is a sample player with pads that you tap to generate sounds and grooves. It lets you assemble a stereo music track from drum and synth samples. Copy the music from the beat box into your computer and add vocals. Examples include the Dave Smith Tempest; Korg Monotribe and Electribe; Akai MPC series and XR20; Native Instruments Maschine 1.6; Arturia Spark; Boss DR-880; Roland TD-20X; and Zoom RT-223. Beat boxes are available as software, too, such as MOTU BPM and Korg iElectribe for the Apple iPad.

LEGACY RECORDING DEVICES

Some recording equipment has become almost obsolete, but you may need to work with it if a client brings older gear into your studio to transfer audio tracks. Some of these formats are listed below.

ANALOG TAPE RECORDER: This records 2 to 24 tracks of analog audio on reels of magnetic tape from 1/4 inch to 2 inches wide. Compared to digital recorders, analog recorders have more noise, distortion, frequency-response errors, and unstable pitch. Their electronics and tape heads need frequent alignment. Still, many people love the sound of them. Some have mic inputs; all have analog line inputs and line outputs.

DAT (DIGITAL AUDIO TAPE) RECORDER: This records 2 tracks of audio digitally on a small DAT tape, a cassette about half the size of a standard analog cassette. The DAT also writes absolute time (hours, minutes, and seconds) on tape as it records. Inputs are analog mic/line level and digital; outputs are analog line level and digital.

MODULAR DIGITAL MULTITRACK (MDM): This records 8 digital tracks on a videocassette, using a rotating drum like a DAT recorder. Two popular models are the Alesis ADAT, which records on S-VHS tape, and the TASCAM DA-88, DA-78, or DA-38, which record on Hi-8 mm tape. ADAT records up to 40 minutes on a single tape; DA-88 records up to 1 hour 48 minutes. With both types, you can sync several 8-track units with a cable to add more tracks. Both have analog line inputs and outputs. ADAT has Lightpipe-format digital outputs; TASCAM models have TDIF-format digital outputs.

MICROPHONE

So far we covered recording devices. Let's move on to other studio equipment. A microphone converts the sound of a voice or instrument into an electrical signal that can be recorded. Microphone sound quality varies widely, so be sure to get one or more good mics costing at least $100 each. Your ears should tell you if a mic's fidelity is adequate for your purpose. Some people are happy to get any sound recorded; others settle for nothing less than professional sound quality.

Probably the most useful mic types for home recording are the cardioid condenser mic and cardioid dynamic mic. The cardioid pickup pattern helps reject room acoustics for a tighter sound. The condenser type is commonly used on cymbals, acoustic instruments , and studio vocals; dynamics are used typically on drums and guitar amps. You'll also need at least one mic cable, mic stand, and boom costing about $25 each.

Do you want to record solo instruments or classical musical ensembles in stereo with two mics at a distance? You'll need either a stereo mic or a matched pair of high-quality condenser or ribbon mics of the same model number, plus a stereo mic-stand adapter.

PHANTOM POWER SUPPLY

A phantom power supply powers the circuits in condenser mics. It uses the same cable that carries the mic's audio signal. You don't need the supply if your condenser mic has a battery, or if your mixer or audio interface supplies phantom power at its mic connectors (most do). Some makers of phantom supplies are Behringer, ART, AudioTechnica, Rolls, and Whirlwind. The price ranges from $20 to $139.

MIC PREAMP

This device amplifies a mic signal up to a higher voltage called "line level," which is required by mixers and recorders. A standalone mic preamp provides a slightly cleaner, smoother, or more colorful sound than a mic preamp built into a mixer or audio interface, but costs much more. A 2-channel preamp costs $120 to $2000, while 8-channel preamps range from $600 to $6000. Studios on a budget can do without a standalone preamp.

Some manufacturers are Manley, True Systems, Focusrite, Universal Audio, GLM, Chandler, A Designs, Millennia Media, Avalon, Great River, John Hardy, Benchmark, AEA, Apogee, Vintech, Grace, Groove Tubes, PreSonus, Mackie, Summit Audio, Studio Projects, Samson, dbx, Aphex, and M-Audio.

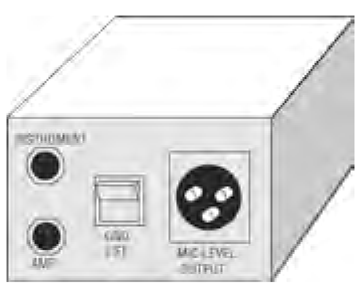


FIGURE 4.10
A direct box.

DIRECT BOX (DI)

A direct box **(Figure 4.10)** is used to connect an electric instrument (guitar, bass, synth) to an XLR-type mic input of a mixer, recorder-mixer, or audio interface. It lets you record electric instruments directly into your mixer without a microphone. A direct box picks up a very clean sound, which may be undesirable for electric guitar. If you want to pick up the distortion of the guitar amp, use a microphone instead. Or use a guitar-amp modeling device or modeling plug-in.

Some recorder-mixers and audio interfaces have "high-impedance" or "instrument" inputs meant for electric guitars and synthesizers.

EFFECTS

A recording without effects sounds rather dead and plain. Effects such as reverberation, echo, and chorus can add sonic excitement to a recording. They are produced by devices called signal processors (see Figure 4.11) or by plug-ins, which are software effects used in a computer recording program.

Although effects are built into most recording programs and recorder-mixers, most analog mixers require external effects units. On the mixer is a set of connectors (labeled "send" and "return") for hooking up an external effects unit, such as a reverb or delay device. A unit with one effects send lets you add one type of effect; a unit with two effects sends lets you add two different effects to create more sonic interest.

FIGURE 4.11
An effects unit.

MISCELLANEOUS EQUIPMENT

Other equipment for your home studio includes audio cables, USB or FireWire cables, power outlet strips, lighting, tables or studio furniture, mic pop filters, masking tape and a pen to label inputs and cables, contact cleaning fluid, MIDI equipment stands, music stands, session forms, connector adapters, pen and paper, a flashlight, and so on.

MIDI STUDIO EQUIPMENT

We've looked at several types of recording setups. All can help you create quality demos. As you go higher in price you get more features and better sound. For example, if you want to record an entire band playing all at once, with each instrument having its own mic, you'll need a system with more microphones, more tracks, and more headphones.

As we've seen, putting together a home studio or project studio needn't cost much. As technology develops, better equipment is available at lower prices. That dream of owning your own studio is within reach.

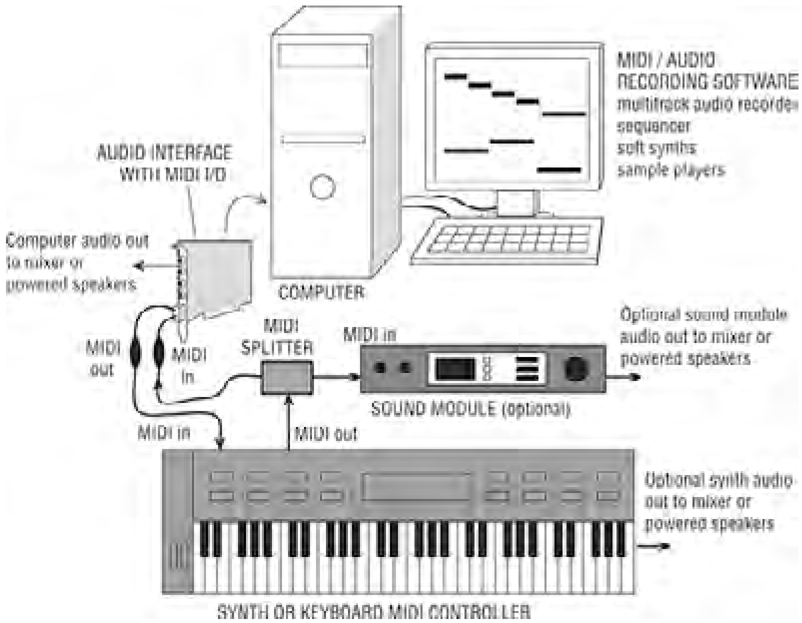


FIGURE 4.12
One type of MIDI studio.

MONITORING

One of the most exciting moments in recording comes when the finished mix is played over the studio monitor speakers. The sound is so clear you can hear every detail, and so powerful you can feel the deep bass throbbing in your chest.

You use the monitor system to listen to the output signals of the console, audio interface, or recorders. It consists of the console monitor mixer, the power amplifiers, loudspeakers, and the listening room. The power amplifier boosts the electrical power of the console signal to a sufficient level to drive a loudspeaker. The speaker converts the electrical signal into sound, and the listening-room acoustics affect the sound from the speaker.

A quality monitor system is a must if you want your mixes to sound good. The power amp and speakers tell you what you're doing to the recorded sound. According to what you hear, you adjust the mix and judge your mic techniques. Clearly, the monitor system affects the settings of many controls on your mixer, as well as your mic selection and placement. And all those settings affect the sound you're recording. So, using inadequate monitors can result in a poor-sounding product coming out of your studio.

EQUIPMENT CONSIDERATIONS

It's important to use accurate speakers that have a flat frequency response. If your monitors are weak in the bass, you will tend to boost the bass in the mix until it sounds right over those monitors. But when that mix is played over speakers with a flatter response, it will sound too bassy because you boosted the bass on your mixer. So, using monitors with weak bass results in bassy recordings; using monitors with exaggerated treble results in dull recordings, and so on. In general, colorations in the monitors will be inverted in your mixdown recording.

That's why it's so important to use an accurate monitor system—one with a wide, smooth frequency response. Such a system lets you hear exactly what you recorded.

SPEAKER REQUIREMENTS

The requirements for an accurate studio monitor are these:

- Wide, smooth frequency response. To ensure accurate tonal reproduction, the on-axis response of the direct sound should be ±4 dB or less from at least 40 Hz to 15 kHz. The low-frequency response of a small monitor speaker should extend to at least 70Hz.
- Uniform off-axis response. The high-frequency output of a speaker tends to diminish off-axis. Ideally the response at 30 degrees off-axis should be only a few decibels down from the response on-axis. That way, a producer and engineer sitting side-by-side will hear the same amount of treble. Also, the treble will not change as the engineer moves around at the console.
- Good transient response. This is the ability of the speaker to accurately follow the attack and decay of musical sounds. If a speaker has good transient response, the bass guitar sounds tight, not boomy, and drum hits have sharp impact. Some speakers are designed so that the woofer and tweeter signals are aligned in time. This aids transient response.
- Clarity and detail. You should be able to hear small differences in the sonic character of instruments, and to sort them out in a complex musical passage.
- Low distortion. Low distortion is necessary because it lets you listen to the speaker for a long time without your ears hurting. A good spec might be total harmonic distortion under 3% from 40 Hz to 20 kHz at 90 dBSPL (sound pressure level).
- Sensitivity. Sensitivity is the sound pressure level a speaker produces at 1 meter (m) when driven with 1 watt (W) of pink noise. Pink noise is random noise with equal energy per octave. This noise is either band-limited to the range of the

speaker or is a one-third-octave band centered at 1 kHz. Sensitivity is measured in dB/W/m (dB sound pressure level per 1 W at 1 m). A spec of 93 dB/W/m is considered high; 85 dB/W/m is low. The higher the sensitivity, the less amplifier power you need to get adequate loudness.

- High output capability. This is the ability of a speaker to play loudly without burning out. You often need to monitor at high levels to hear quiet details in the music. Plus, when you record musicians who play loudly in the studio, it can be a letdown for them to hear a quiet playback. So you may need a maximum output of 110 dBSPL.

This formula calculates the maximum output of a speaker (how loud it can play):

$$dBSPL=10\log(P)+S$$

where dBSPL is the sound pressure level at 1 m, P is the continuous power rating of the speaker in watts, and S is the sensitivity rating in dB/W/m.

For example, if a speaker is rated at 100 W maximum continuous power, and its sensitivity is 94 dBSPL/W/m, its maximum output SPL is 10 log(100) + 94 = 114 dBSPL (at 1 m from the speaker). The level at 2 m will be about 4 to 6 dB less.

NEARFIELD™ MONITORS

Many professional recording studios use large monitor speakers that have deep bass. However, they are expensive, heavy, and difficult to install, and they are affected by the acoustics of the control room. All studios, large or small, need a pair of Nearfield monitor speakers **(Figure 5.1)**. A Nearfield monitor is a small, wide-range speaker typically using a cone woofer and dome-shaped tweeter. You place a pair of them about 3 or 4 feet apart, on stands just behind the console, about 3 or 4 feet from you. Nearfields are far more popular than large wall-mounted speakers.

FIGURE 5.1
A Nearfield monitor speaker.

This technique, developed by audio consultant Ed Long, is called Nearfield monitoring. Because the speakers are close to your ears, you hear mainly the direct sound of the speakers and tend to ignore the room acoustics. Plus, Nearfield monitors sound very clear and provide sharp stereo imaging. Some units have bass or treble tone controls built in to compensate for the effects of speaker placement and room surfaces.

Nearfield monitors have enough bass to sound full when placed far from walls. Although most Nearfields lack deep bass, they can be supplemented with a subwoofer to reproduce the complete audio spectrum. Or you can check the mix occasionally with headphones that have deep bass.

Some Nearfields are in a satellite-subwoofer format. The two satellite speakers are small units, typically including a 4-inch woofer and 3/4-inch dome tweeter. The satellites are too small to produce deep bass, but that is handled by the subwoofer—a single cabinet with one or two large woofer cones. Typically, the subwoofer (sub) produces frequencies from 100 Hz down to 40 Hz or below. Because we do not localize sounds below about 100 Hz, all the sound seems to come from the satellite speakers. The sub-satellite system is more complicated to set up than two larger speakers, but offers deeper bass.

POWERED (ACTIVE) MONITORS

Most monitors have a power amplifier built in. You feed them a line-level signal (labeled MONITOR OUT or STEREO L/R) from your mixing console or audio interface. Most powered monitors are bi-amplified: they have one amplifier for the woofer and another for the tweeter. The advantages of bi-amplification include:

- Distortion frequencies caused by clipping the woofer power amplifier will not reach the tweeter, so there is less likelihood of tweeter burnout if the amplifier clips. In addition, clipping distortion in the woofer amplifier is made less audible.
- Intermodulation distortion is reduced.
- Peak power output is greater than that of a single amplifier of equivalent power.
- Direct coupling of amplifiers to speakers improves transient response—especially at low frequencies.
- Bi-amping reduces the inductive and capacitive loading of the power amplifier.
- The full power of the tweeter amp is available regardless of the power required by the woofer amp.

THE POWER AMPLIFIER

If your monitor speakers are not powered, you need a power amplifier **(Figure 5.2)**. It boosts your mixer's line-level signal to a higher power in order to drive the speakers.

How many watts of power do you need? The monitor speaker's data sheet gives this information. Look for the specification called "Recommended amplifier power." A power amp of 50 W per channel continuous is about the minimum for Nearfield monitors; 150 W is better. Too much power is better than too little, because an underpowered system is likely to clip or distort. This creates high frequencies that can damage tweeters.

FIGURE 5.2
Crown D-75A, an example of a power amplifier.

A good monitor power amp has distortion under 0.05% at full power. It should have a high damping factor—at least 100—to keep the bass tight. The amp should be reliable. Look for separate level controls for left and right channels. The amplifier should have a clip or peak light that flashes when the amp is distorting.

SPEAKER CABLES AND POLARITY

When you connect the power amp to the speakers, use good wiring practice. Long or thin cables waste amplifier power by heating. So put the power amp(s) close to the speakers and use short cables with thick conductors—at least 16 gauge. The low resistance of these cables helps the power amplifier to damp the speaker motion and tighten the bass.

If you wire the two speakers in opposite polarity, one speaker's cone moves out while the other speaker's cone moves in. This causes vague stereo imaging, weak bass, and a strange sense of pressure on your ears. Be sure to wire the speakers in the same polarity as follows: in both channels, connect the amplifier positive (+ or red) terminal to the speaker positive (+ or red) terminal. Setting the correct polarity is also called "speaker phasing."

SPEAKER PLACEMENT

Once you have acquired the speakers and worked on the room acoustics, you can install the speakers.

- Mount them at ear height so the mixer doesn't block their sound.
- To prevent sound reflections off the mixing console, place the speakers on stands behind the console's meter bridge, rather than putting them on top.
- For best stereo imaging, align the speaker drivers vertically and mount the speakers symmetrically with respect to the side walls.

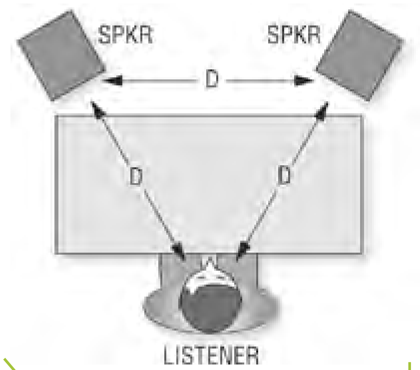


FIGURE 5.3
The recommended speaker/listener relationship for best stereo imaging.

- Place the two speakers as far apart as you're sitting from them; aim them toward you, and sit exactly between them **(Figure 5.3)**. Some engineers recommend aiming the speakers 12–18 inches behind the listener.
- To get the smoothest low-frequency response, put the speakers near the shorter wall, and sit about 1/3 of the way back from the front wall.
- Use a foam isolation device under each speaker to ensure deep, tight bass. Auralex 1 MoPads and Primacoustics Recoil Stabilizers work well.

Try to position the monitors several feet from the nearest wall. Wall reflections can degrade the frequency response and stereo imaging. The closer to the wall the monitors are, the more bass you hear. In small rooms you might have to place the monitors against the wall, which will exaggerate the bass. But some monitors have a low-frequency attenuation switch to compensate.

KRK Systems ERGO is a monitor controller that corrects a room's phase and frequency response errors, resulting in accurate monitoring—so mixes translate better to other systems.

USING THE MONITORS

You've treated the room acoustics, and you've connected and placed the speakers as described earlier. Now it's time to adjust the stereo balance.

1. Play a mono musical signal into an input channel on your mixer, and assign it to the stereo output channels 1 and 2.
2. Adjust the input channel's pan pot so that the signal reads the same on the stereo output channel 1 and 2 meters.
3. Place the two speakers the same distance from you.
4. Sit at the mixer exactly midway between the speakers. If you sit off-center, you will hear the image shifted toward one side. Listen to the image of the sound between the speaker pair. You should localize it midway between the monitors; that is, straight ahead.
5. If necessary, center the image by adjusting the left or right volume control on your power amp or powered monitors.

When you do a mixdown, try to keep the listening level around 85 dBSPL—a fairly loud home listening level. As discovered by Fletcher and Munson, we hear less bass in a program that is played quietly than in the same program played loudly. If you mix a program while monitoring at, say, 100 dBSPL, the same program will sound weak in the bass when heard at a lower listening level—which is likely in the home. So, programs meant to be heard at 85 dBSPL should be mixed and monitored at that level.

Loud monitoring also exaggerates the frequencies around 4 kHz. A recording mixed loud may sound punchy, but the same recording heard at a low volume will sound dull and lifeless.

Here's another reason to avoid extreme monitor levels: loud sustained sound can damage your hearing or cause temporary hearing loss at certain frequencies. If you must do a loud playback for the musicians (who are used to high SPLs in the studio), protect your ears by wearing earplugs or leaving the room.

You can get a low-cost sound level meter from Radio Shack. Play a musical program at 0 VU or 0 dB on the mixer meters and adjust the monitor level to obtain an average reading of 85 dBSPL on the sound level meter. Mark the monitor-level setting.

Before doing a mix, you may want to play some familiar commercial CDs over your monitors to remind yourself what a good tonal balance sounds like. Listen to the amount of bass, midrange, and treble, and try to match those in your mixes. But listen to several CDs because they vary.

While mixing, monitor the program alternately in stereo and mono to make sure there are no out-of-phase signals that cancel certain frequencies in mono. Also beware of center-channel buildup. Instruments or vocals that are panned to center in the stereo mix sound 3 dB louder when monitored in mono than they do in stereo. That is, the balance changes in mono—the center instruments are a little too loud. To prevent this, don't pan tracks hard left and hard right. Bring in the side images a little so they will be louder in mono.

You'll mix the tracks to sound good on your accurate monitors. But also check the mix on small inexpensive speakers to see whether anything is missing or whether the mix changes drastically. Make sure that bass instruments are recorded with enough edge or harmonics to be audible on the smaller speakers. It's a good idea to make a CD copy of the mix for listening in a car, boom box, or compact stereo.

HEADPHONES

Compared to speakers, headphones have several advantages:

- They cost much less.
- There is no coloration from room acoustics.
- The tone quality is the same in different environments.
- They are convenient for on-location monitoring.
- It's easy to hear small changes in the mix.
- Transients are sharper due to the absence of room reflections.

Headphones have several disadvantages:

- They become uncomfortable after long listening sessions.
- Cheap headphones have inaccurate tone quality.
- Headphones don't project bass notes through your body.
- The bass response varies due to changing headphone pressure.
- The sound is in your head rather than out front.
- You hear no room reverberation, so you may add too much or too little artificial reverb.
- It's difficult to judge the stereo spread. Over headphones, panned signals tend to sound closer to center than the same signals heard over speakers. The same is true of stereo recordings made with a coincident pair of mics.

Because speakers sound different from headphones, it's best to do mixes over speakers. But check your mixes on headphones too because so many listeners use them with MP3 players.

Focusrite's Saffire PRO 24 DSP Audio Interface with VRM technology simulates the sound of studio monitor loudspeakers over headphones. You can choose three room models and 15 different speaker emulations.

If your monitor speakers are in the same room as your microphones, the mics pick up the sound of the speakers. This causes feedback or a muddy sound. In this case you must monitor only with headphones while recording or overdubbing, then monitor with speakers during playback or mixdown.

If you monitor with headphones as the musicians are playing, external sounds leak through the headphones and mask the sounds you are trying to monitor. That makes it hard to judge the sound quality you are picking up. Check it during playback. Closed-cup (closed back) headphones or in-the-ear earphones provide the best isolation from outside sounds.

THE CUE SYSTEM

The cue system is a monitor system for musicians to use as they're recording. It includes some of the aux knobs in your mixer, a small power amplifier or headphone amp, a headphone connector box, and headphones. Musicians often can't hear each other well in the studio, but they can listen over headphones to hear each other in a good balance. Also, they can listen to previously recorded tracks while overdubbing.

FIGURE 5.4
ART HeadAmp 4, an example of a headphone amplifier.

Headphones for a cue system should be durable and comfortable. They should be closed-cup to avoid leakage into microphones—especially for a click track. Also, the cue "phones" should have a smooth response to reduce listening fatigue, and should play loud without burning out. Make sure they are all the same model so each musician hears the same thing. A built-in volume control is convenient.

A headphone amplifier is shown in Figure 5.4. Connect it to a headphone monitor jack on your mixer or audio interface. It drives up to eight headphones and lets you adjust the volume of each one. Examples include the PreSonus HP60 and HP4, ART HeadAmp 4 and HeadAmp 6 PRo, Samson S-Phone, and C-Que 8.

Does your mixer or audio interface have a strong signal at its headphone jack? You could install four 1/4-inch stereo jacks in a small metal box wired in parallel (tip to tip, ring to ring, and sleeve to sleeve). Connect them by a single cable to the headphone jack of your mixer or audio interface.

Although some consoles can provide several independent cue mixes, the ideal situation is to set up a personal monitor mixer near each musician. Then they can set their own cue mix and listening level. The inputs of these mixers are fed from the console output buses. Other names for this device are headphone monitor mixer or cue mixer. Some examples are the Behringer PowerPlay System, Furman HR-6, Mackie HMX-56, Aviom A16II, Hear Technologies Hear-Back-Four-Pack, and Roland M-48.

Suppose a vocalist sings into a microphone and hears that mic's signal over the cue headphones. If the singer's voice and the headphone's sound are opposite in polarity, the voice partially cancels or sounds funny in the headphones. Make sure that the voice and headphones are the same polarity.

Here's how. While talking into a mic and listening to it on headphones, reverse the ground and signal leads to the headphones connector. The position that gives the fullest, most solid sound in the headphones is correct.

All the headphones in your studio should be the same model, so that everyone will hear with correct polarity.

CHAPTER

4 PRE-PRODUCTION

PRE-PRODUCTION

PLANNING AS AN ARTIST

by Jesse Cannon

While the time you take to craft a great song is the most important part of the process of creating a song people love, the second most important part is pre-production. During pre-production you should get together with the other creative people involved in your project (producer, engineer, A&R, etc.) and begin to plan for making a representation of your song that you will be creatively happy with. This time can begin to eliminate the many variables that can go wrong in your project, and provide you with focus and a clear vision of what your record should sound like.

ELIMINATING VARIABLES

One of the key elements of pre-production is to eliminate variables so you can have a clear head and intent in the studio. Whether the studio clock is running up a bill or not, you want to be able to focus on a great performance in the studio, not be thinking about the possible changes you could have made to a song while in rehearsal or pre-production. You can assure a focus that is greatly helpful to making good decisions for your songs when you are sure of your song selection, key choice, tempo, arrangement and a variety of other choices which we will address below.

DEMOING

Pre-production can often be synonymous with demoing a song. While you may do a lot of pre-production in your rehearsal room, it's important to record your song down and hear them back in a recorded context since this is how they will ultimately be heard. Taking the time to listen back to each part back one by one, analyze them, and pick them apart is crucial to finding small refinements you can do to your song and save time in the studio while keeping the vibe up during critical performance time in the studio.

It is also helpful to a get recording of the band playing the song live and for the producer to be in the room during this process since it can help them understand how to capture the group.

PERSPECTIVE

One of the most important parts of pre-production is the new-found objectivity you can receive. Hearing your songs back on more than a iPhone voice memo recording can give you clarity and creative direction. It's helpful to make fast demos of your song that focus less on tight performances and just hearing back your compositions. The more clearly you can hear your song represented, the easier it is to make refinements to your song and make great decisions about their direction.

This objectivity is helpful in making great decisions for your songs. Producer Greg Wells (Katy Perry, Adele) has a great saying, "The toughest part of recording is we can never hear songs we produce the way a listener hears it for the first time." Making demos and hearing your song in new ways can help get you some of the objectivity he's speaking of. By hearing your song on a new set of speakers instead of a rehearsal or close-miked instead of a stereo ambient recording can give great perspective to the way you hear your song.

PREPAREDNESS

Preproduction is the best time to find out if everyone is actually ready to record. Often times this can be where a producer sees a drummer isn't up to par to play the songs the way they should and may tell the band it is best to hire a session drummer or will need to rehearse for another month or two to be ready to record. The producer may realize the singer isn't good at singing harmonies and then decide to take some time on a home studio and work with the singer or send them to vocal lessons. These realizations can make or break a record as well as keep it on track to stay on budget.

You may also want to refine a player's techniques and give them time to get used to them before recording. Many times a guitarist may not be used to doing more downstrokes on their guitar which is often needed in aggressive music. A drummer may not be used to having to move their cymbals away from the drums or hitting the drum in the center of the drum. Spotting these flaws and having time to rehearse to rid musicians of bad habits can be a key element to pre-production

If the songs need vocal harmonies written it can be helpful to flesh them out during this process instead of having to write them in the studio. The same goes for session musician's parts, percussion overdubs, and any other auxiliary musicians' parts. **The more of these things you have out of the way before you hit the studio, the better a session you will have and a higher chance of making a great recording of your songs.**

SONG CHOICES

It's common practice on most professional recording to write more songs than will actually appear on your record. This way you ensure the quality of material you present is as high as possible by having a vast selection of material to choose from. One of the most successful producers of all time, Rick Rubin, often has groups write 2.5 albums worth of material for each album they release. During the pre-production process you will begin to demo and figure out which songs make the most sense to appear on your record.

These decisions can be based on both the strongest compositions and those that fit the mood of your recording. For example, you may have written the dance jam of the century, but making that fit with your blues rock band may be a real let down to those emotionally connecting with your blues rockin' jams, so it may be best to release the dance jam separately. As well, figuring out which songs are best placed to be brought to fruition in the studio can be extremely helpful. By cutting songs during the preproduction process you free time in studios to concentrate on only the songs with the most potential and save money if you are incurring studio costs.

TONES

One of the first choices you need to make is the sonic direction of the record you are going to make. Do you want it to sound classic like a 70's record or ultra modern and tight with tons of modern production tricks? These decisions will influence what gear you should have on hand. If the guitarist wants a modern metal tone but plays a Vox AC-30 you will need to make sure you can acquire an amp that is capable of getting this tone. If your drummer wants a jazzy tone, modern, heavy bright cymbals will kill this vibe immediately. If you want it to sound vintage and authentically old, you may need to find a studio with analog gear and a tape machine.

The experienced ear of a producer or a musician with a vision should begin to hear and imagine the tones you will get. This is also a great time to think of the other instruments you may want on your recording. If you are going to need strings or a marimba, you should begin to plan on making sure your studio has suitable sounding samples or players for your recording.

If you need big room sounds on your recording for large Led Zeppelin-like drums or distant gang vocals, you should begin to think about what rooms to rent out in order to get these sounds. Many recordings acquire large studio spaces for specific instruments they want room tones on and do much of the overdubbing in a home studio. Developing this plan to keep within your budget while fulfilling your creative vision is crucial to being happy with your project.

TEMPO, KEY, ARRANGEMENT

After you have written and rehearsed your songs it's important to experiment with the key, tempo and arrangement for your song. These three attributes should be experimented with until they feel perfect. Finding the sweet spot for each of these attributes is one of the most crucial parts of presenting your song and getting a great recording for it. Figuring it out during the pre-production process allows you to have crucial decisions solved before entering the recording studio so your mind can focus on more important factors.

The tempo of the song is one of the most important factors for the feel of a song. A too-slow tempo can feel like a lack of energy, whereas a too-fast tempo can feel like a jumbled mess. A tempo is often chosen for what showcases the melody of the song best and brings out its feel and groove. During pre-production it's often helpful to rehearse to a click track if one will be used in the studio so the tempo is established and everyone is comfortable with it. If one won't be used it can he helpful to establish a tempo by listening back to a recording and tapping it out for later reference. You can then use the click track to count the band in so they start the song at the proper tempo each time.

The key is crucial to the feeling of the song as well as the range in which your singer can sing in. If your melody is in a key where your singer has trouble hitting notes, this can be a huge problem in the studio and may cause the song to need to be re-tracked in a key where your singer can properly sing the melody. The key also has a lot to do with the emotional feeling in a song and the way it is presented tonally. Especially in guitar-based music the various tunings and voicing a guitarist will need to use to play in a certain key can greatly determine the emotion behind the song.

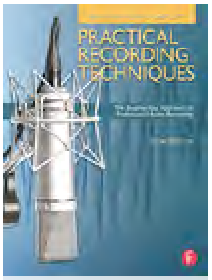
The arrangement is the order of the different parts of the song. Experimenting with adding a second chorus at the end of the song is best to hash out during this process instead of under the clock and variables of the studio environment. The arrangement can also be how long a part is played for, if a certain instrument comes in or drops out and a handful of other variables. While the studio can be great for fleshing these ideas out, experimenting with them during rehearsal and pre-production can free your mind of these distractions and give you a better focus while recording.

PLANNING

Lastly, pre-production is the ultimate time to plan. If a singer blows their voice out often, you may want to schedule recording sessions a week apart. As well if they get allergies every year in April it may be best to get to work in mid-May. If you're going to have guest musicians you want to make sure they aren't going to be on tour. You also want to make sure everyone's work schedules align with the studio of your choosing.

PRE-PRODUCTION

PLANNING AS AN ENGINEER



The following is excerpted from *Practical Recording Techniques* by Bruce Bartlett and Jenny Bartlett.

You can purchase a copy HERE.

"We're rolling. Take One." These words begin the recording session. It can be an exhilarating or an exasperating experience, depending on how smoothly you run it.

The musicians need an engineer who works quickly yet carefully. Otherwise, they may lose their creative inspiration while waiting for the engineer to get it together. And the client, paying by the hour, wastes money unless the engineer has prepared for the session in advance.

This chapter describes how to conduct a multitrack recording session. These procedures should help you keep track of things and run the session efficiently.

There are some spontaneous sessions, especially in home studios, that just "grow organically" without advance planning. The instrumentation is not known until the song is done! You just try out different musical ideas and instruments until you find a pleasing combination.

In this way, a band that has its own recording gear can afford to take the time to find out what works musically before going into a professional studio. In addition, if the band is recording itself where it practices, the microphone setup and some of the console settings can be more or less permanent. This chapter, however, describes procedures usually followed at professional studios, where time is money.

PREPRODUCTION

Long before the session starts, you're involved in preproduction—planning what you're going to do at the session, in terms of overdubbing, track assignments, instrument layout, and mic selection.

INSTRUMENTATION

The first step is to find out from the producer or the band what the instrumentation will be and how many tracks will be needed. Make a list of the instruments and vocals that will be used in each song. Include such details as the number of tom toms, whether acoustic or electric guitars will be used, and so on.

RECORDING ORDER

Next, decide which of these instruments will be recorded at the same time and which will be overdubbed one at a time. It's common to record the instruments in the following order, but there are always exceptions:

1. Loud rhythm instruments—bass, drums, electric guitar, electric keyboards
2. Quiet rhythm instruments—acoustic guitar, piano
3. Lead vocal and doubled lead vocal (if desired)
4. Backup vocals (in stereo)
5. Overdubs—solos, percussion, synthesizer, sound effects
6. Sweetening—horns, strings

The lead vocalist usually sings a guide vocal or scratch vocal along with the rhythm section so that the musicians can get a feel for the tune and keep track of where they are in the song. You record the vocalist's performance but you will probably rerecord it later. That eliminates leakage and lets you focus on the lead vocal.

In a MIDI studio, a typical order might be:

1. Drum machine or soft synth drum set (playing programmed patterns)
2. Synthesizer bass sound
3. Synthesizer chords
4. Synth melody
5. Synth solos, extra parts
6. Vocals and miked solos

TRACK ASSIGNMENTS

Now you can plan your track assignments. Decide what instruments will go on which tracks of the multitrack recorder. The producer may have a fixed plan already.

What if you have more instruments than tracks? Decide what groups of instruments to put on each track. In a 4-track recording, for example, you might put guitars on track 1, bass and drums on track 2, vocals on track 3, and keyboards on track 4.

Remember that when several instruments are assigned to the same track, you can't separate their images in the stereo stage. That is, you can't pan them to different positions—all the instruments on one track sound as if they're occupying the same point in space. For this reason, you may want to do a stereo mix of the rhythm section on tracks 1 and 2, for instance, and then overdub vocals and solos on tracks 3 and 4.

It's possible to overdub more than four parts on a 4-track recorder. To do this, bounce (mix) several tracks onto one, then record new parts over the original tracks. Your recorder manual describes this procedure.

If you have many tracks available, leave several tracks open for experimentation. For example, you can record several takes of a vocal part using a separate track for each take, so that no take is lost. Then combine the best parts of each take into a single final performance on one track. Most recorder-mixers let you do these extra takes on virtual tracks.

It's also a good idea to record the monitor mix on one or two unused tracks. The recorded monitor mix can be used as a cue mix for overdubs, or to make a recording for the client to take home and evaluate.

SESSION SHEET

Once you know what you're going to record and when, you can fi ll out a session sheet (Figure 15.1). This simple document is adequate for home studios. "OD" indicates an overdub. Note the recorder-counter time for each take, and circle the best take.

SONG: Escape to Air Island

TRACK	INSTRUMENT	MICROPHONE
1	BASS	DIRECT
2	KICK	AKG D-112
3	DRUMS	CROWN GLM-100
4	LEAD VOC OD	STUDIO PROJECTS B1
5	HARM. VOC OD	STUDIO PROJECTS B1
6	LEAD GUIT OD	SHURE SM57
7	KEYS L	DIRECT
8	KEYS R	DIRECT

TAKE		
1	03:21 - 06:18	FS
2	06:25 - 09:24	INC
3	10:01 - 13:02	

FIGURE 15.1
A session sheet for a home studio.

PRODUCTION SCHEDULE

In a professional recording studio, the planned sequence of recording basic tracks and overdubs is listed on a production schedule (Figure 15.2).

WEST WIND STUDIOS
PRODUCTION SCHEDULE

Artist: Steve Mills
Album: Long Distance Music
Producer: B. Brauning
Engineer: D. Scriven
Date: 1-17-09

Project files: c:\cakewalk projects\mills\
mr_potato_head.cwp and sambatina.cwp

Audio data folders: d:\mills\mr_potato_head\ and \sambatina\

1. Record Mr. Potato Head
Instrumentation: Electric bass, drums, electric rhythm guitar, electric lead guitar, acoustic piano, sax, lead vocal.
Comments: Record rhythm section with scratch vocal. OD sax, piano, lead vocal. Double lead guitar in stereo.

2. Record Sambatina
Instrumentation: Electric bass, drums, acoustic guitar, percussion, synth
Comments: Record rhythm section with scratch acoustic guitar. Record synth MIDI. OD acoustic guitar, percussion and synth.

3. Mix Mr. Potato Head
Comments: Add 40msec echo to toms. Increase reverb on sax solo.

4. Mix Sambatina
Comments: Add flanger to bass on intro only. Do automated flange on percussion.

FIGURE 15.2
A production schedule.

TRACK SHEET

Another document used in a pro studio is the track sheet or multitrack log (Figure 15.3). Write down which instrument or vocal goes on which track. The track sheet also has blanks for other information such as take numbers. If you are using a DAW, you can enter this information by typing it in.

WEST WIND STUDIOS
TRACK SHEET

Artist: S. Mills — Album: Long Distance Music
Producer: B. Brauning — Engineer: D. Scriven
Date: 1-17-09 — Song title: Dig Up Nebraska

Take # 1	Start 00:31	Stop 03:31
Take # 2 NC	Start 03:54	Stop 06:49
Take # 3 LFS	Start 07:21	Stop 08:38
Take # 4 FS	Start 09:01	Stop 09:12
Take # 5	Start 09:25	Stop 12:27

Track 1: elec bass (Fender Precision DI)
Track 2: kick
Track 3: snare
Track 4: rack tom
Track 5: floor tom
Track 6: overhead L
Track 7: overhead R
Track 8: rhythm gtr L (Martin HD-28)
Track 9: rhythm gtr R (Taylor 814ce)
Track 10: lead gtr (Parker PM-20PRO, Roland Cube 60)
Track 11: keys L (Prophet '08 DI)
Track 12: keys R (Prophet '08 DI)
Track 13: scratch lead vocal
Track 14: lead vocal
Track 15: background vocals 1
Track 16: background vocals 2

FIGURE 15.3
A track sheet (multitrack log).

MICROPHONE INPUT LIST

Make up a microphone input list similar to that seen in **Table 15.1**. Later you will place this list by the mic snake box and by the mixing console.

Be flexible in your microphone choices—you may need to experiment with various mics during the session to find one giving the best sound with the least console equalization. During lead-guitar overdubs, for example, you can set up a direct box, three close-up microphones, and one distant microphone—then find a combination that sounds best. Find out what sound the producer wants—a "tight" sound; a "loose, live" sound; an accurate, realistic sound, etc. Ask to hear recordings having the kind of sound the producer desires. Try to figure out what techniques were used to create those sounds, and plan your mic techniques and effects accordingly.

TABLE 15.1 ■ A MICROPHONE INPUT LIST

Input	Instrument	Microphone
1	Bass	Direct
2	Kick	EV N/D868
3	Snare	AKG C451
4	Overhead L	Shure SM81
5	Overhead R	Shure SM81
6	High Toms	Sennheiser MD421-II
7	Floor Tom	Sennheiser MD421
8	Electric Lead Guitar	Shure SM57
9	Electric Lead Guitar	Shure SM57
10	Piano Treble	Crown PZM-6D
11	Piano Bass	Crown PZM-6D
12	Scratch Vocal	Beyer M88

INSTRUMENT LAYOUT CHART

Work out an instrument layout chart, indicating where each instrument will be located in the studio, and where baffles and isolation booths will be used (if any). In planning the layout, make sure that all the musicians can see each other and are close enough together to play as an ensemble. Often a circular arrangement works well.

Keep all these documents in a single folder or notebook that is labeled with the band's name and recording date. Also include contact information, time sheets, invoices, and all the correspondence about the project.

ADDITIONAL ADVICE FROM CARLOS LELLIS FERREIRA

THE 'RECCE' / VISITING THE RECORDING VENUE

It is vital for the production team to visit the recording venue for their impending sessions in advance. This action, commonly referred to as the 'recce', can help save a lot of time and avoid embarrassment, as in some cases a venue may be deemed unsuitable for sessions on personal inspection, e.g. contrary to advertising it may not accommodate the number of musicians required comfortably, etc.

From excessive extraneous noise to the lack of infrastructure, several reasons may make a location inappropriate for music production. On this basis, engineers/

producers must consider all possible issues that might affect the result of sessions and discuss these with the rest of the team, e.g. artists, A&R, etc. Some of the aforementioned issues include:

- Unsuitability of equipment
- Difficulty in access (transport)
- Noise level constraints
- Size limitations
- Restricted access to mains power
- Environmental conditions, e.g. ambient noise, air conditioning, etc.
- Health and safety issues
- Lack of services, e.g. catering, parking, Internet access, etc.

It is important to note that in some cases what initially appears to be a hindrance may not have a negative effect on production, e.g. nature-related extraneous sounds that may become the 'sonic imprint' of a record.

SIZE CONSIDERATIONS

Recordists must examine session details regarding personnel closely, as these frequently dictate specific requirements regarding venue layout, size, etc. As an example, sessions involving classical musicians may surprise engineers that normally work with pop music, as the performers might require a greater amount of 'personal space' than what might be expected (a minimum of approximately 2 to 3 square metres per musician).

Other key aspects that are affected by venue size include the isolation between sound sources, reverberation time, etc. and all of these factors must be investigated and considered during the 'recce'.

EQUIPMENT RENTAL

It is common for large recording projects to require the rental of equipment, e.g. instruments, amplifiers, microphones, etc. These should be sourced from a trusted supplier and added to the audio inventory list.

THE SESSION PLAN

The session plan should be the final product of the planning stage, summarising all relevant information regarding production requirements and scheduling. This document must ideally include:

- Date, time and nature of sessions, e.g. tracking, overdub or bouncing
- Venue
- Suggested song(s) to work on (with key and tempo)
- Personnel and instrumentation breakdown (including call times)
- Engineering notes, e.g. click-track, backing track, general musician placement, music and microphone stands, number of cue mixes, etc.
- General requirements, e.g. media, catering, 'atmosphere'-related items, e.g. candles, rugs, etc.

It is important to note that under pressurised conditions, e.g. a busy production house or studio, the session plan may never be committed to paper, existing only in the producer's mind (or diary). It is nevertheless important to remember that planning equates to control and even in extreme conditions individuals can benefit greatly from mapping out their strategies.

SESSION SCHEDULE / DATE PLANNING

The scheduling of recording dates is one of the most challenging steps of session planning, requiring producers to be extremely organised and sensible when making projections. Ideally, only after extensive forethought should a schedule entry be made and backup plans are indispensable if costly studio time is at stake.

The allocation of funds or the breakdown of the budget between the different stages of production is dependant on a number of variables, e.g. how prepared the performers appear to be, etc., and figures will change as projects develop. Producers should always work with the most current, updated version of the budget and must be prepared to react immediately to situations as they arise. It is important to remember that projects must be finalised within their given budget and in many cases slight overestimation of costs may be advisable, i.e. the production team should err on the side of caution, without being unrealistic in their planning.

While it may not be practical to setout fixed, ideal time-related guidelines for the scheduling of sessions, it is certainly possible to split the production process into stages, allowing for resources and time to be allocated proportionally.

The following is an example of how the recording phase may be split:

1. Basic Tracking

   Loud rhythm instruments, e.g. drums, bass, electric guitar, electric keyboards, synthesisers, percussion (and possibly 'scratch vocals' for navigation and 'feel')

2. Overdubbing
   - Quieter rhythm instruments, e.g. piano, acoustic guitar
   - Lead vocals
   - Backing vocals
   - Instrumental solos
   - Special sound effects, hand percussion
   - Strings, horns, woodwinds, etc.

Upon careful examination of the steps necessary for the completion of a given project, it should be possible to estimate how the budget is to be distributed and one of the roles of the producer is to ensure that artists stick to the plan, while minimising the impact of finances on creativity.

PLANNING THE SOUND-CHECK

The inventory of audio tools allows for a sound-check signal path chart to be drafted. This preliminary plan of action should list suggested equipment chains for the different signals to be recorded, e.g. possible combinations of microphones and preamplifiers for acoustic sources. The selection of devices to be interconnected should be based on technical data and familiarity with the equipment, and it should provide the team with enough variety and/or backup options. The following is an example of a sound-check signal path chart for a singer-songwriter vocal and piano recording:

A – Main Vocals

Microphones	**Preamplifiers**
AKG C12	Neve 1073 DPA
Neumann U87	API 512C
Shure SM7	Focusrite Red 1

B – Grand Piano

Microphones	**Preamplifiers**
Neumann KM184 (x2)	Millennia HV-3C
AKG C414 (x2)	Manley Dual Mono
Earthworks QTC50 (x2)	Focusrite Red 8

From the preceding example it is possible to gather that the producer wants to try recording the main vocals and the piano using three different models of microphones and preamplifiers. These should be tested in all possible combinations, unless at some point of the audition stage a given chain is deemed ideal and further testing appears unnecessary.

A sound-check plan can help streamline the initial stages of the recording process, especially if an assistant is able to setup all microphones and preamps for A/B comparison prior to the beginning of sessions. This should allow the production team to spend more time auditioning signals and to focus on the aesthetic evaluation of sounds.

As producers combine all the knowledge gathered throughout the preproduction stage and finalise the session plan, the team should be ready to start production and enter the recording environment. At this stage, a few reminders can be extremely helpful and save all involved time and money. The following are some important final instructions for the performers:

- Be well prepared and warm up before sessions (do not rely excessively on inspiration).
- Practise using headphones and with a click-track (if sessions will require their use).
- Try to be focused and minimise possible distractions (do not invite unnecessary people to sessions).
- Change strings, drum heads, etc. ('break them in'), purchase spares and check instrument intonation, oil moving parts, etc.
- Check communication channels frequently, e.g. phones, email, etc.

LIVE SOUND TECHNICAL RIDERS VS. STUDIO SESSION PLANS

Recordists can benefit from analysing live concert technical riders, as such documents can help them create efficient and comprehensive session plans. Good technical riders commonly incorporate all relevant information regarding personnel, equipment, stage plans, etc., and as critical documents they must be assembled with extreme caution and attention to detail.

CHAPTER

5 PRODUCTION

The following is excerpted from *Practical Recording Techniques* by Bruce Bartlett and Jenny Bartlett.

You can purchase a copy HERE.

MICROPHONE TECHNIQUE BASICS

Suppose you're going to mike a singer, a sax, or a guitar. Which mic should you choose? Where should you place it? Your mic technique has a powerful effect on the sound of your recordings. In this chapter we'll look at some general principles of miking that apply to all situations.

WHICH MIC SHOULD I USE?

Is there a "right" mic to use on a piano, a kick drum, or a guitar amp? No. Every microphone sounds different, and you choose the one that gives you the sound you want. Still, it helps to know about two main characteristics of mics that affect the sound: frequency response and polar pattern.

Most condenser mics have an extended high-frequency response—they reproduce sounds up to 15 or 20 kHz. This makes them great for cymbals or other instruments that need a detailed sound, such as acoustic guitar, strings, piano, and voice. Dynamic moving-coil microphones have a response good enough for drums, guitar amps, horns, and woodwinds. Loud drums and guitar amps sound dull if recorded with a flat-response mic; a mic with a presence peak (a boost around 5kHz) gives more edge or punch.

Suppose you are choosing a microphone for a particular instrument. In general, the frequency response of the mic should cover at least the frequencies produced by that instrument. For example, an acoustic guitar produces fundamental frequencies from 82 Hz to about 1 kHz, and produces harmonics from about 1 to 15 kHz. So a mic used on an acoustic guitar should have a frequency response of at least 82 Hz to 15 kHz if you want to record the guitar accurately. **Table 7.1** shows the frequency ranges of various instruments.

The polar pattern of a mic affects how much leakage and ambience it picks up. Leakage is unwanted sound from instruments other than the one at which the mic is aimed.

Ambience is the acoustics of the recording room—its early reflections and reverb. The more leakage and ambience you pick up, the more distant the instrument sounds. An omni mic picks up more ambience and leakage than a directional mic when both are the same distance from an instrument. So an omni tends to sound more distant. To compensate, you have to mike closer with an omni.

TABLE 7.1 ■ FREQUENCY RANGES OF VARIOUS MUSICAL INSTRUMENTS

Instrument	Fundamentals (Hz)	Harmonics (kHz)
Flute	261-2349	3-8
Oboe	261-1568	2-12
Clarinet	165-1568	2-10
Bassoon	62-587	1-7
Trumpet	165-988	1-7.5
French Horn	87-880	1-6
Trombone	73-587	1-7.5
Tuba	49-587	1-4
Snare Drum	100-200	1-20
Kick Drum	30-147	1-6
Cymbals	300-587	1-15
Violin	196-3136	4-15
Viola	131-1175	2-8.5
Cello	65-698	1-6.5
Acoustic Bass	41-294	1-5
Electric Bass	41-300	1-7
Acoustic Guitar	82-988	1-15
Electric Guitar	82-1319	1-3.5 (through amp)
Electric Guitar	82-1319	1-15 (direct)
Piano	28-4196	5-8
Bass (Voice)	87-392	1-12
Tenor (Voice)	131-494	1-12
Alto (Voice)	175-698	2-12
Soprano (Voice)	247-1175	2-12

HOW MANY MICS?

The number of mics you need varies with what you're recording. If you want to record an overall acoustic blend of the instruments and room ambience, use just two microphones or a stereo mic **(Figure 7.1)**. This method works great on an orchestra, symphonic band, choir, string quartet, pipe organ, small folk group, or a piano/voice recital. Stereo miking is covered in detail later in this chapter. To record a pop-music group, you mike each instrument or instrumental section. Then you adjust the mixer volume control for each mic to control the balance between instruments **(Figure 7.2)**.

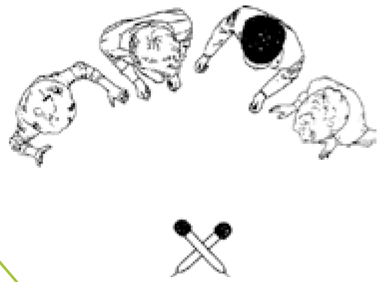


FIGURE 7.1
Overall miking of a musical ensemble with two distant microphones.

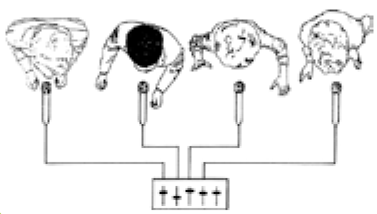


FIGURE 7.2
Individual miking with multiple close mics and a mixer.

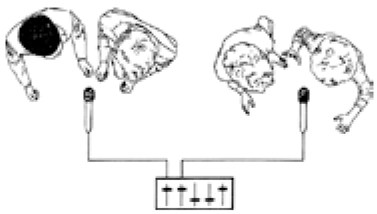


FIGURE 7.3
Multiple miking with several sound sources on each microphone.

To get the clearest sound, don't use two mics when one will do the job. Sometimes you can pick up two or more sound sources with one mic **(Figure 7.3)**. You could mike a brass section of four players with one mic on four players, or with two mics on every two players. Or mike a choir in a studio in four groups: put one mic on the basses, one on the sopranos, and so on.

Picking up more than one instrument with one mic has a problem: during mixdown, you can't adjust the balance among instruments recorded on the same track. You have to balance the instruments before recording them. Monitor the mic, and listen to see if any instrument is too quiet. If so, move it closer to the mic.

HOW CLOSE SHOULD I PLACE THE MIC?

Once you've chosen a mic for an instrument, how close should the mic be? Mike a few inches away to get a tight, present sound; mike farther away for a distant, spacious sound. (Try it to hear the effect.) *Play audio clip 17 at www.taylorandfrancis.com/cw/bartlett-9780240821535*. The farther a mic is from the instrument, the more ambience, leakage, and background noise it picks up. So mike close to reject these unwanted sounds. Mike farther away to add a live, loose, airy feel to overdubs of drums, lead- guitar solos, horns, etc. You'll need a room with good acoustics for best results.

Close miking sounds close; distant miking sounds distant. Here's why. If you put a mic close to an instrument, the sound at the mic is loud. So you need to turn up the mic gain on your mixer only a little to get a full recording level. And because the gain is low, you pick up very little reverb, leakage, and background noise **(Figure 7.4A)**.

If you put a mic far from an instrument, the sound at the mic is quiet. You'll need to turn up the mic gain a lot to get a full recording level. And because the gain is high, you pick up a lot of reverb, leakage, and background noise **(Figure 7.4B)**.

If the mic is very far away—maybe 10 feet—it's called an ambience mic or room mic. It picks up mostly room reverb. A popular mic for ambience is a boundary microphone taped to the wall. You mix it with the usual close mics to add a sense of space. Use two for stereo. When you record a live concert, you might want to place ambience mics over the audience, aiming at them from the front of the hall, to pick up the crowd reaction and the hall acoustics.

Classical music is always recorded at a distance (about 4 to 20 feet away) so that the mics will pick up reverb from the concert hall. It's a desirable part of the sound.

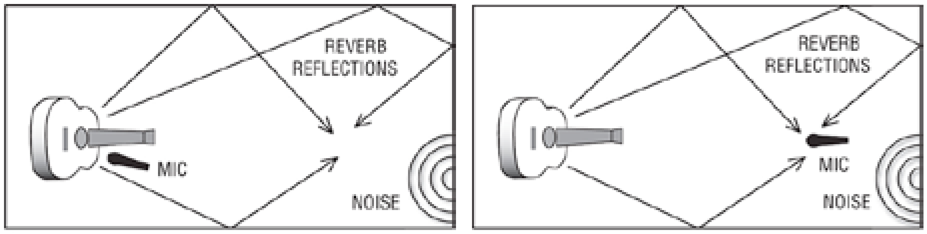


FIGURE 7.4
(a) A close microphone picks up mainly direct sound, which results in a close sound quality. (b) A distant microphone picks up mainly reflected sound, which results in a distant sound quality.

LEAKAGE (BLEED OR SPILL)

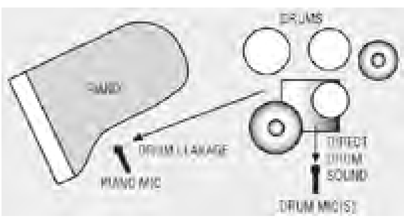


FIGURE 7.5
Example of leakage. The piano mic picks up leadage from the drums, which changes the close drum sound to distant.

Suppose you're close-miking a drum set and a piano at the same time **(Figure 7.5)**. When you listen to the drum mics alone, you hear a close, clear sound. But when you mix in the piano mic, that nice, tight drum sound degrades into a distant, muddy sound. That's because the drum sound leaked into the piano mic, which picked up a distant drum sound from across the room.

Audio clip 11 at www.taylorandfrancis.com/cw/bartlett-9780240821535 is an example of leakage.

There are many ways to reduce leakage:

- Mike each instrument closely. That way the sound level at each mic is high. Then you can turn down the mixer gain of each mic, which reduces leakage at the same time.
- Overdub each instrument one at a time.
- Record direct. Record an acoustic guitar off its pickup during tracking, then overdub the guitar with a mic. Record an electric guitar off its pickup during tracking, then play the guitar signal through a guitar-amp modeling plug-in during mixdown. Or record the electric guitar through a Line 6 Pod, which is a guitar-amp emulator.
- Filter out frequencies above and below the range of each instrument.
- Use directional mics (cardioid, etc.) instead of omni mics.
- Record in a large, fairly dead studio. In such a room, leakage reflected from the walls is weak.
- Put portable walls (goboes) between instruments.
- Use noise gates on drum tracks.

DON'T MIKE TOO CLOSE

Miking too close can color the recorded tone quality of an instrument. If you mike very close, you might hear a bassy or honky tone instead of a natural sound. Why? Most musical instruments are designed to sound best at a distance, at least 1½ feet away. The sound of an instrument needs some space to develop. A mic placed a foot or two away tends to pick up a well-balanced, natural sound. That is, it picks up a blend of all the parts of the instrument that contribute to its character or timbre.

Think of a musical instrument as a loudspeaker with a woofer, midrange, and tweeter. If you place a mic a few feet away, it will pick up the sound of the loudspeaker accurately. But if you place the mic close to the woofer, the sound will be bassy. Similarly, if you mike close to an instrument, you emphasize the part of the instrument that the microphone is near. The tone quality picked up very close may not reflect the tone quality of the entire instrument.

Suppose you place a mic next to the sound hole of an acoustic guitar, which resonates around 80 to 100 Hz. A microphone placed there hears this bassy resonance, giving a boomy recorded timbre that does not exist at a greater miking distance. To make the guitar sound more natural when miked close to the sound hole, you need to roll off the excess bass on your mixer, or use a mic with a bass rolloff in its frequency response.

The sax projects highs from the bell, but projects mids and lows from the tone holes. So if you mike close to the bell, you miss the warmth and body from the tone holes. All that's left at the bell is a harsh tone quality. You might like that sound, but if not, move the mic out and up to pick up the entire instrument. If leakage forces you to mike close, change the mic or use equalization (EQ).

Usually, you get a natural sound if you put the mic as far from the source as the source is big. That way, the mic picks up all the sound-radiating parts of the instrument about equally. For example, if the body of an acoustic guitar is 18 inches long, place the mic 18 inches away to get a natural tonal balance. If this sounds too distant or hollow, move in a little closer.

WHERE SHOULD I PLACE THE MIC?

Suppose you have a mic placed a certain distance from an instrument. If you move the mic left, right, up, or down, you change the recorded tone quality. In one spot, the instrument might sound bassy; in another spot, it might sound natural, and so on. So, to find a good mic position, simply place the mic in different locations—and monitor the results—until you find one that sounds good to you.

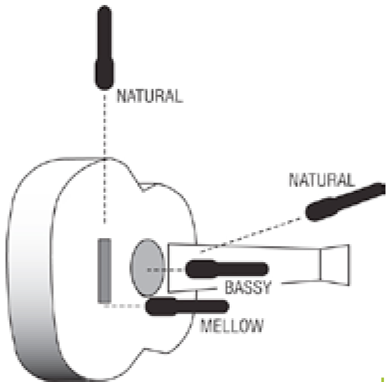


FIGURE 7.6
Microphone placement affects the recorded tonal balance.

Here's another way to do the same thing. Close one ear with your finger, listen to the instrument with the other ear, and move around until you find a spot that sounds good. Put the mic there. Then make a recording and see if it sounds the same as what you heard live. Don't try this with kick drums or screaming guitar amps! You could also move a mic around while monitoring its signal with good headphones.

Why does moving the mic change the tone quality? A musical instrument radiates a different tone quality in each direction. Also, each part of the instrument produces a different tone quality. For example, Figure 7.6 shows the tonal balances picked up at various spots near a guitar.

Audio clip 18 at www.taylorandfrancis.com/cw/bartlett-9780240821535 illustrates the effect of mic placement on guitar tonal balance. Audio clip 19 demonstrates close and distant stereo miking of the acoustic guitar.

Other instruments work the same way. A trumpet radiates strong highs directly out of the bell, but does not project them to the sides. So a trumpet sounds bright when miked on-axis to the bell and sounds more natural or mellow when miked off to one side. A grand piano miked one foot over the middle strings sounds fairly natural, under the soundboard sounds bassy and dull, and in a sound hole it sounds mid-rangy.

It pays to experiment with all sorts of mic positions until you find a sound you like. There is no one right way to place the mics because you place them to get the tonal balance you want.

THE THREE-TO-ONE RULE

FIGURE 7.9
The three-to-one rule of microphone placement avoids phase interference between microphone signals.

Suppose you're recording a singer/guitarist. There's a mic on the singer and a mic on the acoustic guitar. When you monitor the mix, something's wrong: the singer's voice sounds hollow or filtered. You're hearing the effect of phase interference.

In general, if two mics pick up the same sound source at different distances, and their signals are mixed to the same channel, this might cause phase cancellations. These are peaks and dips in the frequency response—a comb filter—caused by some frequencies combining out of phase. The result is a colored, filtered tone quality that sounds like mild flanging.

To prevent this problem, follow the three-to-one rule: space the mics at least three times the mic-to-source distance (as in Figure 7.9). For example, if two mics are 12 inches apart, they should be less than 4 inches from their sound sources to prevent phase cancellations. *Play audio clip 21 at www.taylorandfrancis.com/cw/bartlett-9780240821535*. The mics can be closer together than 3:1 if you use two cardioid mics aiming in opposite directions. The goal is to get at least 9 dB of separation between recorded tracks.

What if you pick up an instrument with two mics that are panned left and right? You don't get phase interference. Instead you get stereo imaging.

OFF-AXIS COLORATION

Some mics have off-axis coloration—a dull or colored effect on sound sources that are not directly in front of the mic. Try to aim the mic at sound sources that put out high frequencies, such as cymbals. When you pick up a large source such as an orchestra, use a mic that has the same response over a wide angle. Such a mic has similar polar patterns at middle and high frequencies. Most large-diaphragm mics have more off-axis coloration than smaller mics (under 1 inch).

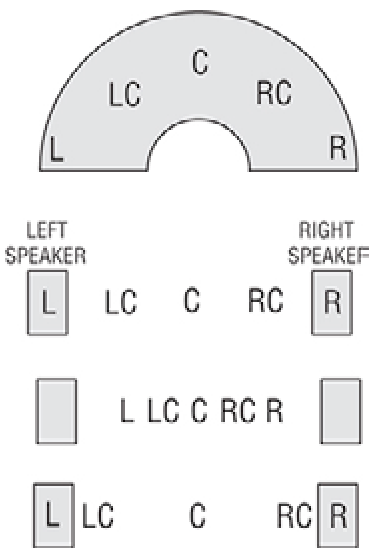


FIGURE 7.10
Stereo localization effects.
(a) Orchestra instrument locations (top view).
(b) Images localized accurately between speakers (the listener's perception).
(c) Narrow stage effect.
(d) Exaggerated separation effect.

STEREO MIC TECHNIQUES

Stereo mic techniques capture the sound of a musical group as a whole, using only two or three microphones. When you play back a stereo recording, you hear phantom images of the instruments in various spots between the speakers. These image locations— left to right, front to back—correspond to the instrument locations during the recording session.

Stereo miking is the preferred way to record classical-music ensembles and soloists. In the studio, you can stereo-mike a piano, drum set cymbals, vibraphone, harmony singers, or other large sound sources.

GOALS OF STEREO MIKING

One goal is accurate localization. That is, instruments in the center of the group are reproduced midway between the two speakers. Instruments at the sides of the group are heard from the left or right speaker. Instruments halfway to one side are heard halfway to one side, and so on.

Figure 7.10 shows three stereo localization effects. **Figure 7.10A** shows some instrument positions in an orchestra: left, left-center, center, right-center, right. In **Figure 7.10B**, the reproduced images of these instruments are accurately localized between the speakers. The stereo spread, or stage width, extends from speaker to speaker. (You might want to record a string quartet with a narrower spread.)

If you space or angle the mics too close together, you get a narrow stage effect **(Figure 7.10C)**. If you space or angle the mics too far apart, you hear exaggerated separation **(Figure 7.10D)**. That is, instruments halfway to one side are heard near the left or right speaker.

To judge stereo effects, you have to sit exactly between your monitor speakers (the same distance from each). Sit as far from the speakers as the spacing between them. Then the speakers appear to be 60 degrees apart. This is about the same angle an orchestra fills when viewed from a typical ideal seat in the audience (say, tenth row center). If you sit off-center, the images shift toward the side on which you're sitting and are less sharp.

TYPES OF STEREO MIC TECHNIQUES

To make a stereo recording, you use one of these basic techniques:

- Coincident pair (XY or MS)
- Spaced pair (AB)
- Near-coincident pair (ORTF, etc.)
- Baffled pair (sphere, OSS, SASS, PZM wedge, etc.)

Let's look at each technique.

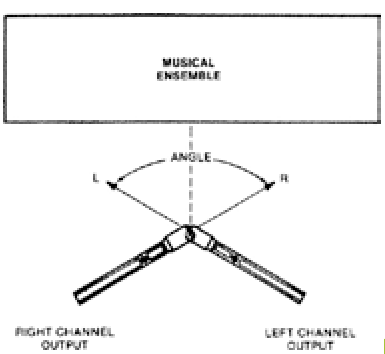


FIGURE 7.11
Coincident-pair technique.

COINCIDENT PAIR

With this method, you mount two directional mics with grilles touching, diaphragms one above the other, and angled apart **(Figure 7.11)**. For example, mount two cardioid mics with one grille above the other, and angle them 120 degrees apart. You can use other patterns too: supercardioid, hypercardioid, or bidirectional. The wider the angle between mics, the wider the stereo spread. If the angle is too wide, center images will be weak (there will be a "hole in the middle").

How does this technique make images you can localize? A directional mic is most sensitive to sounds in front of the mic (on-axis) and progressively less sensitive to sounds arriving off-axis. That is, a directional mic puts out a high-level signal from the sound source it's aimed at, and produces lower-level signals from other sound sources.

The coincident pair uses two directional mics that are angled symmetrically from the center line (Figure 7.11). Instruments in the center of the group make the same signal from each mic. During playback, you hear a phantom image of the center instruments midway between your speakers. That's because identical signals in each channel produce an image in the center.

If an instrument is off-center to the right, it is more on-axis to the right-aiming mic than to the left-aiming mic. So the right mic will produce a higher level signal than the left mic. During playback of this recording, the right speaker will play at a higher level than the left speaker. This reproduces the image off-center to the right—where the instrument was during recording.

The coincident pair codes instrument positions into level differences between channels. During playback, the brain decodes these level differences back into corresponding image locations. A pan pot in a mixing console works on the same principle. If one channel is 15 to 20 dB louder than the other, the image shifts all the way to the louder speaker.

Suppose we want the right side of the orchestra to be reproduced at the right speaker. That means the far-right musicians must produce a signal level 20 dB higher from the right mic than from the left mic. This happens when the mics are angled far enough apart. The correct angle depends on the polar pattern.

Instruments partway off-center produce interchannel level differences less than 20 dB, so you hear them partway off-center.

Listening tests have shown that coincident cardioid mics tend to reproduce the musical group with a narrow stereo spread. That is, the group does not spread all the way between speakers.

A coincident-pair method with excellent localization is the Blumlein array. It uses two bidirectional mics angled 90 degrees apart and facing the left and right sides of the group.

A special form of the coincident-pair technique is mid-side or MS (Figure 7.12). In this method, a cardioid or omni mic faces the middle of the orchestra. A matrix circuit sums and differences the cardioid mic with a bidirectional mic aiming to the sides. This produces left- and right-channel signals. You can remotely control the stereo spread by changing the ratio of the mid signal to the side signal. This remote control is useful at live concerts, where you can't physically adjust the mics during the concert. MS localization can be accurate.

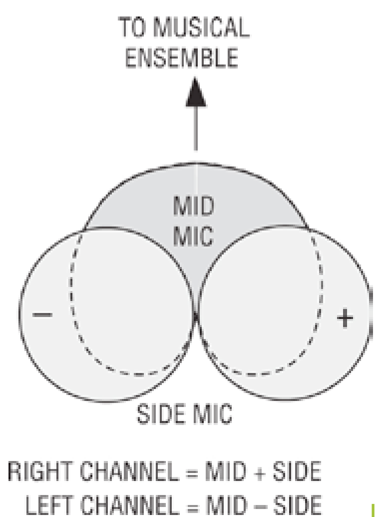


FIGURE 7.12
Mid-side (MS) technique.

To make coincident recordings sound more spacious, boost the bass 4 dB (+2 dB at 600 Hz) in the L–R or side signal.

A recording made with coincident mics is mono compatible. That is, the frequency response is the same in mono or stereo. Because the mics occupy almost the same point in space, there is no time or phase difference between their signals. And when you combine them to mono, there are no phase cancellations to degrade the frequency response. If you expect that your recordings will be heard in mono (say, on TV), then you'll probably want to use coincident methods.

SPACED PAIR

Here, you mount two identical mics several feet apart and aim them straight ahead (Figure 7.13). The mics can have any polar pattern, but omni is most popular for this method. The greater the spacing between mics, the greater the stereo spread.

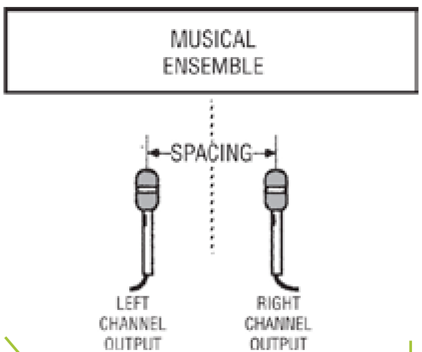


FIGURE 7.13
Spaced-pair technique.

How does this method work? Instruments in the center of the group make the same signal from each mic. When you play back this recording, you hear a phantom image of the center instruments midway between your speakers. If an instrument is off-center, it is closer to one mic than the other, so its sound reaches the closer microphone before it reaches the other one. Both mics make about the same signal, except that one mic signal is delayed compared with the other.

If you send a signal to two speakers with one channel delayed, the sound image shifts off-center. With a spaced-pair recording, off-center instruments produce a delay in one mic channel, so they are reproduced off-center.

The spaced pair codes instrument positions into time differences between channels. During playback, the brain decodes these time differences back into corresponding image locations.

A delay of 1.2 msec is enough to shift an image all the way to one speaker. You can use this fact when you set up the mics. Suppose you want to hear the right side of the orchestra from the right speaker. The sound from the right-side musicians must reach the right mic about 1.2 msec before it reaches the left mic. To make this happen, space the mics about 2 to 3 feet apart. This spacing creates the correct delay to place right-side instruments at the right speaker. Instruments partway off-center make interchannel delays less than 1.2 msec, so they are reproduced partway off-center.

If the spacing between mics is, say, 12 feet, then instruments that are slightly off-center produce delays between channels that are greater than 1.2 msec. This places their images at the left or right speaker. I call this "exaggerated separation" or a "ping-pong" effect **(Figure 7.10D)**.

On the other hand, if the mics are too close together, the delays produced will be too small to provide much stereo spread. Also, the mics will tend to emphasize instruments in the center because the mics are closest to them.

To record a good musical balance of an orchestra, you might need to space the mics about 10 or 12 feet apart. But then you get too much separation. You could place a third mic midway between the outer pair and mix its output to both channels. That way, you pick up a good balance, and you hear an accurate stereo spread.

The spaced-pair method tends to make off-center images unfocused or hard to localize. Why? Spaced-pair recordings have time differences between channels. Stereo images produced solely by time differences are unfocused. You still hear the center instruments clearly in the center, but off-center instruments are hard to pinpoint. Spaced-pair miking is a good choice if you want the sonic images to be diffuse or blended, instead of sharply focused.

Another flaw of spaced mics is that if you mix both mics to mono, you may get phase cancellations of various frequencies. This may or may not be audible.

Spaced mics, however, give a "warm" sense of ambience, in which the concert hall reverb seems to surround the instruments and, sometimes, the listener. Here's why: the two channels of recorded reverb are incoherent, that is, they have random phase relationships. Incoherent signals from stereo speakers sound diffuse and spacious.

Because spaced mics pick up reverb incoherently, it sounds diffuse and spacious. The simulated spaciousness caused by this phasiness is not necessarily realistic, but it is pleasant to many listeners.

Another advantage of the spaced pair is that you can use omni mics. An omni condenser mic has deeper bass than a uni condenser mic.

NEAR-COINCIDENT PAIR

In this method, you angle apart two directional mics, and space their grilles a few inches apart horizontally **(Figure 7.14)**. Even a few inches of spacing increases the stereo spread and adds a sense of ambient warmth or air to the recording. The greater the angle or spacing between mics, the greater the stereo spread. If the angle is too wide, center images will be weak (there will be a "hole in the middle").

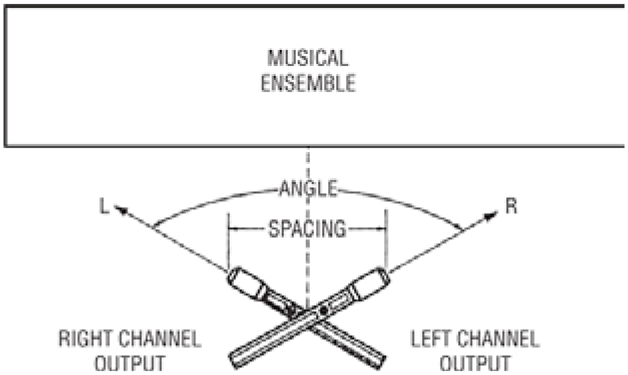


FIGURE 7.14
Near-coincident pair technique.

How does this method work? Angling directional mics produces level differences between channels. Spacing mics produces time differences. The level differences and time differences combine to create the stereo effect.

If the angling or spacing is too great, you get exaggerated separation. If the angling or spacing is too small, you'll hear a narrow stereo spread.

A common near-coincident method is the ORTF system, which uses two cardioids angled 110 degrees apart and spaced 7 inches (17 cm) horizontally. Usually this method gives accurate localization. That is, instruments at the sides of the orchestra are reproduced at or very near the speakers, and instruments halfway to one side are reproduced about halfway to one side.

BAFFLED OMNI PAIR

This method uses two omni mics, usually ear-spaced, and separated by either a hard or soft baffle **(Figure 7.15)**. To create stereo, it uses time differences at low frequencies and level differences at high frequencies. The spacing between mics creates time differences. The baffle creates a sound shadow (reduced high frequencies) at the mic farthest from the source. Between the two channels, there are spectral differences—differences in frequency response.

Some examples of baffled-omni pairs are the Schoeps or Neumann sphere microphones, and the Jecklin Disk.

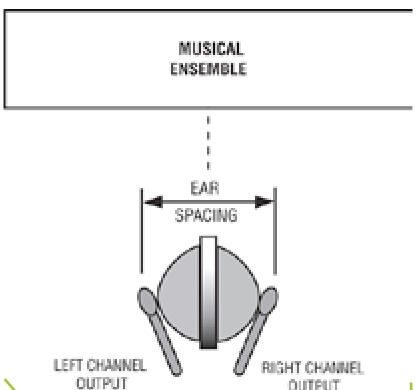


FIGURE 7.15
Baffled-omni technique.

COMPARING THE FOUR TECHNIQUES

Coincident pair:

- Uses two directional mics angled apart with grilles touching.
- Level differences between channels produce the stereo effect.
- Images are sharp.
- Stereo spread ranges from narrow to accurate.
- Signals are mono compatible.

Spaced pair:

- Uses two mics spaced several feet apart, aiming straight ahead.
- Time differences between channels produce the stereo effect.
- Off-center images are diffuse.
- Stereo spread tends to be exaggerated unless a third center mic is used, or unless spacing is under 2 to 3 feet.
- Provides a warm sense of ambience.
- Tends not to be mono compatible, but there are exceptions.
- Good low-frequency response if you use omni condensers.

Near-coincident pair:

- Uses two directional mics angled apart and spaced a few inches apart horizontally.
- Level and time differences between channels produce the stereo effect.
- Images are sharp.
- Stereo spread tends to be accurate. Provides a greater sense of air than coincident methods.
- Tends not to be mono compatible.

Play audio clip 22 at www.taylorandfrancis.com/cw/bartlett-9780240821535 to hear a comparison of the coincident, near-coincident, and spaced-pair techniques.

Baffled omni pair:

- Uses two omni mics, usually ear-spaced, with a baffle between them.
- Level, time, and spectral differences produce the stereo effect.
- Images are sharp.
- Stereo spread tends to be accurate but is not adjustable (except partly by panning).
- Good low-frequency response.

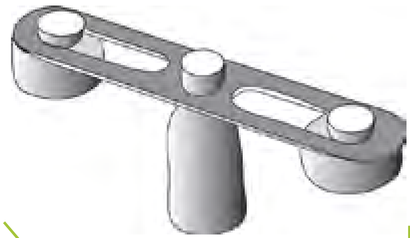


FIGURE 7.16
Stereo mic adapter.

- Good imaging with headphones.
- Provides more air than coincident methods.
- Tends not to be mono compatible, but there are exceptions.

HARDWARE

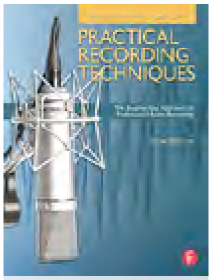
A handy device is a stereo mic adapter or stereo bar **(Figure 7.16)**. It mounts two mics on a single stand, and lets you adjust the angle and spacing. You might prefer to use a stereo mic instead of two mics. It has two mic capsules in a single housing for convenience.

HOW TO TEST IMAGING

Here's a way to check the stereo imaging of a mic technique.

1. Set up the stereo mic array in front of a stage.
2. Record yourself speaking from various locations on stage where the instruments will be—center, half-right, far right, half-left, far left. Announce your position.
3. Play back the recording over speakers.

You'll hear how accurately the technique translated your positions, and you'll hear how sharp the images are. We looked at several mic arrays to record in stereo. Each has its pros and cons. Which method you choose depends on the sonic compromises you're willing to make.



The following is excerpted from *Practical Recording Techniques* by Bruce Bartlett and Jenny Bartlett.

You can purchase a copy HERE.

This chapter describes some ways to select and place mics for musical instruments and vocals. These techniques are popular, but they're just suggested starting points. Feel free to experiment.

Before you mike an instrument, listen to it live in the studio, so you know what sound you're starting with. You might want to duplicate that sound through your monitor speakers.

ELECTRIC GUITAR

Let's start by looking at the chain of guitar, effects, amplifier, and speaker. At each point in the chain where you record, you'll get a different sound **(Figure 8.1)**.

1. The electric guitar puts out an electrical signal that sounds clean and clear.
2. This signal might go through some effects boxes, such as distortion, wah wah, compression, chorus, or stereo effects.
3. Then the signal goes through a guitar amp, which boosts the signal and adds distortion. At the amplifier output (preamp out or external speaker jack), the sound is very bright and edgy.
4. The distorted amp signal is played by the speaker in the amp. Because the speaker rolls off above 4 kHz, it takes the edge off the distortion and makes it more pleasant.

You can record the electric guitar in many ways **(Figure 8.1)**:

- With a mic in front of the guitar amp
- With a direct box
- Both miked and direct
- Through a signal processor or stomp box

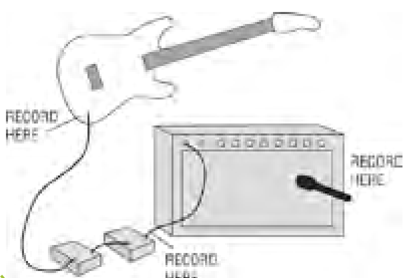


FIGURE 8.1
Three places to record the electric guitar.

The song you're recording will tell you what method it wants. Just mike the amp when you want a rough, raw sound with tube distortion and speaker coloration. Rock 'n' roll or heavy metal usually sounds best with a miked amp. If you record through a direct box, the sound is clean and clear, with crisp highs and deep lows. That might work for quiet jazz or R&B. Use whatever sounds right for the particular song you're recording.

First, try to kill any hum you hear from the guitar amp. Turn up the guitar's volume and treble controls so that the guitar signal overrides hum and noise picked up by the guitar cable. Ask the guitarist to move around, or rotate, to find a spot in the room where hum disappears. Flip the polarity switch on the amp to the lowest-hum position. To remove buzzes between guitar notes, try a noise gate, or ask the player to keep his or her hands on the strings.

MIKING THE AMP

Small practice amplifiers tend to be better for recording than large, noisy stage amps. If you use a small one, place it on a chair to avoid picking up sound reflections from the floor (unless you like that effect).

A common mic for the guitar amp is a cardioid dynamic type with a "presence peak" in its frequency response (a boost around 5 kHz). The cardioid pattern reduces leakage (off-mic sounds from other instruments). The dynamic type handles loud sounds without distorting, and the presence peak adds "bite." Of course, you can use any mic that sounds good to you.

As a starting point, try miking the amp about an inch from the grille near a speaker cone, slightly off-center—where the cone meets the dome.

The closer you mike the amp, the bassier the tone. The farther off-center the mic is, the duller the tone. Often, distant miking sounds great when you overdub a lead guitar solo played through a stack of speakers in a live room. Try a boundary mic on the floor or on the wall, several feet away.

RECORDING DIRECT

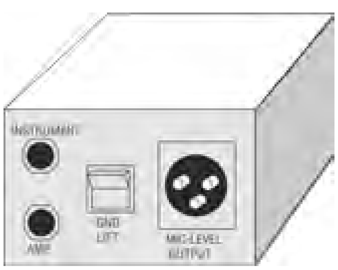


FIGURE 8.2
Typical direct box.

Now let's look at recording direct (also known as direct injection or DI). The electric guitar produces an electrical signal that you can plug into your mixer. You bypass the mic and guitar amp, so the sound is clean and clear. Just remember that amp distortion is desirable in some songs.

Mixer mic inputs tend to have an impedance (Z) around 1500 ohms. But a guitar pickup is several thousand ohms. So if you plug a high-Z electric guitar directly into a mic input, the input will load down the pickup and give a thin or dull sound.

To get around this loading problem, use a direct box between the guitar and your mixer (Figure 8.2). The DI box has a high-Z input and low-Z output, thanks to a built-in transformer or circuit. Some mixers and audio interfaces have a high-Z (instrument) input jack built in, so you can plug the electric guitar or bass directly into this jack.

The direct box should have a ground-lift switch to prevent ground loops and hum. Set it to the position where you monitor the least hum. You might try a mix of direct sound and miked sound.

Play audio clip 23 at www.taylorandfrancis.com/cw/ bartlett-9780240821535 to hear demonstrations of electric-guitar recording methods.

It's a good idea to record the guitar direct on its own track even if you mike the amp. Later during mixdown, you can run the DI track through a guitar-amp simulator plugin, which might sound better than the real guitar amp did.

ELECTRIC GUITAR EFFECTS

If you want to record the guitarist's effects, connect the output of the effects boxes into the direct-box input. Many players have a rack of signal processors that creates their unique sound, and they just give you their direct feed. Be open to their suggestions, and be diplomatic about changing the sound. If they are studio players, they often have a better handle on effects than you might as the engineer.

You might want a "fat" or spacious lead-guitar sound. Here are some ways to get it:

- Send the guitar signal through a digital delay set to 20 to 30 msec. Pan guitar left, delay right. Adjust levels for nearly equal loudness from each speaker. (Watch out for phase cancellations in mono.)
- Send the guitar signal through a pitch-shifter, set for about 10 cents of pitch bending. Pan guitar left, pitch-shifted guitar right. (A cent is 1/100 of an equal-temperament semitone. There are 100 cents in a half-tone or semitone interval of pitch).
- Record two guitarists playing identical parts, and pan them left and right. This works great for rhythm-guitar parts in heavy metal.
- Double the guitar. Have the player rerecord the same part on an unused track while listening to the original part. Pan the original part left and pan the new part right.
- Add stereo reverb or stereo chorus.

Some guitar processors add many effects to an electric guitar, such as distortion, EQ, chorus, and compression. An example is the Line 6 Pod. You simply plug the electric guitar into the processor, adjust it for the desired sound, and record the signal direct. You wind up with a fully produced sound with a minimum of effort.

Reamping is a technique that lets you work on the amp's sound during mixdown rather than during recording. Record the guitar direct, then feed that track's signal into a guitar processor or miked guitar amp during mixdown. Use a low- to high-Z transformer between the track output and the processor or amp input. Record the processor or amp on an open track. In a digital audio workstation (DAW), you can start with a track of a direct-recorded guitar, then insert a guitar-amp modeling plug-in.

ELECTRIC BASS

BWAM, dik diddy bum. Do your bass tracks sound that clear? Or are they more muddy, like, "Bwuh, dip dubba duh"? Here's how to record the electric bass so it's clean and easy to hear in a mix.

As always, first you work on the sound of the instrument itself. Put on new strings if the old ones sound dull. Adjust the pickup screws (if any) for equal output from each string. Also adjust the intonation and tuning.

Usually, you record the electric bass direct for the cleanest possible sound. A direct pickup gives deeper lows than a miked amp, but the amp gives more midrange punch. You might want to mix the direct and miked sound. Use a condenser or dynamic mic with a good low-frequency response, placed 1 to 6 inches from the speaker.

When mixing a direct signal and a mic signal, make sure they are in phase with each other. To do this, set them to equal levels and reverse the polarity of the direct signal or the mic signal. The polarity that gives the most bass is correct.

Have the musician play some scales to see if any notes are louder than the rest. You might set a parametric equalizer to soften these notes, or use a compressor.

The bass guitar should be fairly constant in level (a dynamic range of about 6 dB) to be audible throughout the song, and to avoid clipping the recording on loud peaks. To do this, run the bass guitar through a compressor. Set the compression ratio to about 4:1, set the attack time fairly slow (8 to 20 msec) to preserve the attack transient, and set the release time fairly slow (1/2 second). If the release time is too fast, you get harmonic distortion.

EQ can make the bass guitar clearer. Try cutting around 60 to 80 Hz, or at 400 Hz. A boost at 2 to 2.5 kHz adds edge or slap, and a boost at 700 to 900 Hz adds "growl" and harmonic clarity. If you boost the lows around 100 Hz, try boosting at a lower frequency in the kick drum's EQ to keep their sounds distinct. A fretless bass will probably need different EQ or less EQ than a fretted bass.

Here are some ways to make the bass sound clean and well defined:

- Record the bass direct. Some direct boxes sound better than others, so experiment.
- Get a clear tone using the bass guitar's tone controls before applying any EQ.
- Use no reverb or echo on the bass.
- Have the bass player turn down the bass amp in the studio, just loud enough to play adequately. This reduces muddy-sounding bass leakage into other mics.
- Better yet, don't use the amp. Instead, have the musicians monitor the bass (and each other) with headphones.
- Have the bass player try new strings or a different bass. Some basses are better for recording than others. Use roundwound strings for a bright tone or flatwounds for a rounder tone.
- Ask the bass player to use the treble pickup near the bridge.
- Be sure to record the bass with enough edge or harmonics so the bass will be audible on small, cheap speakers.
- Try a bass-guitar signal processor such as the Zoom B1.
- Mix in a synth bass. For drum 'n' bass, dub-step, and electro-house genres, add some sub-bass at 30-60 Hz with space between notes, plus some mid-bass around 80-600 Hz. To create a pulsing effect, compress the synth bass with a kick drum as the sidechain input.

If the bass part is full and sustained, it's probably best to go for a mellow sound without much pluck. Let the kick drum define the rhythmic pattern. But if both the bass and kick are rhythmic and work independently, then you should hear the plucks. Listen to the song first, then get a bass sound appropriate for the music. A sharp, twangy timbre is seldom right for a ballad; a full, round tone will get lost in a fusion piece.

Often, a musician plays bass lines on a synth or sound module. The module is triggered from a keyboard, a sequencer, or a bass guitar plugged into a pitch-to-MIDI converter. Connect the module output to your mixer or audio interface line input.

Two effects boxes for the electric bass are the octave box and the bass chorus. The octave box takes the bass signal and drops it an octave in pitch. That is, it divides the bass signal's fundamental frequency in two. You put 82 Hz in; you get 41 Hz out. This gives an extra deep, growly sound. So does a 5-string bass.

A bass chorus gives a wavy, shimmering effect. Like a conventional chorus box, it detunes the signal and combines the detuned signal with the direct signal. Also, it removes the lowest frequencies from the detuned signal, so that the chorus effect doesn't thin out the sound.

SYNTHESIZER, DRUM MACHINE, AND ELECTRIC PIANO

For the most clarity, you usually DI a synth, MIDI sound module, drum machine, or electric piano. Set the volume on the instrument about three-quarters up to get a strong signal. Try to get the sound you want from patch settings rather than EQ.

Plug the instrument into a phone jack input on your mixer, or use a direct box. If you connect to a phone jack and hear hum, you probably have a ground loop. Here are some fixes:

- Power your mixer and the instrument from the same outlet strip. If necessary, use a thick extension cord between the outlet strip and the instrument.
- Use a direct box instead of a guitar cord, and set the ground-lift switch to the position where you monitor the least hum.
- To reduce hum from a low-cost synth, use battery power instead of an AC adapter.

A synth can sound dry and sterile. To get a livelier, funkier sound, you might run the synth signal into a power amp and speakers, and mike the speakers a few feet away.

If the keyboard player has several keyboards plugged into a keyboard mixer, you may want to record a premixed signal from that mixer's output. Record both outputs of stereo keyboards.

LESLIE ORGAN SPEAKER

This glorious device has a rotating dual-horn on top for highs and a woofer on the bottom for lows. Only one horn of the two makes sound; the other is for weight balance. The swirling, grungy sound comes from the phasiness and Doppler effect of the rotating horn, and from the distorted tube electronics that drive the speaker. Here

FIGURE 8.3
Miking a Leslie organ speaker.

are a few ways to record it **(Figure 8.3)**:

- In mono: Mike the top and bottom separately, 3 inches to 1 foot away. Aim the mics into the louvers. In the top mic's signal, roll off the lows below 150 Hz.
- In stereo: Record the top horn with a stereo mic or a pair of mics out front. Put a mic with a good low end on the bottom speaker, and pan it to center.

When you record the Leslie, watch out for wind noise from the rotating horn and buzz from the motor. Mike farther away if you monitor these noises.

Rather than recording an actual Hammond B3 organ and Leslie speaker, you might prefer to use a software emulation of those instruments: an organ soft synth or sample and a Leslie speaker plug-in. Trigger the synth or sample with a MIDI sequencer or MIDI controller. You can automate the horn rotation speed in the Leslie speaker plug-in.

DRUM SET

The first step is to make the drums sound good live in the studio. If the set sounds poor, you'll have a hard time making it sound great in the control room! You might put the drum set on a riser 1½ feet high to reduce bass leakage and to provide better eye contact between the drummer and the rest of the band. To reduce drum leakage into other mics, you could surround the set with goboes—padded thick-wood panels about 4 feet tall. For more isolation, place the set in a drum booth, a small padded room with windows. It's also common to overdub the set in a live room.

TUNING

One secret to creating a good drum sound lies in careful tuning. It's easier to record a killer sound if you tune the set to sound right in the studio before miking it.

First let's consider drum heads. Plain heads have the most ring or sustain, while heads with sound dots or hydraulic heads dampen the ring. Thin heads are best for recording because they have crisp attack and long sustain. Old heads become dull, so use new heads.

When you tune the toms, first take off the heads and remove the damping mechanism, which can rattle. Put just the top head on and hand-tighten the lugs. Then, using a drum key, tighten opposite pairs of lugs one at a time, one full turn. After you tighten all the lugs, repeat the process, tightening one-half turn. Then press on the head to stretch it. Continue tightening a half-turn at a time until you reach the pitch you want. You'll get the most pleasing tone when the heads are tuned within the range of the shell resonance. However, toms tuned very low can sound big and powerful.

To reduce ugly overtones, try to keep the tension the same around the head. While touching the center of the head, tap with a drumstick on the head near each lug. Adjust tension for equal pitch around the drum. Damping the head with a Dead Ringer or folded paper towel helps too. If you want a downward pitch bend after the head is struck, loosen one lug.

Keep the bottom head off the drum for the most projection and the broadest range of tuning. In this case, pack the bottom lugs with felt to prevent rattles. But you may want to add the bottom head for extra control of the sound. Projection is best if the bottom head is tighter than the top head—say, tuned a fourth above the top head. There will be a muted attack, an "open" tone, and some note bending. If you tune the bottom head looser than the top, the tone will be more "closed," with good attack.

With the kick drum (bass drum), a loose head gives lots of slap and attack, and almost no tone. The opposite is true for a tight head. Tune the head to complement the style of music. For more attack or click, use a hard beater.

Tune the snare drum with the snares off. A loose batter head or top head gives a deep, fat sound. A tight batter head sounds bright and crisp. With the snare head or bottom head loose, the tone is deep with little snare buzz, while a tight snare head yields a crisp snare response. Set the snare tension just to the point where the snare wires begin to "choke" the sound, then back off a little.

DAMPING AND NOISE CONTROL

Usually the heads should ring without any damping. But if the toms or snare drum ring too much, put some plastic damping rings (Dead Ringers) on them. Or tape some gauze pads, folded paper towels, or folded handkerchiefs to the edge of the heads. Put masking tape on three sides of the pad so that the untaped edge is free to vibrate and dampen the head motion. Don't overdo the damping, or the drum set will sound like cardboard boxes.

Oil the kick drum pedal to prevent squeaks. Tape rattling hardware in place.

Sometimes a snare drum buzzes in sympathetic vibration with a bass-guitar passage or a tom-tom fill. Try to control the buzz by wedging a thick cotton wad between the snares and the drum stand. Or tune the snare to a different pitch than the toms.

DRUM MIKING

Now you're ready to mike the set. For a tight sound, place a mic near each drum head. For a more open, airy sound, use fewer mics or mix in some room mics placed several feet away. Typical room mics are omni condensers or boundary mics. Compressing the room mics can provide an explosive sound.

Figure 8.4 shows typical mic placements for a rock drum set. Let's look at each part of the kit.

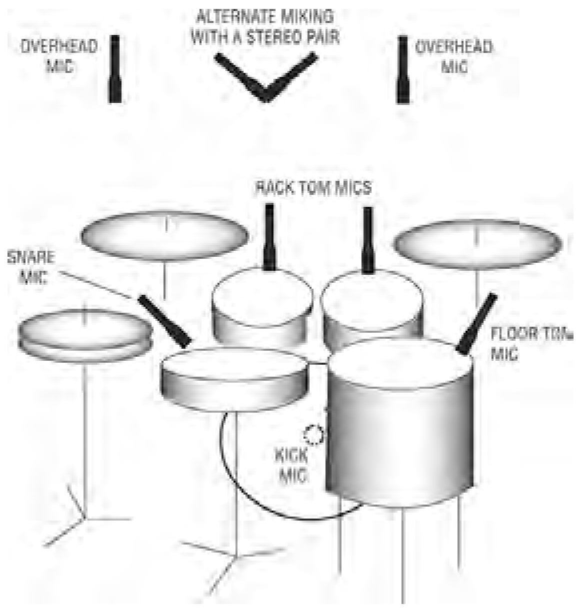


FIGURE 8.4
Typical mic placements for a rock drum set.

SNARE

The most popular type of mic for the snare is a cardioid dynamic with a presence peak. The cardioid pattern reduces leakage; its proximity effect boosts the bass for a fatter sound. The presence peak adds attack. You might prefer a cardioid condenser for its sharp transient response.

Bring the mic in from the front of the set on a boom. Place the mic even with the rim, 2 inches above the head (Figure 8.5). Angle the mic down to aim where the drummer hits, or attach a mini condenser mic to the side of the snare drum so it "looks at" the top head over the rim.

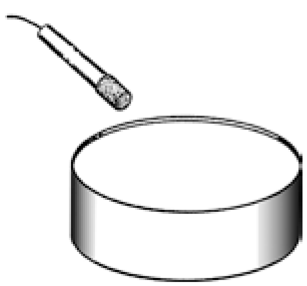


FIGURE 8.5
Snare-drum miking.

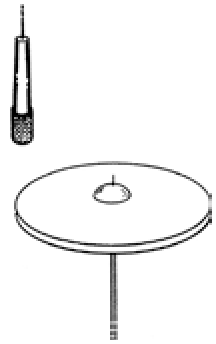


FIGURE 8.6
Hi-hat miking.

FIGURE 8.7
Tom-tom miking.

Some engineers mike both the top and bottom heads of the snare drum, with the microphones in opposite polarity. A mic under the snare drum gives a zippy sound; a mic over the snare drum gives a fuller sound. You might prefer to use just a top mic, and move it around until it picks up both the top head and snares. The sound is full with the mic near the top head, and thins out and becomes brighter as you move the mic toward the rim and down the side of the drum.

Whenever the hi-hat closes, it makes a puff of air that can "pop" the snare-drum mic. Place the snare mic so the air puff doesn't hit it. To prevent hi-hat leakage into the snare mic:

- Mike the snare closely.
- Bring the snare boom in under the hi-hat, and aim the snare mic away from the hi-hat.
- Use a piece of foam or pillow to block sound from the hi-hat.
- Use a de-esser on the snare.
- Don't play the hi-hat during tracking—overdub it later.

HI-HAT

Try a cardioid condenser mic about 6 inches over the cymbal edge that's farthest from the drummer **(Figure 8.6)**. To avoid the air puff just mentioned, don't mike the hi-hat off its side; mike it from above aiming down. This also reduces snare leakage. Filter out the lows below about 500 Hz. You may not need a hi-hat microphone, especially if you use room mics. Usually the overhead mics pick up enough hi-hat.

TOM-TOMS

You can mike the toms individually, or put a mic between each pair of toms. The first option sounds more bassy. Place a cardioid dynamic about 2 inches over the drumhead and 1 inch in from the rim, angled down about 45 degrees toward the head **(Figure 8.7)**. Again, the cardioid's proximity effect gives a full sound. Another way is to clip mini condenser mics to the toms, peeking over the top rim of each drum.

If the tom mics pick up too much of the cymbals, aim the "dead" rear of the tom mics at the cymbals. If y ou use a supercardioid or hypercardioid mic, aim the null of best rejection at the cymbals (125° off axis for supercardioid; 110° for hypercadioid). It's common to gate the toms to reduce the low rumble of vibrating heads.

FIGURE 8.8
Kick-drum miking.

KICK DRUM

Place a blanket or folded towel inside the drum, pressing against the beater head to dampen the vibration and tighten the beat. The blanket shortens the decay portion of the kick-drum envelope. To emphasize the attack, use a wood or plastic beater—not felt—and tune the drum low.

A popular mic for a kick drum is a large-diameter, cardioid dynamic type with an extended low-frequency response. Some mics are designed specifically for the kick drum, such as the AKG D112, Audio-Technica AT AE2500, Electro-Voice N/D868, and Shure Beta 52 A.

For starters, place the kick mic inside on a boom, a few inches from where the beater hits **(Figure 8.8)**. Mic placement close to the beater picks up a hard beater sound; off-center placement picks up more skin tone, and farther away picks up a boomier shell sound.

How should the recorded kick drum sound? Well, they don't call it kick drum for nothing. THUNK! You should hear a powerful low-end thump plus an attack transient.

CYMBALS

To capture all the crisp "ping" of the cymbals, a good mic choice is a cardioid condenser with an extended high-frequency response, flat or rising at high frequencies. Place the overhead mics about 2 to 3 feet above the cymbal edges; closer miking picks up a low-frequency ring. The cymbal edges radiate the most highs. Place the cymbal mics to pick up all the cymbals equally. If your recording will be heard in mono, or for sharper imaging, you might want to mount the mic grilles together and angle the mics apart **(Figure 8.4)**. This results in a narrow stereo spread. Another option is a stereo mic overhead. For wide stereo and sharp imaging, try a near-coincident pair aimed at the high-hat and floor tom.

Recorded cymbals should sound crisp and smooth, not muffled or harsh.

ROOM MICS

Besides the close-up drum mics, you might want to use a distant pair of room mics when you record drum overdubs. Place the mics about 10 or 20 feet from the set to pick up room reverb. When mixed with the close-up mics, the room mics give an open, airy sound to the drums. Popular room mics are omni condensers or boundary mics taped to the control-room window. You might compress the room mics for special effect. If you don't have enough tracks for room microphones, try raising the overhead mics.

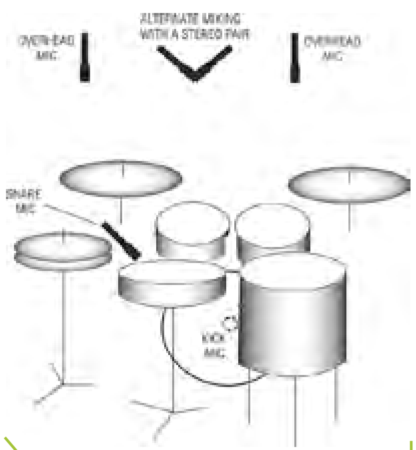


FIGURE 8.9
Miking a drum set with four mics.

BOUNDARY MIC TECHNIQUES

Boundary mics let you pick up the set in unusual ways. You can strap one on the drummer's chest to pick up the set as the drummer hears it. Tape them to hard-surfaced goboes surrounding the drummer. Put them on the floor under the toms and near the kick drum, or hang a pair over the cymbals. Try a supercardioid boundary mic in the kick drum.

RECORDING WITH TWO TO FOUR MICS

Sometimes you can mike the set simply. Place a single large-diaphragm cardioid condensor mic, two mics, or a stereo mic overhead and put another mic in the kick. If necessary, add a snare-drum mic **(Figure 8.9)**. This method works well for acoustic jazz, and often for rock. If you want the toms to sound fuller, boost the lows with a narrow Q around 80Hz in the overhead mics. Try reversing the polarity of the kick mic and see which position sounds best.

FIGURE 8.10
Miking a drum set with a stereo pair of mics.

As an alternative try two SDC mics on a small kit **(Figure 8.10)**. Place one mic over the snare about 8 inches up, 3 inches in from the rim, and midway between the hi-hat, rack tom, and snare drum. Place the other mic about 14 inches above the center of the floor tom. Set the cymbals low. You might boost the lows around 60 Hz to add fullness to the toms and cut around 200 Hz to make the snare less tubby **(Figure 8.11)**. Use a kick-drum mic too.

Audio clip 24 at www.taylorandfrancis.com/cw/bartlett-9780240821535 demonstrates several methods of miking a drum set.

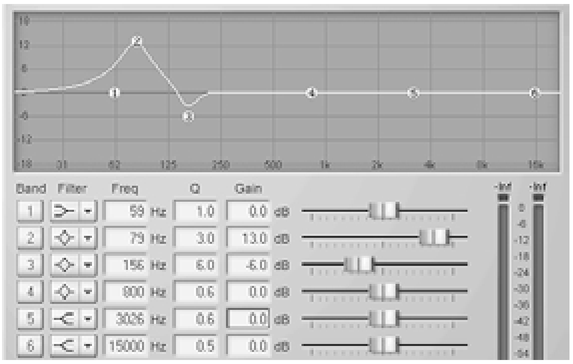


FIGURE 8.11
Possible EQ used on the overhead mics.

DRUM RECORDING TIPS

After you set up all the mics, ask the drummer to play. Listen for rattles and leakage by soloing each microphone. Try not to spend much time getting a sound; otherwise you waste the other musicians' time and wear out the drummer.

To keep the drum sound tight during mixdown, mute or delete drum tracks that are not in use in a particular tune, use a noise gate on the kick and toms, or overdub the drums. One dated effect for the snare drum is gated reverb. It's a short splash of brightsounding reverberation, which is rapidly cut off by a noise gate or expander. Many effects units have a gated-reverb program.

Another trick is recording "hot." Using an analog multitrack (or its plug-in equivalent), record the drums at a high level so they distort just a little. It's also common to compress the kick and snare.

A drummer might use drum pads, or drum triggers, fed into a sound module. Record directly off the module. You might want to mike the cymbals anyway for best sound. If you're recording a drum machine and it sounds too mechanical, add some real drums. The machine can play a steady background while the drummer plays fills.

When miking drums on stage for PA, you don't need a forest of unsightly mic stands and booms. Instead, you can use short mic holders that clip onto drum rims and cymbal stands, or use mini condenser mics.

In a typical rock mix, the drums either are the loudest element, or are slightly quieter than the lead vocal. The kick drum is almost as loud as the snare. If you don't want a wimpy mix, keep those drums up front!

Try these EQ settings to enhance the recorded sound of the drums:

- Snare: Fat at 200 Hz, crack at 5 kHz, sizzle at 10 or 12 kHz. Some snare drums ring a lot at one note. To fix it, set up an equalizer on the snare-drum track. Set a narrow, high-Q boost around 500 Hz and sweep it in frequency until you amplify the frequency that is ringing. Apply a narrow, high-Q cut there to remove the ringing.
- Toms: Cut around 600 to 800 Hz to reduce the papery sound. Then if necessary, boost around 100-200 Hz for more fullness on rack toms or 80 to 100 Hz on floor toms. Boost around 5 kHz for more attack.
- Cymbals: Sizzle at 10 kHz or higher. If you are close-miking the toms, you might roll off the lows in the cymbal mics below 500 Hz to reduce low-frequency leakage.
- Kick drum: To remove the "cardboard" sound, cut at 300 to 600 Hz. Then if necessary, boost at 3 to 5 kHz for more click. Don't overdo the high-frequency boost; usually you don't want too much "point" on the kick sound. Boost 60 to 80 Hz if the kick sounds thin. Filter out highs above 9 kHz to reduce leakage from cymbals.

Try these tricks to come up with unusual drum sounds:

- Record with a cheap dynamic or crystal mic, maybe in a can.
- Run the drums through extreme processing: compression, gating, distortion, pitch shifting, tremolo, and so on.
- Substitute other objects for drums, cymbals, drumsticks, and brushes.
- Move a mic around a cymbal or drumhead while recording it.
- Put the drums in a reverberant room or hallway.

Instead of recording an acoustic drum set, you might use an electronic drum set or CDs of drum samples. Copy the samples into a sampler or sampling software, then trigger them with a MIDI sequencer or MIDI controller with drum pads.

What if the drums sound too "live," "muddy," or reverberant? That means the drums suffer from too many early reflections because the drums are in a small, hard-surfaced room. Here are some solutions:

Add acoustical damping to the ceiling and walls. You can use acoustic foam, ATS Acoustics fiberglass panels, or Owens-Corning 703 pressed fiberglass insulation panels covered in muslin.

During mixdown, gate the snare and toms to remove early reflections. Here's how:

1. Solo the snare track.
2. Insert a gate plug-in in the snare track.
3. Solo the snare or tom track.
4. Set lookahead to 10 msec.
5. Gradually turn up the gate's input-level slider until the gate cuts off the reverb between snare or tom hits, but does not cut off the hits themselves.
6. Set the gate's hold time to 0-to-100 msec (experiment to see what sounds good).

There still will be some early snare reflections in the cymbal tracks. To reduce the snare sound in the cymbal tracks, roll off everything below about 1 kHz in the cymbal tracks. Here's how:

1. Solo the cymbal track(s).
2. Insert an equalizer plug-in in the cymbal tracks.
3. Solo the cymbal tracks.
4. Set band 1 to highpass.
5. Set band 1 Q to 1.7.
6. Set band 1 frequency to 1000 Hz. The cymbal tracks should sound thinner, with not as much snare leakage in them.

Now when you listen to the whole mix, the snare and tom sounds will come mostly from their tracks, and not from the cymbal tracks. It should sound a lot tighter and more professional.

Another way to tighten the sound is to signal-align the drum tracks. That way, the cymbal tracks and room-mic tracks play the drum hits at exactly the same time as the close-mic tracks. The result is a more focused sound. Zoom in on the track waveforms. Slide the distant-mic tracks to the left (earlier in time), or slide the other tracks right (later in time) until the waveforms align. It also helps to use a narrow stereo reverb on the snare rather than a wide, ambient reverb.

PERCUSSION

Let's move on to percussion, such as the cowbell, triangle, tambourine, or bell tree. A good mic for metal percussion is a condenser type because it has sharp transient response. Mike at least 1 foot away so the mic doesn't distort.

You can pick up congas, bongos, and timbales with a single mic between the pair, a few inches over the top rim, aimed at the heads. Or put a mic on each drum. It often helps to mike these drums top and bottom, with the bottom mic in opposite polarity. A cardioid dynamic with a presence peak gives a full sound with a clear attack.

For xylophones and vibraphones, place two cardioid mics 1½ feet above the instrument, aiming down. Cross the mics 135 degrees apart or place them about 2 feet apart. You'll get a balanced pickup of the whole instrument.

ACOUSTIC GUITAR

The acoustic guitar has a delicate timbre that you can capture through careful mic selection and placement. First prepare the acoustic guitar for recording. To reduce finger squeaks, try commercial string lubricant, a household cleaner/waxer, talcum powder on fingers, or smooth-wound strings. Ask the guitarist to play louder; this increases the "music-to-squeak" ratio!

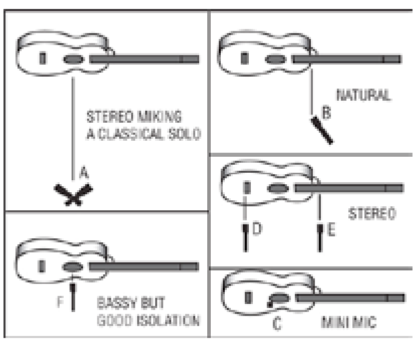


FIGURE 8.12
Some mic techniques for acoustic guitar.

Replace old strings with new ones a few days before the session. Experiment with different kinds of guitars, picks, and finger picking to get a sound that's right for the song.

For acoustic guitar, a popular mic is a pencil-type condenser with a smooth, extended frequency response from 80 Hz up. This kind of mic has a clear, detailed sound. You can hear each string being plucked in a strummed chord. Usually the sound picked up is as crisp as the real thing.

Now let's look at some mic positions. To record a classical guitar solo in a recital hall, mike about 3 to 6 feet away to pick up room reverb. Try a stereo pair **(Figure 8.12A)** such as XY, ORTF, MS, or a spaced pair. If you record a classical guitar solo in a dead studio, mike about 1.5 to 2 feet away and add artificial reverb.

When you record pop, folk, or rock music, try a spot about 6 to 12 inches from where the fingerboard joins the guitar body **(Figure 8.12B)**. That's a good starting point for capturing the acoustic guitar accurately. Still, you need to experiment and use your ears. Close to the bridge, the sound is woody and mellow.

In general, close miking gives more isolation, but tends to sound harsh and aggressive. Distant miking lets the instrument "breathe"; you hear a gentler, more open sound.

Another spot to try: tape a mini omni mic onto the body near the bottom of the sound hole, and roll off the excess bass. This spot gives good isolation **(Figure 8.12C)**.

The guitar will sound more real if you record in stereo. Try one mic near the bridge, and another near the 12th fret **(Figures 8.12D and E)**. Pan partway left and right. Another way to record stereo is with an XY pair of cardioid mics about 6 inches from the 12th or 16th fret, mixed with a 3-foot-spaced pair of omni mics about 3 feet away.

Is feedback or leakage a problem? Mike close to the sound hole **(Figure 8.12F)**. The tone there is very bassy, so turn down the low-frequency EQ on your mixer until the sound is natural. Also cut a few decibels around 3 kHz to reduce harshness.

You get the most isolation with a contact pickup. It attaches to the guitar, usually under the bridge. The sound of a pickup is something like an electric guitar. You can mix a mic with a pickup to add air and string noise to the sound of the pickup. That way, you get good isolation and good tone quality. You might record the acoustic guitar off its pickup while tracking to prevent leakage, then overdub the guitar later with a microphone for its better sound quality.

SINGER/GUITARIST

Normally you overdub the guitar and vocal separately. But if you have to record both at once, the vocal might sound filtered or hollow because of phase cancellations between the vocal mic and guitar mic. This can happen whenever two mics pick up the same source at approximately equal levels, at different distances, and both mixed to the same channel. Try one of these methods to solve the problem:

- Angle the vocal mic up and angle the guitar mic down to isolate the two sources. Follow the three-to-one rule.
- Mike the voice and guitar very close. Roll off the excess bass with your mixer's EQ.
- Use a pickup on the guitar instead of a mic.
- Place two bidirectional mics so the tops of their grilles touch. This gets rid of any delay between their signals. Aim the "dead" side of the vocal mic at the guitar; aim the dead side of the guitar mic at the mouth.

- Use just one mic, or a stereo mic, midway between the mouth and guitar about 1 foot out front. Adjust the balance between voice and guitar by changing the mic's height.
- Delay the vocal mic signal by about 1 msec. Then the signals of the two mics will be more in phase, preventing phase cancellations when they are mixed to the same channel. Some multitrack recorders have a track-delay feature for this purpose.
- Use Sound Radix Auto-Align ($149), an AU/RTAS plug-in. It automatically aligns the signals from two mics picking up the same source.

GRAND PIANO

This magnificent instrument is a challenge to record well. First have the piano tuned, and oil the pedals to reduce squeaks. You can prevent thumps by stuffing some foam or cloth under the pedal mechanism.

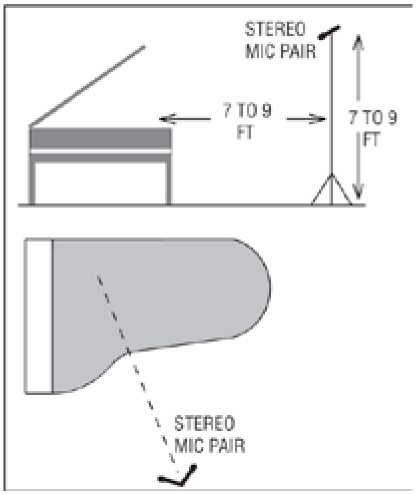


FIGURE 8.13
Suggested grand-piano miking for classical music (using cardioid mics).

For a classical-music solo, record in a reverberant room such as a recital hall or concert hall. Reverb is part of the sound. Set the piano lid on the long stick. Use condenser mics with a flat response. Place a stereo mic, or a stereo pair of cardioid mics, about 7 feet away and 7 feet high, up to 9 feet away and 9 feet high (Figure 8.13). Move the mics closer to reduce reverb, farther to increase it. When using a pair of omni mics, place them 1.3 to 2 feet apart, 3 to 6.5 feet from the piano, and 4 to 5 feet high (Figure 8.14). You might need to mix in a pair of hall mics: try cardioids aiming away from the piano about 25 feet away.

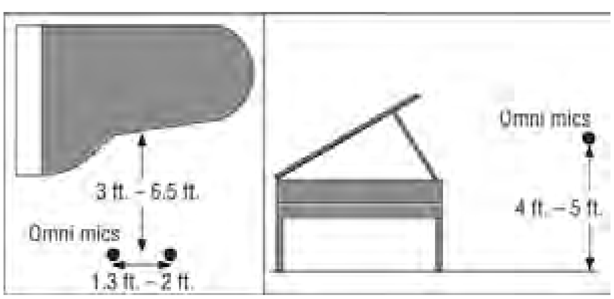


FIGURE 8.14
Suggested grand-piano miking for classical music (using omnidirectional mics).

When recording a piano concerto, give the piano a spot mic about 1 to 3 feet away. Put the mic in a shock mount.

Pop music demands close miking. Close mics pick up less room acoustics and leakage, and give a clear sound that cuts through the mix. Try not to mike the strings closer than 8 inches, or else you'll emphasize the strings closest to the mics. You want equal coverage of all the notes the pianist plays.

One popular method uses two spaced mics inside the piano. Use omni or cardioid condensers, ideally in shock mounts. Put the lid on the long stick. If you can, remove the lid to reduce boominess. Center one mic over the treble strings and one over the bass strings. Typically, both mics are 8 to 12 inches over the strings and 8 inches horizontally from the hammers (**Figure 8.15A**, bass and treble mics). Aim the mics straight down or angle them to aim at the hammers. Pan the mics partly left and right for stereo.

One alternative is to put the treble mic near the hammers, and put the bass mic about 2 feet toward the tail **(Figure 8.15B)**. Another method uses two ear-spaced omni condensers or an ORTF pair about 12 to 18 inches above the strings.

Audio clip 25 at www.taylorandfrancis.com/cw/bartlett-9780240821535 demonstrates some mic techniques for grand piano.

The spaced mics might have phase cancellations when mixed to mono, so you might want to try coincident miking (**Figure 8.15A** , stereo pair). Boom-mount a stereo mic, or an XY pair of cardioids crossed at 120 degrees. Miking close to the hammers sounds percussive; toward the tail has more tone.

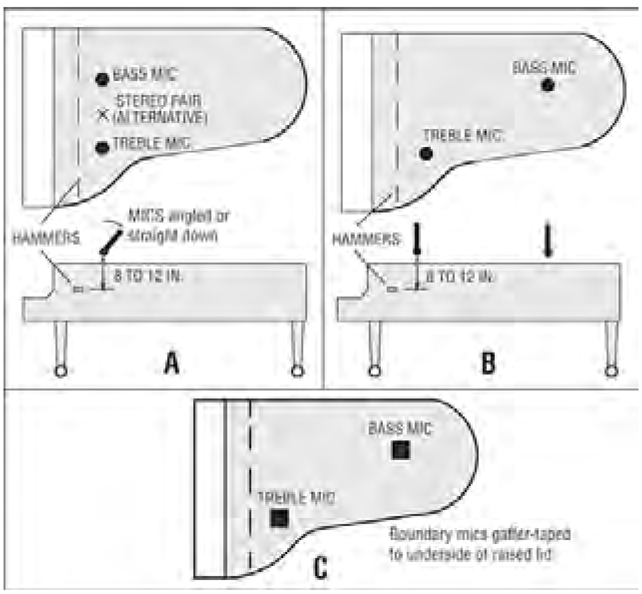


FIGURE 8.15
Suggested grand-piano miking for popular music.

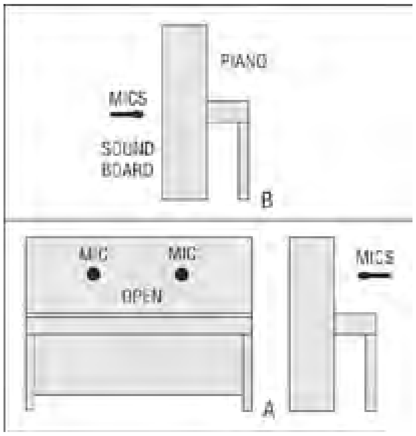


FIGURE 8.16
Some mic techniques for upright piano.

For more clarity and attack, boost EQ around 10 kHz or use a mic with a rising high-frequency response.

Boundary mics work well, too. If you want to pick up the piano in mono, tape a boundary mic to the underside of the raised lid, in the center of the strings, near the hammers. Use two for stereo over the bass and treble strings.

- Remove the top lid and upper panel. Put a stereo pair of mics about 1 foot in front and 1 foot over the top. If the piano is against a wall, angle the piano about 17 degrees from the wall to reduce tubby resonances.
- Aim the soundboard into the room. Mike the bass and treble sides of the soundboard a few inches away. In this spot, the mics pick up fewer pedal thumps and other noises. Try cardioid dynamic mics with a presence peak.

ACOUSTIC BASS

The acoustic bass (string bass, double bass, upright bass) puts out frequencies as low as 41 Hz, so use a mic with an extended low-frequency response such as a large-diaphragm condenser mic or ribbon mic. As always, closer miking improves isolation, while distant miking tends to sound more natural but can pick up too much room sound. Try these techniques **(Figure 8.17)**:

- 4 to 8 inches in front of the bridge, a few inches above the bridge.
- 4 to 6 inches under the bridge, a few inches from the strings. This mic will pick up a deep sound with good definition. You might mix in a second mic near the plucking fingers for clarity, a few inches from the side of the fingerboard.
- Mix a pickup with a mic, or use a pickup alone and use EQ to improve the sound.

With EQ you can emphasize slap at 5 kHz, or boxiness at 1.25 kHz. Try compressing the body mic 2:1 and the slap mic 6:1.

Here are some methods that isolate the bass and let the player move around. They work well for PA:

- Wrap a mini omni condenser mic in foam rubber (or in a foam windscreen) and mount it in the bridge aiming up **(Figure 8.17)**.
- Tape a mini omni mic to the bridge, or wedge it into a slot in the bridge.

FIGURE 8.17
Some mic techniques for the acoustic bass.

- Wrap a regular cardioid mic in foam padding (except the front grille) and squeeze it behind the bridge **(Figure 8.17)** or tailpiece.
- For best isolation, try a direct feed from a pickup. This method adds clarity and deep bass, but probably will need some EQ. You might mix the pickup with a microphone.

BANJO

Try a flat-response condenser mic about 1 foot away from the center **(Figure 8.18)**. Or try a cardioid dynamic mic 6 inches from the lower-right area or 6 inches from the lower rim, aiming at the bridge. If you need more isolation, mike closer and roll off some bass. The banjo sounds pleasantly mellow when miked toward the edge of the head, near the resonator holes (if the banjo has them). Cloth stuffed inside will tighten the sound and will reduce feedback in PA situations.

For the most isolation, tape a mini omni condenser mic to the head about halfway between the bottom edge and the bridge. You can wedge a pickup between the strings below the bridge and the banjo head. Put the pickup flat against the head surface.

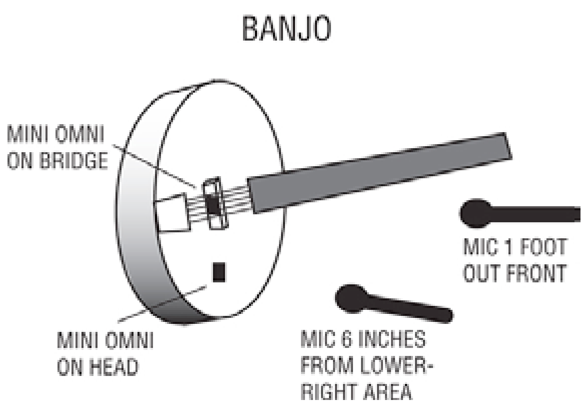


FIGURE 8.18
Four methods for miking a banjo.

MANDOLIN, DOBRO, BOUZOUKI, AND LAP DULCIMER

Mike these about 6 to 8 inches away from the hole with a small-diaphragm condenser mic. Some engineers also mix in a mic near the neck or dobro resonator. If you need more lows and more isolation, tape a mini omni condenser mic near a hole and tweak EQ for the best sound.

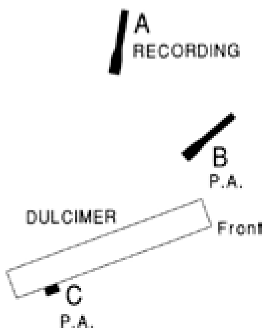
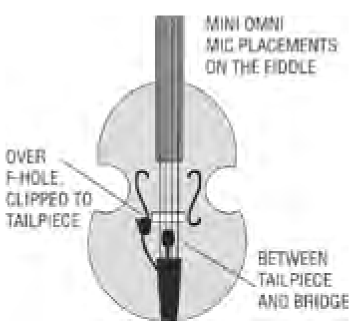


FIGURE 8.19
Some mic techniques for hammered dulcimer.

FIGURE 8.20
Three fiddle-miking methods.



HAMMERED DULCIMER

Place a flat-response condenser mic about 2 feet over the center of the soundboard (Figure 8.19A). On stage, place a cardioid dynamic or condenser 6 to 12 inches over the middle of the top end (Figure 8.19B). For the best gain-before-feedback in a PA system, mix in a mini omni condenser mic (or a cardioid with bass rolloff) very near the sound hole (Figure 8.19C).

FIDDLE (VIOLIN)

Listen to the fiddle itself to make sure it sounds good. Correct any instrument problems before miking.

First try a flat-response condenser mic (omni or cardioid) about 2 feet over the bridge. This distant miking gives an airy, silky sound. Close miking (about 6 to 12 inches, see Figure 8.20) sounds more aggressive, which is desirable in old-time or bluegrass music. Aim the mic toward the f-holes for warmth or toward the fingerboard for clarity. A fiddle that sounds too bright and scratchy can be tamed using a ribbon mic, or by miking it from the side or even underneath. If the ceiling over the fiddle is low, nail a square yard of acoustic foam up there to prevent reflections.

If you have to mike close—say, for a singing fiddler—aim the mic at the player's chin from 6 to 12 inches above the end of the fingerboard. The mic will pick up both the singer and the fiddle.

If you need more isolation, try a mini omni mic or a Bartlett Fiddle Mic. Wrap its cable in foam rubber (or a windscreen) 1½ inches from the capsule. Wedge the foam under the tailpiece, and position the mic capsule halfway between the tailpiece and bridge, a half inch over the body (Figure 8.21). If necessary, cut a little at 10 kHz to reduce harshness and boost around 200 Hz for warmth. Another way to get a warm sound is to mount the mic near an f-hole.

A good spot for a pickup is on the left side of the top (player's view), on the player's side of the bridge.

To record a classical violin solo, try a stereo mic (or a stereo pair) 5 to 15 feet away in a reverberant room.

FIGURE 8.21
Two ways to close-mike a fiddle for isolation.

STRING SECTION

Place the strings in a large, live room and mike them at distance to pick up a natural acoustic sound. A common mic choice is a condenser with a fl at response. First try a stereo mic or stereo pair of mics about 4 to 20 feet behind the conductor, raised about 13-15 feet.

If the room is noisy or too dead, or the balance is poor, you'll need to mike close and add digital reverb. Try one mic on every two to four violins, 6 feet off the floor, aiming down, with the same setup for the violas. Mike the cello about 1 to 2 feet from the bridge, to the right side between the bridge and f-hole. When you mix the strings to stereo, pan them evenly between the monitor speakers. Spread them left, center, and right to make a "curtain of sound." If you can spare only one track for the strings, use a stereoizer effect during mixdown.

STRING QUARTET

Record a quartet in stereo using a stereo mic or a pair of mics. Place them about 6 to 10 feet away to capture the room ambience. The monitored instruments should not spread all the way between speakers. If you want to narrow the stereo stage, angle or space the mics closer together.

BLUEGRASS BAND AND OLD-TIME STRING BAND

Suppose you're recording a group that has a good acoustic balance. Try a stereo mic or stereo pair of mics about 2 to 3 feet away and 5 feet high (lower if the group is seated). Move the players toward or away from the mics to adjust their balance. You'll have more control if you mike all the instruments up close and mix them. This also gives a more "commercial" sound. Arrange the players in a circle facing each other. The production style aims for a natural timbre on all the instruments, either with no effects or with slight reverb.

HARP

Use a condenser mic with a flat response. If the harp is playing with an orchestra, mike the harp about 18 inches from the front of the soundboard, or 18 inches from the player's left hand. With a Celtic group, try a mic 1/3 of the way up the sound board about 6 to 12 inches away.

Tape a mini omni condenser mic to the soundboard if you need isolation. A mic on the inside of the soundboard has more isolation; a mic on the outside sounds more natural. Also try a cardioid condenser wrapped in foam, stuck into the center hole from the rear.

HORNS

FIGURE 8.22
Miking for trumpet tone control.

"Horns" in studio parlance refers to the brass instruments: trumpets, cornets, trombones, baritones, french horns, and tubas.

All the brass radiate strong highs straight out from the bell, but do not project them to the sides. A mic close to and in front of the bell picks up a bright, edgy tone. To mellow out the tone, mike the bell off-axis with a flat-response mic **(Figure 8.22)**. The sound on-axis to the bell has a lot of spiky high harmonics that can overload a condenser mic, mixer input, or analog tape. That's another reason to mike off-axis.

Mike the trumpet with a dynamic or ribbon mic to take the edge off the sound. Use a condenser mic if you want a lot of sizzle. Mike about 1 foot away for a tight sound; mike several feet away for a fuller, more dramatic sound.

You can pick up two or more horns with one microphone. Several players can be grouped around a single omni mic, or around a stereo pair of mics. The musicians can play to a pair of boundary mics taped on the control-room window or on a large panel.

Record a classical brass quartet in a reverberant room. Use a stereo mic, or a stereo pair of mics, about 6 to 12 feet away.

SAXOPHONE

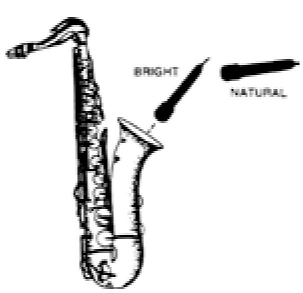


FIGURE 8.23
Two ways to mike a saxophone.

A sax miked very near the bell sounds bright, breathy, and rather hard **(Figure 8.23)**. Mike it there for best isolation. To get a warm, natural sound, mike the sax about 1½ feet away, halfway down the wind column **(Figure 8.23)**. Don't mike too close, or else the level varies when the player moves. A compromise position for a close-up mic is just above the bell, aiming at the holes. You can group a sax section around one mic.

Figure 8.24 shows a typical miking setup for big-band jazz. It uses the techniques already described for the drums, bass, piano, electric guitar, trumpet, and sax.

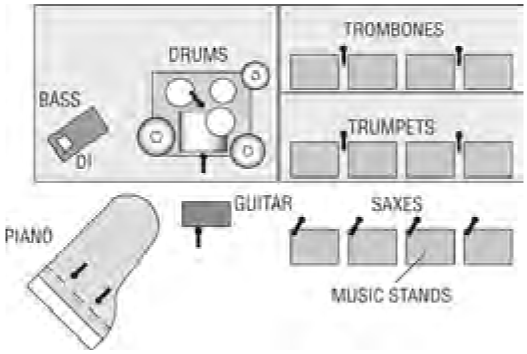


FIGURE 8.24
Typical miking setup for big-band jazz.

FIGURE 8.25
Miking a clarinet from the side.

WOODWINDS

With woodwinds, most of the sound radiates not from the bell, but from the holes. So aim a flat-response mic at the holes about 1 foot away **(Figure 8.25)**.

When miking a woodwind section within an orchestra, you need to reject nearby leakage from other instruments. To do that, try aiming a bidirectional mic down over the woodwind section. The side nulls of the mic cut down on leakage.

To pick up a flute in a pop-music group, try miking a few inches from the area between the mouthpiece and the first set of finger holes **(Figure 8.26)**. You may need a pop filter. If you want to reduce breath noise, roll off high frequencies or mike farther away. You also can attach a mini omni mic to the flute a few inches above the body, between the mouthpiece and finger holes.

For classical music solos, try a stereo pair 4 to 12 feet away.

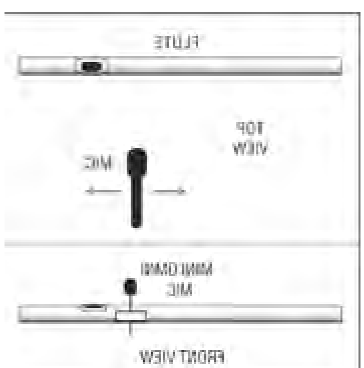


FIGURE 8.26
Two methods of miking a flute.

HARMONICA, ACCORDION, AND BAGPIPE

One way to mike a harmonica (harp) is to use a cardioid dynamic mic with a ball grille. Place the mic very close to the harmonica or have the player hold it. A condenser mic about 1 foot away gives a natural sound. To get a bluesy, dirty sound, use a "bullet"-type harmonica mic or play the harmonica through a miked guitar amp.

For an accordion, try a mic about 6 to 12 inches from the sound holes near the keyboard. Some accordions have sound holes on both sides, so you'll need two mics. Follow the three-to-one rule. The distance between mics should be at least three times the mic-to-source distance. One end of the accordion is in constant motion, so you might want to attach a mini omni mic to that end. A solo accordion or concertina could be miked with a stereo pair of flat-response cardioid condenser mics about 3 to 6 feet in front.

A bagpipe has two main sound sources: the chanter, which the musician plays with the fingers, and the drone pipes, which make a steady tone. Mike the chanter about a foot away from the side, and mike the drone pipes a foot from the end. Again, follow the three-to-one rule. You could also mike the bagpipe a few feet away with one mic.

LEAD VOCAL

The lead vocal is the most important part of a pop song, so it's critical to record it right. First set up a comfortable environment for the singer. Put down a rug, add some flowers or candles, dim the lights. Supply some water and let the singer stretch and warm up. Set up a good cue mix with effects to help the singer get into the mood of the song.

You might want to turn off the reverb in the singer's headphones; this makes it easier to hear pitch. If the vocalist is singing flat, reduce their headphone volume, and vice versa.

With any vocal recording, there are some problems to overcome, but we can deal with them. Among these are proximity effect, breath pops, wide dynamic range, sibilance, and sound reflections from the music stand. Let's look at these in detail.

MIKING DISTANCE

When you sing or talk close to most directional mics, the microphone boosts the bass in your voice. This is called the proximity effect. We've come to accept this bassy sound as normal in a PA system, but the effect just sounds boomy in a recording.

FIGURE 8.27
Typical miking technique for a lead vocal.

To prevent boomy bass, mike the singer at a distance, about 8 inches away (Figure 8.27). A popular mic choice is a flat-response condenser mic with a large diaphragm (1 inch or larger diameter). As always, you can use any mic that sounds good to you. If the mic has a bass rolloff switch, set it to "flat." Singers should maintain their distance to the mic. I ask the singer to spread the fingers, touch lips with the thumb, and touch the mic with the pinky. The hand forms a spacer for keeping a constant distance.

If the singer's voice is too harsh or edgy, try miking partly to the side of the mouth, aiming at the mouth. Or try a ribbon mic, or a multiband compressor set to compress from 2000 Hz and up.

Some singers feel more comfortable singing into a handheld mic. You can give them a handheld mic but also mike them several inches away with a good condenser microphone. Record both mics on separate tracks and choose whatever sounds the best. If you must record the singer and the band at the same time—as in a concert—you'll have to mike close to avoid picking up the instruments with the vocal mic. Try a cardioid or supercardioid mic with a bass rolloff and a foam pop filter. The sound will be bassy because of proximity effect, so roll off the excess lows at your mixer. For starters, try –6 dB at 100 Hz to 200 Hz. Some mics have a bass filter switch for this purpose. Aim the mic partly toward the singer's nose to prevent a nasal or closed-nose effect. This close-up method works well if you want an intimate, breathy sound.

When recording a classical-music singer who is accompanied by an orchestra, place the mic about 1 to 2 feet away. If the singer is a soloist (maybe accompanied by piano), use a stereo pair of boundary mics about 8 to 15 feet away to pick up room reverb.

BREATH POPS

When you sing a word with "p" or "t" sounds, a turbulent puff of air shoots out of the mouth. The puff hits the mic and makes a thump or small explosion called a pop. To reduce it, put a foam-plastic pop filter on the mic. Some mics have a ball grille screen to cut pops, but foam works better. The pop filter should be made of special open-cell foam to pass high frequencies. For best pop rejection, allow a little air space between the foam and the front of the mic grille.

Foam pop filters reduce the highs a little. So they should be left off instrument mics, except for outdoor recording or dust protection. Pop filters do not reduce breathing sounds or lip noises. To get rid of these problems, mike farther away or roll off some highs.

The most effective pop filter is a hoop with a nylon stocking stretched over it **(Figure 8.27)** or a disk of perforated metal. You can buy those, or make one with a coat hanger and a crochet hoop. Place the filter a few inches from the mic.

Another way to get rid of pop is to put the mic at forehead height, aiming at the mouth. This way the puffs of air shoot under the mic and miss it. Make sure the vocalist sings straight ahead, not up at the mic, or the mic will pop.

WIDE DYNAMIC RANGE

During a song, vocalists often sing too loud or too soft. They blast the listener or get buried in the mix. That is, many singers have a wider dynamic range than their instrumental backup. To even out these extreme level variations, ask the singer to use proper mic technique. Back away from the mic on loud notes; come in closer for soft ones. Or you can ride gain on singers: gently turn them down as they get louder, and vice versa. Usually this is done during mixdown with automation or volume envelopes.

Another solution is to pass the vocal signal through a compressor, which acts like an automatic volume control. Plug the compressor into the vocal channel's insert jacks or insert a compressor plug-in in the vocal track during mixdown. A typical compressor setting for vocals is a 2:1 ratio, –10 dB threshold, and about 3 to 6 dB of gain reduction. Of course, you should use whatever settings are needed for the particular singer.

Audio clip 26 at www.taylorandfrancis.com/cw/bartlett-9780240821535 demonstrates vocal compression, as well as breath pops and the effects of miking distance on recorded vocals.

If the singer moves toward and away from the mic while singing, the average level will go up and down. Try to mike the singer at least 8 inches away, so that small movements of the singer won't affect the level.

An alternative to compressing the vocal is Waves Vocal Rider software (www.waves.com). It automatically adjusts the vocal volume in real time and provides a more natural sound than compression.

If you must mike close to prevent leakage or feedback, ask the vocalist to sing with lips touching the foam windscreen to keep the same distance to the mic. Turn down the excess bass using your mixer's low-frequency EQ (typically –6 dB at 100 Hz to 200 Hz).

SIBILANCE

Sibilance is the emphasis of "s" or "sh" sounds, which are strongest around 3 to 10 kHz. They help intelligibility. In fact, many producers like sizzly "s" sounds, which add a bright splash to the vocal reverb. But the sibilance should not be piercing or strident.

If you want to reduce sibilance, use a mic with a flat response—rather than one with a presence peak—or cut the highs little around 8 kHz on your mixer. Better yet, use a de-esser signal processor or plug-in, which cuts the highs only when the singer makes sibilant sounds. You can set a multiband compressor to act as a de-esser.

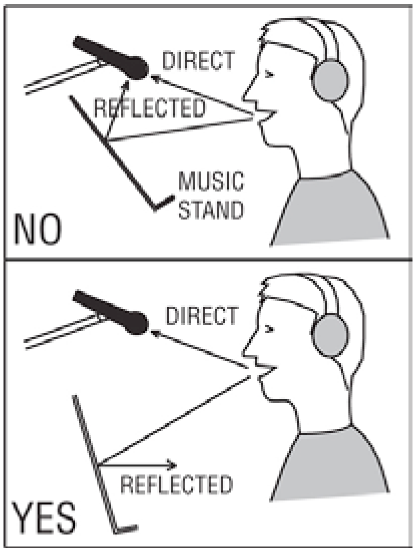


FIGURE 8.28
Preventing reflections from a music stand.

REFLECTIONS FROM THE MUSIC STAND AND CEILING

Suppose that a lyric sheet or music stand is near the singer's mic. Some sound waves from the singer go directly into the mic. Other sound waves reflect off the lyric sheet or music stand into the mic (Figure 8.28, top). The delayed reflections will interfere with the direct sound, making a colored tone quality like mild flanging.

To prevent this, lower the music stand and tilt it almost vertically (Figure 8.28, bottom). This way, the sound reflections miss the mic.

If your studio has a low ceiling, the recorded vocal might have a colored tone quality due to phase cancellations from ceiling reflections. Try putting the mic lower and use a hoop-type pop filter. Also put a 3-foot square of acoustic foam or pressed-fiberglass panel on the ceiling over the singer and mic.

VOCAL EFFECTS

Some popular vocal effects are stereo reverb, echo, and doubling. You can record real room reverb by miking the singer at a distance in a hard-surfaced room. Slap echo provides a 1950s rock 'n' roll effect. Often a vocal is mixed dry, with no reverb. A little distortion might even be effective on some songs. You might try a vocal processor, which offers a variety of effects. Try different EQ or different effects on each section of a song.

For the best sound, put vocal plug-ins in this order: (1) de-ess, (2) compressor, (3) EQ.

Doubling a vocal gives a fuller sound than a single vocal track. Overdub a second take of the vocal on an empty track, in sync with the original take. During mixdown, mix the second vocal take with the original, at a slightly lower level than the original. You can double a vocal track by running it through a digital delay set at 15 to 35 msec, or through a pitch shifter that is detuned 10 to 15 cents.

BACKGROUND VOCALS

When you overdub background vocals (harmony vocals), you can group two or three singers in front of a mic. The farther they are from the mic, the more distant they will sound in the recording. Pan the singers left and right for a stereo effect. Because massed harmonies can sound bassy, roll off some lows in the background vocals.

If you want independent control of each background singer, give each one a close-up mic and record them with separate mixer channels or separate tracks.

Barbershop or gospel quartets with a good natural blend can be recorded with a stereo mic or stereo pair of mics about 2 to 4 feet away. If their balance is poor, close-mike each singer about 8 inches away, and balance them with your mixer. This also gives a more "commercial" sound. If you close-mike, spread the singers at least 2 feet apart in a circle to prevent phase cancellations.

SPOKEN WORD

The tips given earlier for a lead vocalist also apply to recording the spoken word (including podcasts, audio books, etc.). Be sure to keep the miking distance constant and use a hoop-type pop filter. To prevent sound reflections into the mic, put the script on a padded music stand that is angled almost vertically, and put the mic in the plane of the stand near the top edge. Fold up a corner of each script page to form a handle for turning pages silently.

The engineer and announcer should both have the same script. Mark the beginning of each misread sentence. The announcer should reread each misread sentence from the beginning to make editing easier.

CHOIR AND ORCHESTRA

Figure 8.29 shows three ways of miking a choir. If the mics will also be used for PA, or if the venue is noisy or sounds bad, try miking close, pan the mic as desired, and add artificial reverb **(Figure 8.29A)**. Otherwise, try a near-coincident pair of cardioid mics **(Figure 8.29B)** or a pair of omni mics spaced about 2 feet apart **(Figure 8.29C)**. Adjust the mic-to-choir distance until you hear the desired amount of hall acoustics in your monitors.

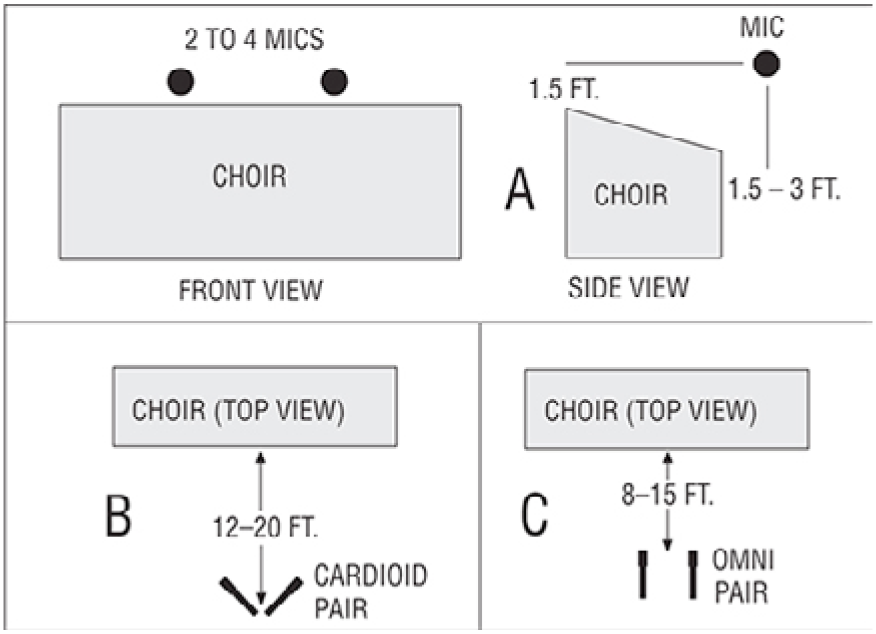


FIGURE 8.29
Choir miking suggestions.
(a) Close-up panned mics.
(b) Near-coincident stereo pair.
(c) Spaced stereo pair.

ADDITIONAL EXAMPLES FROM SOS

CASE STUDY ■ BACK TO FRONT

Always check the spec to see which part of the mic you should sing into. Most large diaphragm studio capacitor mics are so-called 'side-fire' designs, which means you sing into the side of them, whereas in contrast, most stage vocal mics are 'end-fire' mics where you sing directly into the end. (These side- and end-fire terms are nonsensical, of course: nothing comes out of them; sound only goes in, but they are widely used, nonetheless!). Usually the 'hot', or active side of a side-address microphone is designated by the manufacturer's logo, or in the case of Røde, a gold spot – we mention the latter only because on one Studio SOS visit we noticed that the user had a 'side-fire' mic set up in an 'end-fire' configuration, with a pop shield over the end of it.

When we asked why, he said it was because he'd, "tried using the side, but it sounded very 'roomy'… and that singing into the end seemed a lot better". Of course, what had happened was that he'd initially tried the back of the mic which, being a cardioid-pattern model, would indeed sound dull and very roomy indeed, but then rather than check the other side, he'd assumed that it was an 'end-fire' mic, and rigged it accordingly. Singing into the top of the mic meant he was actually using it 90 degrees off-axis, where the output level is only about 6dB lower than the on-axis level, and only slightly duller than it should have been! That was clearly a massive improvement on using it directly from the back in the cardioid null, but still not as good as it was capable of!

If in doubt about the polar pattern or the orientation, listen to the output of the mic while moving around it in a complete circle talking or singing as you go. That will quickly reveal the polar pattern and intended main axis!

CASE STUDY ■ ROCK BAND IN THE GARDEN SHED!

The following example came about as a request for help from a band recording rock music in a garden shed just about large enough to accommodate all of the band members at the same time! Noise leakage wasn't a problem, as they were well away from any neighbours and they were already using power soaks for recording two large, closed-back guitar stacks and a bass amp, alongside a miked drum kit. Their problem was that, even with the power soaks, too much of the guitar sound was getting into the drum overhead mics, making their tracks difficult to mix.

We improved the spill situation by keeping the power soaks on the guitar amps but turned their speaker cabinets around to face directly into the mattress-lined walls of the shed, leaving just enough space in front of each cabinet to place a microphone. In this way most of the high frequency amp sound was absorbed by the mattresses instead of bouncing around the room and finding its way into the drum overheads. We also took a DI feed from the preamp output of the bass amp as that can sometimes be used in addition to, or instead of, the miked feed. The band was still able to monitor their performance as loudly as they liked via headphones, but when we did our first take the amount of guitar spilling into the drum mics was too low to worry about whereas before, with the amps facing the centre of the room, the drum overheads were picking up more guitar than drums! The lesson here is that it may sometimes take a combination of two or more techniques and tricks to solve a problem.

CASE STUDY ■ PLAY IT AGAIN, SAM

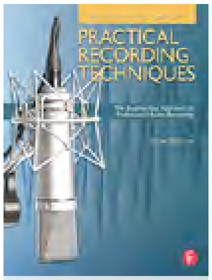
One mix we tried to help out on suffered because the original guitar had been DI'd with too much input gain, resulting in a significant amount of digital clipping. The player had then used the amp-modelling software that came with his DAW to try to shape this into a suitable overdriven guitar sound, but the result was far too unfocussed and messy to work properly in the track. After a few futile minutes spent adding EQ and trying different amp models, we persuaded him to play the part again, this time using his small valve combo. We set up a back-electret capacitor mic a few inches from the speaker grille and re-recorded the part using slightly less distortion than was used on the original track. The new part sat perfectly in the mix and the player was amazed how much difference miking his little amp had made.

This again demonstrates that too often we are tempted to take the most convenient approach to a recording when that approach might compromise the end result to a very significant degree. It also underlines the reality that it is often much quicker to play or sing a substandard part again than to spend ages trying to salvage something with plug-ins and still not getting it to sound good!

SUMMARY

We can sum up mic placement like this: If leakage or feedback is a problem, place the mic near the loudest part of the instrument, and add EQ to get a natural sound. Otherwise, place the mic in various spots until you fi nd a position that sounds good over your monitors. There is no single "correct" mic technique for any instrument. Just place the mic where you hear the desired tonal balance and amount of room reverb.

Try the techniques described here as a starting point, then explore your own ideas. Trust your ears! If you capture the power and excitement of electric guitars and drums, and the beautiful timbre of acoustic instruments and vocals, you've made a successful recording.



The following is excerpted from *Practical Recording Techniques* by Bruce Bartlett and Jenny Bartlett.

SETTING UP THE STUDIO

About an hour before the session starts, clean up the studio to promote a professional atmosphere. Lay down rugs and place AC power boxes according to your layout chart.

Now position the baffles (if any). Put out chairs, stools, and music stands according to the layout. Run a headphone extension cable from each artist's location to the headphone junction box in the studio.

Place mic stands approximately where they will be used. Wrap one end of a microphone cable around each microphone-stand boom, leaving a few extra coils of cable near the mic-stand base to allow slack for moving. Run the rest of the cable back to the mic input panel or snake box. Plug each cable into the appropriate wall-panel or snake-box input according to your mic input list.

Some engineers prefer to run cables in reverse order, connecting to the input panel first and running the cable out to the microphone stand. That procedure leaves less of a confusing tangle at the input panel where connections might be changed.

Now set up the microphones. Check each mic to make sure its switches are in the desired positions. Put the mics in their stand adapters, connect the cables, and balance the weight of each boom against the microphone.

Finally, connect the musicians' headphones for cueing. Set up a spare cue line and microphone for last-minute changes.

SETTING UP THE CONTROL ROOM

Having prepared the studio, make sure the control room is ready for the session. Then turn up the monitor system. Carefully bring up each fader one at a time and listen to each microphone. You should hear normal studio noise. If you hear any problems such as dead or noisy microphones, hum, bad cables, or faulty power supplies, correct them before the session.

Verify the mic input list. Have an assistant scratch each mic grille with a fingernail and identify the instrument the microphone is intended to pick up. If you have no assistant, listen on headphones as you scratch the grilles.

Check all the cue headphones by playing a tone or music through them and listening while wiggling each cable. Make sure they play at a reasonable level, not too loud.

SESSION OVERVIEW

This is the typical sequence of events:

1. For efficiency, record the basic rhythm tracks for several songs on the first session.
2. Do the overdubs for all the songs in a dubbing session.
3. Mix all the tunes in a mixdown session.
4. Edit the tunes and master the album.

Some musicians prefer instead to record all the parts for one song at a time.

After the musicians arrive, allow them 1/2 hour to 1 hour free setup time for seating, tuning, and mic placement. Show them where to sit, and work out new seating arrangements if necessary to make them more comfortable.

Once the instruments are set up, listen to their live sound in the studio and do what you can to improve it. A dull-sounding guitar may need new strings, a noisy guitar amp may need new tubes, the drums may need tuning, and so on. Adjust the studio lighting for the desired mood.

RECORDING

Before you start recording, you might want to make connections to record the monitor mix. This recording is for the producer to take home to evaluate the performance.

Many DAW programs have session templates: a group of tracks with EQ, compression, and effects plug-ins already in place. You simply load in the template and you're ready to go. You can create your own session templates, such as "16 tracks with aux 1 and 2" or "drum set."

Some DAWs let you create a track template, which is a format for a single track. You might have a vocal track template with the input channel, reverb, EQ, and compression already set up. Each time you're about to record another vocal, import the vocal track template.

Set recording levels, then set cue mixes for the musicians' headphones. The monitor mix affects only what is heard, not what is recorded. To clarify the beat of the cue mix, you might create a mix of only a few instruments such as drums, bass, chords, and vocal.

When you're ready to record the tune, briefly play a metronome to the group at the desired tempo, or play a click track (an electronic metronome) through the cue system.

Or just let the drummer set the tempo with stick clicks. A click track is helpful if you are starting with an acoustic guitar and are adding drums and bass later.

Start recording. Note the recorder counter time. Hit the slate button (if any) and announce the name of the tune and the take number. Then the group leader or the drummer counts off the beat, and the group starts playing.

The producer listens to the musical performance while the engineer watches levels and listens for audio problems. As the song progresses, you may need to make small level adjustments.

The assistant engineer (if any) runs the multitrack recorder and keeps track of the takes on the track sheet, noting the name of the tune, the take number, and whether the take was complete **(Figure 15.3)**. Use a code to indicate whether the take was a false start, long false start, nearly completed, a "keeper," and so on.

While the song is in progress, don't use the solo function, because the abrupt monitoring change may disturb the producer. The producer should stop the performance if a major flub (mistake) occurs but should let the minor ones pass.

At the end of the song, the musicians should be silent for several seconds after the last note. Or, if the song ends in a fade-out, the musicians should continue playing for about 30 seconds so there is enough material for a fade-out during mixdown.

After the tune is done, you can either play it back or go on to a second take. If you connected your multitrack recorder to the insert jacks, use the faders to set a rough mix with EQ and effects. The musicians will catch their flubbed notes during playback; you just listen for audio quality.

Now record other takes or tunes. Pick the best takes. Punch in to correct errors on each best take, or fix errors with editing.

To protect your hearing and to prevent fatigue, try to limit tracking sessions to four hours or less. Take breaks to give your ears and body a rest.

RELATING TO THE MUSICIANS

During a session, the engineer needs not only technical skills, but also people skills. It's important to respect artistic personalities and to keep the creative energy flowing.

First, learn the names of the band members and refer to them by name during the session. Musicians are often nervous at the beginning of a session.

Having a sense of humor and exuding confidence helps to put the artists at ease. Talk about the band's music, origin, or instruments. Be sure that the artists are comfortable—adjust the layout and lighting, offer beverages and snacks. Tell new clients that mistakes are normal and are easy to fix. Ask band members, "Is the mix okay in your headphones? Is the volume all right? Does your recorded instrument sound okay?"

Don't badmouth other musicians, and respect the privacy of your clients. Each musician you record should feel confident that you're not telling others about their mistakes.

When you want to try something that could enhance a song, don't just do it. Ask first. Say, "What would you think about doubling the guitar in stereo?" or "I want your input on this."

During a playback, try not to point out the errors. The musicians will hear them and will try to do better on the next take. Don't say, "That sucked." Say "It's almost there— do you want to punch in a few spots?" Or say "That was good, but how about another take?" The goal is to make the best record you can, not to make clients feel bad about themselves. Give a musician time to practice during overdubs, too.

If the band members are getting tired or are having trouble getting through a particular song, you might say "Let's take a break" or "Let's try another song and come back to this one later."

Finally, if a piece of gear breaks or your software develops a glitch, try to work around it quietly without making it obvious. Musicians need an engineer who handles technical problems in a professional way.

OVERDUBBING

After recording the basic or rhythm tracks for all the tunes, add overdubs. A musician listens to recorded tracks over headphones and records a new part on an open track.

You might have the musician play in the control room while overdubbing. You can patch a synth or electric guitar into the console through a direct box, and feed the direct signal to a guitar amp in the studio via a cue line. Pick up the amp with a microphone, and record and monitor the mic signal.

Do any drum overdubs right after the rhythm session because the microphones are already set up, and the overdubbed sound will match the sound of the original drum track.

BREAKING DOWN

When the session is over, tear down the microphones, mic stands, and cables. Put the microphones back in their protective boxes and bags. Wind the mic cables onto a power-cable spool, connecting one cable to the next. Wipe off the cables with a damp rag if necessary. Some engineers hang each cable in big loops on each mic stand. Others wrap the cable "lasso style" with every other loop reversed. You learn this on the job.

Put the labeled recording and session documents in a labeled box or file folder. Normally the studio keeps the multitrack recording for possible future work unless the group wants to buy or rent it. Backup the multitrack recordings (audio files) and session files (track setups for each song) to another medium, such as an external hard drive, CD-R, or DVD-R. Glyph makes pro-quality external hard drives for audio and video storage.

If you're using a hardware mixing console, log the console and effects settings by writing them in the track sheet or reading them slowly into a portable recorder. At a future session you can play back the recording and reset the console the way it was for the original session. Automated consoles can store and recall the console's control adjustments. DAWs remember both the mixer settings and the plug-in settings.

A FEW WORDS ON STUDIO PSYCHOLOGY
from Carlos Lellis Ferreira

It is not uncommon for producers to find their artists acting in an unpredictable or unreasonable manner during sessions and it is important for the former to remember that the talent may feel under a lot of pressure while tracking. The recording process has an enormous importance in the life of a performer, who may or may not get to develop an artistic career depending on the success of their studio efforts. With this in mind, individuals in the control room must be fully committed and supportive of their artists, assuring them that the success of a recording session is equally important to all members of the production team (this may help distribute some of the pressure).

The following is a brief list of what may happen to recording artists under stress:

'RED LIGHT FEVER'

- Problem: Some artists may seem incapable of performing naturally and without making mistakes when the recorder is 'rolling'.
- Possible solution: Set the multitrack to record constantly, i.e. without a stop-and-go 'take' approach, and instruct the artists to perform as if the session was a rehearsal, i.e without worrying about the control room.

UNNATURAL UNDER OR OVER-CONFIDENCE

- Problem: Some performers may lose their confidence or become overly arrogant while tracking.
- Possible solutions:
  - Reduce the number of people in the studio.
  - Dim the control/live room lights.
  - Play the performers some of the music they love (inspiration).
  - Create an 'atmosphere' in the live room.

CHANGE IN PLANNED RECORDING PARTS

- Problem: Some musicians may change their playing parts or overplay when recording, with negative results.
- Possible solutions:
  - Reduce the number of people in the studio (the musicians may be trying to impress someone or may be feeling intimidated).
  - Discuss the lyrical content and the other elements (parts) of the song and how they connect or support each other (mention a possible theatrical role of each part).
  - Bring up the original part emphasising how an instrument will sit better in the mix by playing the agreed part.

LACK OF 'FEELING'

- Problem: Artists may play their parts in a 'sterile' or 'clinical' way when recording.
- Possible solutions:
  - Discuss the lyrical content and the other elements (parts) of the song and how they connect or support each other (mention the theatrical role of each part).
  - Play the demo.
  - Create an imaginative or artistic 'atmosphere' in the studio.

LACK OF OBJECTIVITY

- Problem: Some musicians may lose sight of the 'big picture' while tracking, e.g. they may want to keep chasing the 'magical' or perfect take or in the case of simultaneous tracking one artist may want to discard a good take and continue recording basic tracks because of errors in their own individual performance.
- Possible solutions:
  - Play the take and highlight its positive attributes.
  - Play the take 'muting' the instrument being questioned and discuss the possibility of overdubbing parts later.
  - Suggest a break and play the recorded take upon returning to the studio (artists may judge it more objectively at this point).
  - Work on a different song temporarily and return to the original one at a later stage (with 'fresh ears').

LACK OF 'DRIVE'

- Problem: Artists may give up too quickly when tracking and deem sub-standard performances as adequate.
- Possible solutions:
  - Suggest a break and play the recorded take upon returning to the studio (artists may judge it more objectively at this point).
  - Play the demo highlighting its positive attributes.
  - Play the performers some of the music they love (inspiration).
  - Create an 'atmosphere' in the live room and discuss the lyrical content of the track and the intention of the composer.

AGGRESSIVE BEHAVIOUR

- Problem: Performers acting unreasonably and aggressively towards each other or towards other members of the production team.
- Possible solutions:
  Call a break and:
  - Step outside the studio (address the offending members of the team individually or in small groups – attempt to difuse situations humorously).
  - Create a distraction, e.g. take the offending members to the lounge and get them to play a video game.
  - Offer the performers something to eat or drink (some individuals become noticeably more aggressive when they are hungry).
  - Consider recording separately / overdubbing parts.

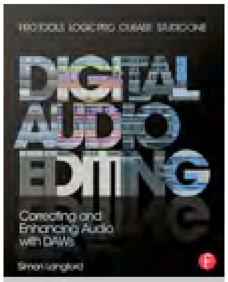
NB Any of the aforementioned problems may be caused or made worse by the presence of cameras or unrelated individuals in the recording environment.

FOR MORE INFORMATION ON THIS TOPIC, PLEASE CHECK OUT THE FOLLOWING CREATIVELIVE VIDEO COURSE:

Check out Your First Studio Job HERE.

Check out Gear Gods presents Studio Pass HERE.

CHAPTER 6 POST PRODUCTION



The following is excerpted from *Digital Audio Editing* by Simon Langford.

You can purchase a copy HERE.

DIFFERENCES BETWEEN FADES AND CROSS-FADES

Both fades and cross-fades are very useful in that they allow us to very quickly and easily make sure that the beginning and end of any audio regions are smooth, with no unexpected glitches. The biggest difference between them is purely a functional one. If there are no regions overlapping on the same track, then you should use regular fades. If your aim is to have one region gradually fade into another, and they are on the same track, then a cross-fade would be used. If, however, you wish to achieve the same thing, but the two regions are on different tracks, then you would still use fade-ins and fade-outs. Equally, if you wanted to have different shapes applied to the fade-out and in portions of a cross-fade and your DAW didn't give as much freedom to customize the crossfades as you might like, then you could move the regions onto separate tracks and apply different "shapes" (as discussed in the following section) on each of the fades. The other benefit to doing it this way is that you can also apply different fade times to the out and in portions, which a "regular" cross-fade wouldn't allow you to do.

DIFFERENT FADE SHAPES

As mentioned above, there are a number of different "shapes" (or curves) that can be used for fades and cross-fades. These shapes are basically a reference to the rate at which the level change happens over the length of the fade. While many DAWs offer you the option to fine-tune or customize the fade curves, here we will be looking only at the commonly offered "preset" curves. There are four main types to consider: linear (sometimes referred to as equal gain), logarithmic (sometimes referred to as equal power), exponential, and S-curve. In order to fully understand the subtleties of difference between the different fade shapes, it might be wise to explain the decibel scale (dB) and the differences between sound levels in your DAW and perceived volumes. There is quite a complex subject, but there is a short summary of this available on the accompanying website, which should give at least an introduction to the concepts referred to below.

LINEAR

The simplest of all fade curves, linear, represents an equal rate of gain increase or decrease throughout the duration of the fade. As we have already stated, though, an equal rate of change of gain doesn't equate to an equal rate of change of the perceived volume. If you look at the diagram above, you will see the gain curves (for fade-in and fade-out) on the left and the associated perceived volume curve on the right. As you can see, this "linear" fade-in curve actually sounds like it stays quite

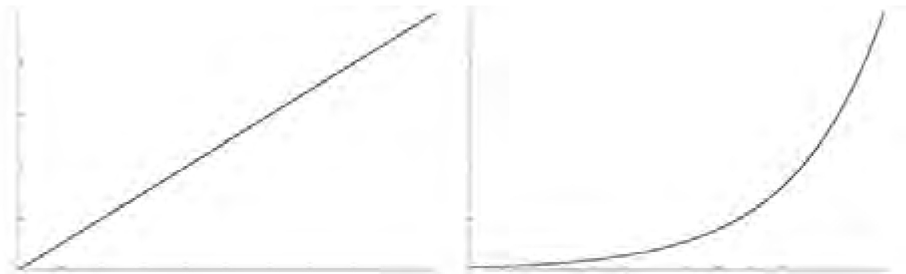


FIGURE 4.1
The curve on the left represents the change level in dB over time, while the curve on the right represents the change in perceived volume over time of a linear fade.

quiet to start with, and then the rate of volume increase builds toward the end. Conversely, the fade-out would sound like there was quite a large drop in volume initially, and then it would seem more consistent toward the end of the fade.

This is quite often the default fade shape in DAWs and for shorter fade-ins and-outs, the kind we might use at the beginning or end of a region, just to ensure that there are no digital clicks at the region borders; these linear fades are perfectly acceptable. Equally, in situations where your audio has a natural ambience or reverb to it that you want to minimize, the quick initial drop in perceived volume of these linear curves could work quite well to seem to shorten the ambience.

This fade curve would do nicely in some situations where longer fades are needed. If you had a sound that you wanted to give the effect of accelerating toward you, then a linear curve might work. An exponential curve (see below) could be used if you wanted to further emphasize this acceleration effect. But for a natural-sounding, longer fade-in or -out, then a linear curve probably isn't the best choice.

The story with cross-fade is slightly different, though. Because the perceived volume drops more quickly at the beginning of the fade, you can see that, at the halfway point in time (halfway across from left to right), the perceived volume is noticeably below 50%. The effect of this on a linear cross-fade is that, at the midpoint of the fade, both sounds are below half of their maximum perceived volume, and as a result the sum of the two will be below the maximum level of either. If the two sounds are both of different levels anyway, and the cross-fade time is long enough, then this might not be a problem, but if you are using a short cross-fade for an edit that is perhaps in the middle of a single note of a performance, then you could have a perceptible dip in volume in the middle of the cross-fade. The diagram below illustrates this.

With this in mind, it is advisable to at least consider one of the other fade curves for short cross-fades between similar levels' audio regions.

LOGARITHMIC

The perceived volume of a sound has a logarithmic relationship with its level in decibels, so it seems sensible to assume that having a fade that works on a logarithmic scale would counteract the curve of the linear fades, and, if you look at the diagram above, you will see that it does do exactly that. If you now compare these curves with the ones for linear fades, you may notice a similarity. In the case of the fade-ins, the shape of the perceived volume curve looks like we have taken the level in decibels curve and pulled the middle of the line toward the bottom right corner. Similarly with the fade-out curves, it looks like we have pulled the line toward the bottom-left corner. In the case of linear fades, it takes the straight line and introduces a curve, and in the case of the logarithmic fades, it takes an already curved line and straightens it out.

The sound of a logarithmic fade should be that of a consistent and smooth increase in (perceived) volume over the whole duration of the fade, and because of this it makes sense that this curve shape can be used for most general-purpose fading tasks. Logarithmic fades are often a good choice for long fade-outs (at the end of songs for example) because of the perceived linear nature of the fade. If you are applying fade-outs to regions with natural ambience, then logarithmic fades would seem very “neutral,” in that they wouldn’t tend to reduce the sound of the ambience, as a linear fade would.

For those of you interested in the mathematics behind how a logarithmic relationship works, I have included an explanation on the accompanying website.

» CLICK HERE

They are also very useful for creating natural cross-fades, especially when you are cross-fading in the middle of a sustained note or word. When you apply a logarithmic cross-fade, the perceived volume at the midpoint of the fade is around 50% of the maximum, so, when the two regions being cross-faded are summed together, you get an output volume that seems pretty much constant.

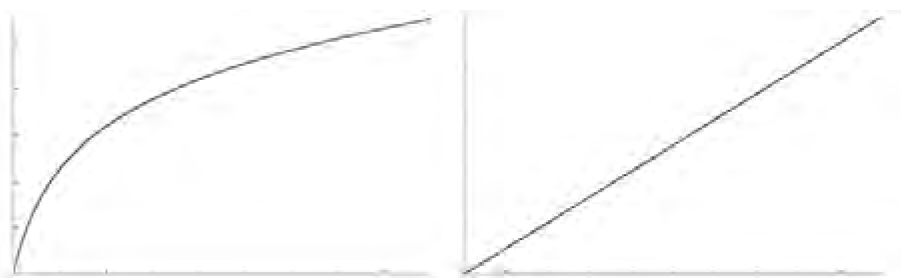


FIGURE 4.2
The curve on the left represents the change level in dB over time, while the curve on the right represents the change in perceived volume over time of a logarithmic fade.

EXPONENTIAL

Exponential fade curves are in many ways the exact opposite of logarithmic curves, in that the fade-ins seem to start off increasing in volume very slowly and then only shoot up really quickly at the very end of the fade, and the fade-outs seem to drop very quickly from the maximum volume and then seem to decrease only very slowly over the rest of the duration. The diagram above illustrates this and also shows that in many ways the perceived volume of an exponential fade could be seen as an exaggerated version of a linear fade. Naturally, it isn't quite as simple as that, but thinking of it in those terms can help to establish what it might be used for. Fading in with an exponential curve would give the impression of a sound accelerating rapidly toward you, which could be useful in certain situations. Equally, using an exponential fade-out at the end of regions with a natural ambience will certainly help to suppress that ambience, and that is very useful in many situations.

Given the shape of the perceived volume curve, it isn't hard to imagine that exponential curves are much less suitable for both sides of cross-fades, unless a very specific effect is required, simply because, at the center point of the crossfade, both signals are quite low in level, so the resulting cross-fade will have a definite dip in the middle. Again, how much of a problem this is depends very much on the length and context of the cross-fade. Longer cross-fades on more ambient sounds may sound fine, with the associated dip giving a little "breathing space" in the middle of the cross-fade, but, if you are trying to use them in the middle of notes or words or for very quick transitions in general, the resulting dip may be very off-putting.

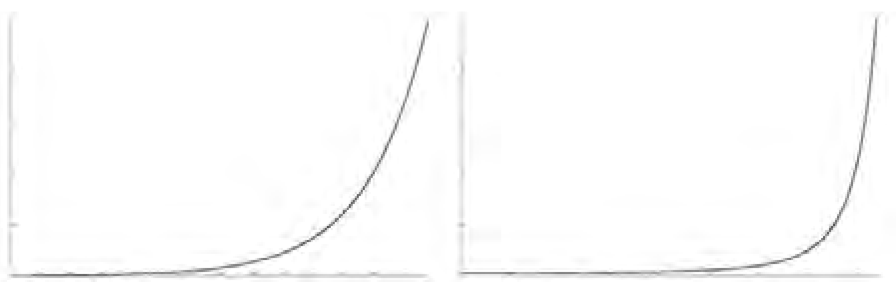
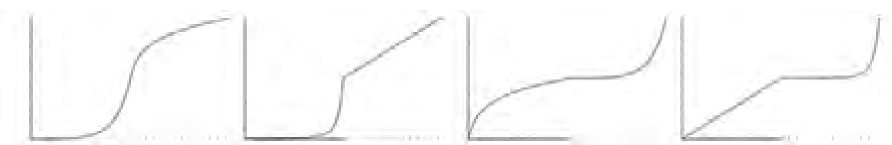


FIGURE 4.3
The curve on the left represents the change level in dB over time, while the curve on the right represents the change in perceived volume over time of an exponential fade.

S-CURVE

The S-curve is quite a difficult one to think about, because it has attributes of each of the previous three types and is further complicated by the fact that there are two different types of S-curves. Like a linear curve, at the midpoint, the level of the sound (decibel level) is at 50%, but the shape of the curve before and after this midpoint is not linear in nature. As you can see from the diagram above, you could almost view S-curves as a combination of a logarithmic and an exponential curve (although mathematically it isn't quite that). In the case of the first example (which I have called Type 1), one that I would consider a more traditional S-curve, you could view fade-ins as an exponential curve up to the midpoint, followed by a logarithmic curve from the midpoint to the end, and fade-outs as logarithmic up to the midpoint and then exponential from the midpoint to the end. Conversely, in the second example (Type 2), the situation is reversed, and the fade-ins are more like a logarithmic fade up to the midpoint and then an exponential fade from there to the end, and fade-outs are exponential to the midpoint and logarithmic from the midpoint to the end.

FIGURE 4.4
From left to right, the curves above represent the change level in dB over time and the change in perceived volume over time of a Type 1 S-curve in level in dB over time and the change in perceived volume over time in a Type 2 S-curve.



When considering applications for S-curves in cross-fading, it can be helpful to think of the Type 1 curves as a kind of halfway house between a linear cross-fade and a simple "cut" between two files. If we had two files directly adjacent to each other with no cross-fades, then the second would start playing at the exact moment the first one finished. If we are lucky, and if we have made sure our regions all start and end at zero-crossing points, the result will be a very abrupt transition from one region to the other. If the edits regions aren't trimmed nicely to zero-crossing points, then we could well get unwelcome glitches or pops. In order to avoid this, we could use any of the cross-fade methods described above, but, in all but the very shortest of cross-fade times, there would be a period when both sounds were audible simultaneously. The same applies with S-curve cross-fades. If there were no point when both sounds were playing simultaneously, then it wouldn't be a cross-fade after all. But what the S-curve (Type 1) does is to minimize the amount of time that both sounds are playing simultaneously. This allows to make edits that sound like direct "cuts" from one sound to another but which have that little extra smoothness and polish to them.

Type 2 S-curves, on the other hand, are generally more suitable for longer crossfades, where we want the smoothness of a cross-fade and the consistency of a cross-fade in terms of overall level but the ability to have both of the crossfading sounds audible for as long as possible. In essence what happens is that there is a relatively short period at the start of each cross-fade where the outgoing sound drops quickly toward 50% and the incoming sound rises correspondingly to around 50%. The rate of changeover then slows, and both sounds will appear to stay at almost the same level for most of the middle half of the crossfade, and then, toward the end, there is a final quick changeover for the final few percent of the fade.

DIFFERENT FADE SHAPES: SUMMARY

To further complicate our descriptions of these basic curve shapes (not to mention custom curves), many DAWs now offer the ability to change the actual shape of the logarithmic, exponential, and S-curve cross-fades. I am not going to go into the mathematics behind these but will simply say that it is equivalent to pulling the logarithmic and exponential curves (and their corresponding equals in the S-curves) away from the position where a linear curve would be. The diagram below illustrates this better, and there is, for consistency, a corresponding perceived volume curve for each one.

In the case of logarithmic fades, changing the shape of this curve affects how close to the start of the fade the sound will rise above 50% and how close to the end of fade-outs it will drop below 50%. Conversely, with exponential fades the change in shape will affect how close to the end of fade-ins the sound will rise above 50% and how close to the start of fade-outs it will drop below 50%. Making changes to the shape of Type 1 S-curves determines how quickly (around the midpoint of the fade) the change will happen, and, in the case of Type 2 S-curves, the change determines the amount of time that both sounds are at an approximately equal level.

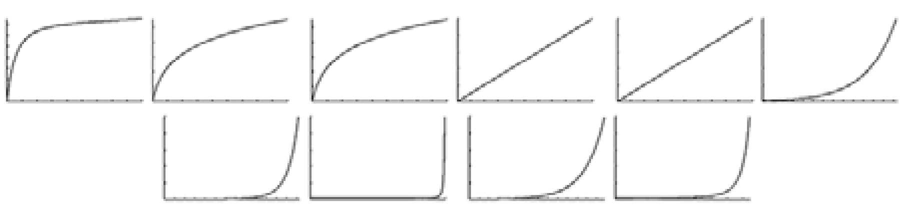


FIGURE 4.5
(From left to right, top to bottom) Decibel level change and perceived volume change for strong logarithmic fade, dB level change and perceived volume change for logarithmic fade, dB level change and perceived volume change for linear fade, dB level change and perceived volume change for exponential fade, and dB level change and perceived volume change for strong exponential fade.

USING FADES

The most basic use of fade-ins and fade-outs is as a means of making sure that individual regions start and finish without the risk of any audible glitches, pops, or other unwanted noises. If there is a good clear section of silence prior to the start of the actual audio, then I recommend a linear fade-in of around 500 ms as a nice, smooth introduction. Equally, and again assuming enough silence at the end of the region, I would recommend a similar 500 ms linear fade-out at the end of each region. If there is insufficient clean space at the beginning or end of the region, then you can always reduce this accordingly, but where possible I like to have these kinds of pre-audio fades as long (up to 500 ms or so) as possible just to make sure everything is nice and gentle. These kind of edits are really just good "housekeeping" edits and aren't always strictly necessary if you've got good basic recordings to work with, but it's always good practice to do them anyway.

Another very common use of fade-ins and -outs is as a means of changing the characteristics of a sound. In its simplest form, this is something as basic as putting a fade-in on the very beginning of a sound to soften the attack of the sound. For vocals this can be very useful in softening up the plosive sounds. These are vocal sounds made by closing the oral passage and then rapidly opening it again, and they occur at the beginning of many different vocal sounds, but the most common culprits are words beginning with "b", "d," and "p." This rapid release of air, especially if the mouth is quite close to the mic at the time, can lead to lots of low-frequency energy and a very pronounced, sharp attack to the sound. Even if we filter out the low frequencies, there might still remain an unnecessarily sharp attack. This isn't always a bad thing, as it can help with the diction and intelligibility of the voice, but, especially in situations where the vocal is otherwise quite soft and gentle, it can make that one word stand out quite a lot. Using a fade-in (probably linear or logarithmic) with a very short time (around 10 ms or so) would be a good starting point, and then the fade time should be adjusted while listening, until the attack has been softened without changing the character of the sound and the intelligibility too much.

You could also consider softening up the attack of drum and percussion sounds. If you have a particularly busy rhythm section in a song with lots of hi-hats, percussion, and even very "snappy" rhythm guitars, the whole thing can start to sound a little messy. This might be down to small timing variations between the instruments and performers. If this is the case, and if these variations make things just a little too loose, then you might consider moving a few things around in order to make them work better together. But there is another potential solution. Instead of trying to match up the timing of a number of elements with sharp attacks to the sounds, you could always try softening up the attack slightly on one or two of the sounds. A fairly short fade-in of something in the region of 25 ms is a good starting point, as that fade-in will give a tiny degree of ambiguity (and therefore fl exibility) to the timing of that sound.

Equally, you can use fade-outs to change the character of sounds by changing their decay characteristics. A snare drum with the snares quite loose has a very distinctive "rattle" to the sound, which can be very interesting tonally but which can also be quite messy. If there isn't too much spill, then a noise gate could perhaps be used to shorten the decay. This would work, however, only if the level was fairly consistent. Any changes to the level of the snare drum going in to the noise gate would change the way the noise gate responded and would mean that some notes (quieter ones) would seem like they had been truncated more than the louder ones. That might be something that works in the context of the editing that you are doing, but, if you want something more consistent, you could always achieve a similar result using fade-outs. If you wanted this consistency, then you would want to do a little preparatory work and make sure that not only was each snare drum a separate region but also, crucially, you would also need to make sure that each one was trimmed nice and tight at the front end and also that they were all pretty much equal in length. Once you know that all of the snare drum regions are the same length, then you can simply select all of them and apply a fade-out. A linear or logarithmic curve would be a good place to start, and it is really just a case of starting with a fade time of around 100 ms and then just adjusting things from there.

Using fade-ins and fade-outs to change the characteristics of sounds in these ways is something that can be very useful in freeing up some sonic space in complex projects, but it can also be very time-consuming, as these kinds of edits need to be performed on a note-by-note basis. However, it is something that I would certainly recommend that you try for yourself and familiarize yourself with, as the effects can be quite special, and, like most editing tasks, the more often you do them, the quicker you get at doing them.

USING CROSS-FADES

The whole point of a cross-fade is to provide a smooth transition between two separate pieces of audio. But within that definition there are a few different situations when we might need to do this. One involves transitioning between two entirely unrelated or at least very different sounds, and the other is when we have two similar sounds and we want to try to make them sound like one continuous one. The process is basically the same, but we have a different set of parameters and criteria to work with in each situation.

The simpler situation is when we wish to apply a cross-fade between two very different (in both tone and pitch) files. In most cases it is simply a case of positioning the two files at the appropriate positions and then selecting an appropriate cross-fade curve and time and making minor adjustments to fine-tune things. In most situations like this, that will be all that is required, simply because the sounds are quite different. There may be times where there is some phase cancellation (see below), but, given the context that these kinds of fades are likely to be used (long cross-fades for background ambiences, cross-fade between different whole songs, and things of that nature), those problems aren't likely to be a major factor all that often. Unfortunately, when we are dealing with two sounds that are quite similar (in tone and pitch), we have a few more things to take into account.

In the last chapter we looked at the importance of zero-crossing edits and also the phase alignment of the two regions. We found perfectly good reasons for this phase alignment simply in the editing process as a part of the pursuit of a natural sound, but there are other implications that become more relevant when we are dealing with cross-fades. In order to fully understand this, we need to briefly look at phase cancellation.

PHASE CANCELLATION

Every sound has a waveform, and every waveform has a phase. In essence, phase is simply the position of any given point in time on the waveform cycle. This is easiest to illustrate with a simple sine wave. Starting from the "neutral" position, the wave first moves upward until it reaches a turning point, then moves downward toward another turning point, and then returns to the neutral position, and the whole process begins again. That initial up-and-down-and-up represents one full cycle and, because phase measurements are in degrees (as used to measure angles), one full cycle is considered to be 360 degrees. Phase alignment (or phase correlation, as it can be called) simply refers to the process of ensuring that the two audio files in question are at the same relative phase angle (measured in degrees) or, at the very least, at a very similar point in this cycle: on an upward trend or a downward trend. The phase of an individual sound doesn't matter in isolation. In fact, our hearing system cannot detect absolute phase. What happens, though, when we don't listen to sounds in isolation, and we start mixing them together?

Any waveform can be considered to have both "positive" and "negative" parts. These correspond to the periods when a speaker is moving toward you and away from you, respectively, and are represented by the parts of the waveform shown to be above or below the imaginary midpoint (often shown by a horizontal line in the waveform display). What is important is the relationship of the sounds that we are cross-fading to each other in this positive/negative sense. If both of the sounds are moving upward at the same time, then, when we add them together, we will get a cumulative effect. This is what we are looking for. Conversely, if one sound is moving upward while the other is moving downward, there is a chance that one will "cancel out" the other. The diagram below illustrates this with simple sine waves.

The top two regions on the left represent two sine waves that are the same frequency and in phase, and the result of cross-fading the two regions is the third region below. With the "in phase" sine waves, you can see that not only is there no change to the waveform, but also there is no change to the volume, because an equal-power cross-fade was used. On the right we have a similar setup, only the two sine waves are now 180 degrees out of phase. If we look at the result of cross-fading the out-of-phase sine waves, we can see there is something present at the beginning (when only the first wave is audible) and something present at the end (when only the second wave is audible), but, at the midpoint of the fade, when both are being mixed at equal levels, there is a period of silence owing to the phase cancellation.

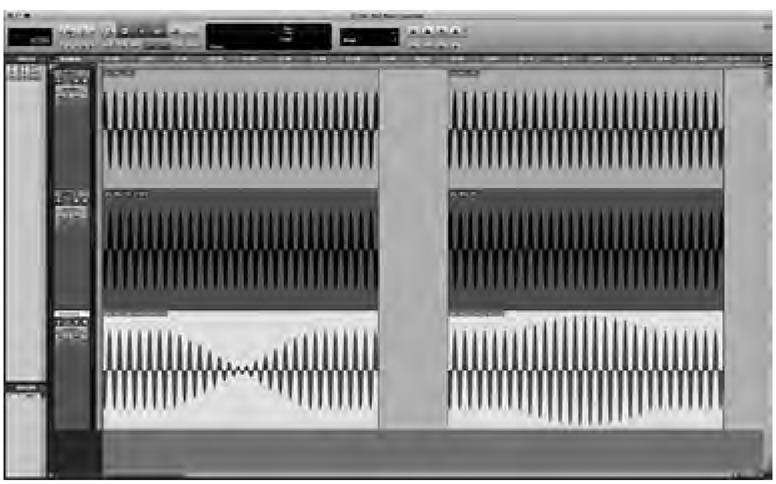


FIGURE 4.6
The top two tracks on the left are sine waves that are 180 degrees out of phase, and the third track is the result of cross-fading those two regions. Note the dip in volume at the center of the cross-fade. On the right the two top tracks are sine waves again, but this time they are in phase. When these two are cross-faded, there is no loss in volume. The slight increase in volume is a result of the cross-fade curve's not being of equal power.

Now I will admit that the times when you will be required to cross-fade two pure sine waves of exactly equal level and frequency are going to be rare, to say the least. And as soon as we change the waveform shape (by adding in harmonics or other notes or sounds), the frequency, or the level, we will not get 100% cancellation as shown above. What could happen instead is that there are parts of the sounds that cancel each other out, which, in reality, can actually sound even more obvious than a simple temporary volume drop. And even the most experienced and technically gifted editor wouldn't be able to tell exactly what the frequency content of a sound was just by looking at a waveform display. That doesn't mean that the waveform display is utterly useless in this regard, though.

PERIODIC PATTERNS

As you zoom in (horizontally) to a waveform display, you will, at a certain point, and in most (but not all) sounds, start to see a pattern emerging. It is very unlikely to be a simple pattern—like that of a sine wave—but there will almost certainly be some kind of cyclic repetition. This is what you need to look for, even if you don't know what frequencies it represents. If you are trying to cross-fade two sounds with similar tones and frequencies (as we are discussing here), then you should be able to see a similar pattern emerging in both sounds. If you have a similar pattern at the end of the first region and the beginning of the second region you are trying to cross-fade, then there is a good chance that you can get things sounding very smooth indeed.

Once you have found these points, you can start to position the regions to get the best result. As you move one of the regions (usually the second one) backward and forward along the timeline, there will start to be an overlap between the two. Making sure you have an appropriate zoom level set (to where you can see a good handful of the "cycles" that you have visually identified on either side of your cross-fade point), you can then visually line up the second region so that its start point (the start of one of these cycles) lines up with the start point of one of the cycles in the first region. Once you have completed this visual step, you should have a listen back. The chances are there will be some kind of audible issue at the edit point. It might not necessarily be a click or a pop or any other artifact of a bad edit, but there could be a perceptible change in pitch or tone (depending on how different the two regions were to start with), and this is where the cross-fades come in.

In the case of two regions with subtle pitch and tone differences, you should start off with a very short cross-fade (just a few milliseconds) and then increase the length, until it sounds right. The reason for this is to minimize the risk of any strange cross-fade effects such as phase cancellation by minimizing the amount of time the cross-fade takes. Because the two regions are very similar, there is a good chance that, even with quite a short cross-fade, we wouldn't hear any major differences. If the two regions were quite different, though, we would hear an abrupt change.

FIGURE 4.7
While it is far from a simple waveform, the vocal in the image above does have a clearly repeating pattern. Ideally you want to make any edits at a similar point in this cycle for maximum smoothness and consistency.

The same applies to sounds with a larger tonal or pitch change, but with the difference that, unless you are specifically happy with a sudden change, you are likely to have to make the cross-fade quite a bit longer before there is any subtlety to the edit. And of course the longer the cross-fade, the more likely we are to have problems. If you find that you just can't get it to sound smooth, then you could consider changing the shape of the cross-fade curve (perhaps even making it an asymmetric one), because, if you absolutely need to get these two regions cross-faded, it might be that a small dip in volume (as a result of the curves used) in the middle of the cross-fade could help to mask a little "bumpiness" in the transition.

Even with all of the options that we have for cross-fading, there will still be times when we can't quite achieve what we want to. One example that comes to mind would be a situation where we wanted to take two sustained vocal notes with different pitches and blend them together smoothly. Using the techniques that we had for cross-fading simply wouldn't do this smoothly, as the fades are not content-aware. The fades don't know what the pitches of the sounds being faded are; they work simply on the volume/level of the sounds. If we used a cross-fade on our two vocal notes, we would end up with a period of time during which both notes could be heard simultaneously rather than gliding from one pitch to another, which we would expect from a singing voice.

In general, though, with the exception of long, subtle fades, most cross-fade use will be when you are trying to blend similar sounds (perhaps different takes of the same performance) or to smooth off a transition when you have had to remove a part of the middle of a region (used extensively in some time-stretching scenarios) and need to move the parts on either side of the gap together. In all of these situations—where we want the end result to be inaudible—we can often get really great results with a little patience and effort.

HANDS ON

INTRODUCTION

In this section we will take a look at the practical ways to create different types of fades and the different shortcuts and tools available in each of the DAWs. Like cutting and pasting, because these operations are carried out so often, there are numerous ways in which you can achieve the same result, and there is sure to be a solution to fit your individual work-flow and needs. The options presented here are just examples of some of the easier and more convenient ways in which you can get results.

LOGIC

At the end of the last chapter, we saw that it was possible to change the default preferences of Logic to include an option for the Marquee tool to be automatically selected under certain circumstances. Equally, it is possible to set things up so that the Fade tool is automatically selected. In order to do this you need to go to Preferences > General > Editing (Tab) and make sure that the check box next to Pointer Tool in Arrange Provides: Fade Tool Click Zones. Like the Marquee tool, choosing this option means that the standard cursor/tool will change to the Fade tool under certain circumstances. In this case, when the cursor is positioned in the top left or top right corners of an audio region, the Fade tool will be automatically chosen. This tool looks like a vertical bar crossed by a horizontal line, with left and right arrows at either end. Please note, however, that if you have Flex Mode activated on a particular region, then you will need to turn the Flex view off (so that the Flex markers are not showing on the region) in order for this to work correctly. If the tool is showing as it should, then all you need to do is click and drag to the right (at the start of a region) or left (at the end of a region), and a fade-in or -out (as applicable) will be created that extends as far as you have dragged.

Once the fade has been created in this way, you have a number of options. At the start (fade-out) or end (fade-in) of the fade, there will be a vertical line overlaid onto the region, and if you place the cursor at the top of the region at this point, you will see the Fade tool appear again, and you can move the start or end of the fade as desired. However, if you move the cursor to the top of the region at a point within the fade area, you will see a slightly different tool appear. This time, rather than being a vertical line with left and right arrows either side, it will look like a horizontal line, with the arrows at either end and a dot in the middle. If this tool is showing, and you click and drag left or right, you will see that the shape of the fade changes. In Logic-speak this is changing the curve of the fade, but this simply equates to a change from linear to either exponential or logarithmic curves of varying strength.

The usefulness of the Fade tool doesn't end there. Both of these variations on the Fade tool can also be used to create and edit cross-fades. If the Fade tool is positioned between two adjacent (and they must be touching or overlapping) audio regions and then clicked and dragged, a cross-fade will be created that is symmetrical either side of the crossover point. Once the initial cross-fade has been created, we can change the length of the crossover, as we did before, with the difference that, in the case of a cross-fade, the length of only one part of the cross-fade will be changed. While the initial cross-fade was created as symmetrical, these length changes will not be, and the resulting midpoint of the cross-fade will move as we adjust the length of either side. Cross-fades can be made between regions with a gap in between, but there is a slight variation in method. In order to do this, you will need to position the cursor to show the Fade tool at the end of one region and then click and drag as if to create a fade-out, but then, while still clicking, drag in the opposite direction to drag over the start of the following file. Once the Fade tool has reached the following region, a cross-fade will show, which can then be varied in the ways described above.

We can also use the variation of the Fade tool to change the shape of the crossfade, as we did with the fades, but things are a little more complex here. Whereas for fade-ins or -outs there is a simple change from exponential through linear and on to logarithmic, with cross-fades there is a reshaping of both sides of the fade but in an inverse manner. What this means is that if one side of the cross-fade moves toward logarithmic, then the other side will simultaneously move toward exponential. Given that an exponential curve is the inverse of a logarithmic curve, this makes a lot of sense, as doing things this way will preserve a constant power throughout the cross-fade, which is, in most cases, the best option for a smooth cross-fade.

The default behavior for dealing with overlapping regions in Logic is to simply switch playback from one region to another. The first region will play until the start of the second region, at which point that one will take over immediately, even if there is a portion of the initial region still to play. There are, however, a number of other modes for dealing with situations like this, and the most useful to us in this context is the option to automatically create cross-fades. There is a drop-down box at the top right of the Arrange Window called Drag, and if you choose X-Fade, any regions that are dragged (or resized) and overlap another region as a result will automatically have a cross-fade applied. The cross-fade itself will last for the duration of the overlap, and once in place the ends of the cross-fade and the shape of the curve can be adjusted in the same ways as we have already described. If you prefer to have this mode set to

"Overlap" in general, you can always change it to X-Fade temporarily if you are working on something specific that would benefit and then change back afterward, as any cross-fades created while X-Fade mode was selected will remain in place if you switch back to one of the other modes. If it is something that you find yourself swapping in and out of regularly, then it is possible to set up custom key commands for each of the modes to enable a quick change.

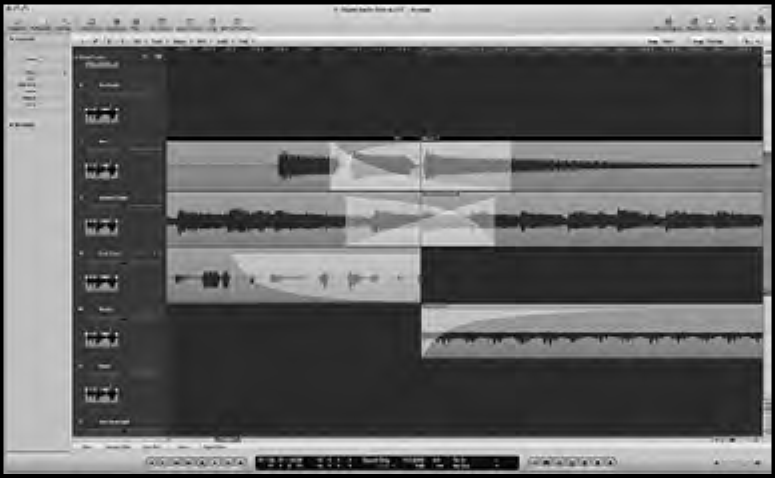


FIGURE 4.8
There are many different options for fades and cross-fades in Logic, including equal power and equal gain options and fully variable logarithmic and exponential curves.

Using a combination of these two techniques, along with some of the other options available, will mean that you spend much less time having to change tools or navigate through various menu options in the middle of an intricate editing process. Such tasks can often be repetitive and time-consuming, so any methods that we can use to speed up the process will be extremely valuable to us.

PRO TOOLS

The first and perhaps most obvious way to create fades is to use the Selector tool to highlight the section that you want to create a fade for. If you choose only the end of a region, then a fade-out will be created; choosing only the beginning will create a fade-in; and choosing a selection that overlaps two regions will create a cross-fade. Once you have the selection in place, all you need to do is press Cmd[Mac]/Ctrl[PC] + F or choose Edit > Fades > Create to open the Fades dialogue.

The other (some would say easier) way of creating fades is to use the Smart tool to create the fades. When the Smart tool is selected and positioned in either of the top two corners, the cursor changes to a square divided in half diagonally, which can be used to create fade-ins and outs. Clicking and dragging from either top corner toward the center of the region will create a fade-in or -out (context-dependent) of a length determined by how far you drag.

If, however, you position the cursor at the bottom corners of two adjacent regions, then the cursor changes once more to a square divided into four diagonally, and this can be used to create cross-fades. As soon as you click, a cross-fade will be created that is symmetrical either side of the cross-fade point and the length of which is determined by how far you drag. If you create fades or cross-fades using the Smart tool, then it will be created using default parameters, but you can still go in and fine-tune the settings once the fade has been created. The method used here is the same method that you would need to use if you wished to make changes to any existing fade no matter which method you used to create it. If you click anywhere within the fade area, it will select the whole fade region, and you can use Cmd[Mac]/Ctrl[PC] + F (or Edit > Fades > Create) to reopen the Fades dialogue and make changes from there. A quicker method, though, is to simply double-click on the fade region with the Grabber tool (or Smart tool at the bottom of the region), and this will also open the Fades dialogue.

Once you've arrived at the Fades dialogue, you will find that it is consistent in layout whether you are doing fade-ins, fade-outs, or cross-fades but with different options visible, depending on which type you are creating. At the very top left there are four buttons that allow you to customize the viewing and auditioning options, but more important are the actual curves themselves, as this is where all the character and feel of the fades are determined. What we have are independent sections for fade-ins and fade-outs and a central Link section that will create curves with a degree of symmetry.

If you are creating a fade-out, then only the left hand Fade-out section and a central Slope section will be visible. There are three options under either Out Shape or In Shape : Standard, S-Curve, and a pop-up selection box with seven different preset shapes that vary from a nearly instantaneous change through exponential, linear, and logarithmic. If you choose with Standard or S-Curve, you can click on the colored line representing the shape of the curve and drag to the left or right to fine-tune the shape of the curve. In the case of Standard, this will move from a linear fade at the center point through to exponential and logarithmic at either extreme. If you choose S-Curve, then moving either side of the central point will give you the Type 1 and Type 2 S-Curves we spoke about in the main chapter. If, however, you choose one of the preset shapes, then no adjustment is possible. Naturally, everything we have just said applies for fade-ins just as much.

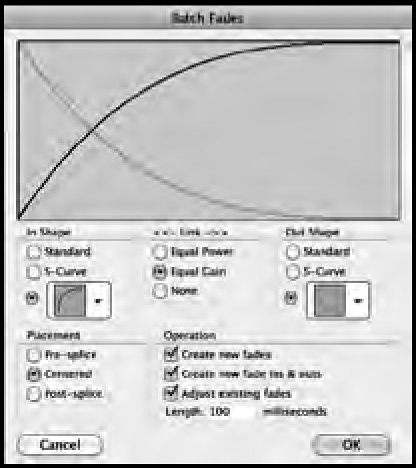


FIGURE 4.9
The Fades dialogue in Pro Tools offers a large number of choices ranging from simple equal power and equal gain cross-fade to more esoteric and customizable options.

The Slope control has a choice of *Equal Power* or *Equal Gain*. We explained in the main chapter that the decibel scale is a logarithmic one, which means that a constant interval of gain change (in steps of 3 dB, for example) does not equal a constant change in power (or perceived volume). As such, Pro Tools gives us the option to scale our fades to either scale. The effect on the curves is simply to give slightly different shapes to each of the options. An Equal Power fade should sound smoother and more consistent than an Equal Gain one, but, as always, let your ears be the judge of what works best in any given situation.

Things become a little more complex if you are working on a cross-fade, simply because you have the option to have both sides of the cross-fade linked to create a degree of symmetry if you so desire. If the Link control is set to either Equal Power or Equal Gain, then the cross-fade created will, by default, be symmetrical about the center position and have curves that are chosen to give (unsurprisingly) either an equal power or equal gain throughout the duration of the cross-fade. However, that isn't the end of the story. As with fade-ins and -outs, if either Standard or S-Curve is chosen, then clicking and dragging on the curve line will change the shape. The obvious addition here is that, as you drag on either curve and change the shape, the shape of the other corresponding curve is updated automatically as well.

If you choose one of the preset shapes from the pop-up list at the bottom, then, while you might not be able to change the shape of each curve individually, you can choose different shapes for the fade-in and -out curves. If you have the Link mode set to Equal Power, you will notice that some of the preset shapes are grayed out, simply because it wouldn't be possible to create an equal power cross-fade with those shaped curves, but, with Equal Gain chosen, you have the full choice for both fade-in and fade-out curves.

Things start to get really interesting when you set the Link mode to None, because not only can you choose Standard, S-Curve, or the presets separately for the fade-in and fade-out portions (with the adjustable curve shapes for both Standard and S-Curve), but also you will see black dots at each end of each curve, which you can click and drag earlier or later to further fine-tune the shape of each curve. Of course, once you start getting into the realms of this much adjustability, then it will be very difficult to preserve either a constant gain or a constant power throughout the cross-fade, but, to be honest, having this level of flexibility in the cross-fades is definitely worth risking a little unevenness, especially given that you can always choose one of the more-standard options if things go awry during your experimentations.

SHOULD THIS BE A PART OF THE EDITING PROCESS?

Some people feel that the subject of level control of any form should be left until the mixing and production stage, and there is a lot of sense in that. However, what we are talking about here, in the context of level control, is the process of correcting problems and anomalies with levels. The aim of level control in the context of audio editing is to correct any problems to make sure that there are no unwelcome surprises at the mixing stage.

Given that largely the same tools will be used for level control during editing and mixing, we could ask why we should bother making these changes during the editing process—surely it is better to keep the maximum amount of flexibility at the mixing stage. While this is true, there are certain things that will be extremely obvious, and, if we can fix those problems during the editing stage, then it will free up the mix engineers to focus on the more creative tasks that they have to do.

Naturally there is a fine line between things that need to be done and things that could be done, and that line can vary from person to person. There are, however, a couple of areas where I believe that it is absolutely necessary to work on level control at the editing stage, and, if that isn't done, the problems can be exaggerated later on. The two most common of these are cleaning up background noise in recordings and achieving consistency in comped tracks. The latter is especially relevant if you are cross-fading in the middle of sounds. But before we take a look at these and other situations where we might use level control during the editing process, let's first look at the different ways in which we can control the levels and address the question of whether "destructive edits" or "nondestructive edits" are preferable.

DESTRUCTIVE VERSUS NONDESTRUCTIVE EDITS

Perhaps the first thing we should do is define and differentiate between destructive edits and nondestructive changes. In essence, the distinction is simple. A destructive edit is one that results in changes to the underlying audio file. A nondestructive edit is essentially a change that is laid on top of a file, is calculated in real time, and does not have any effect on the underlying file. With that in mind, why would you ever choose to use destructive edits if the same thing could be achieved nondestructively?

To begin with, destructive edits on computers have a number of levels of safeguards to prevent you from doing anything irreversible (unlike any destructive edits on tape that were either permanent or a huge problem to rectify). The first and most obvious is the use of some kind of preview feature. Quite often you will be able to preview the effect of an edit before you commit to it. To do this, the software creates a copy of the file you are working on and then applies the edit to that copy. You can then compare the edited copy to the original, and once you are happy to move forward, you commit the edit, and the original is overwritten with the edited copy. This is a first line of defense against applying an edit where you have incorrectly set a parameter. What it doesn't do is allow for you to change your mind afterward for any reason.

Then we have the Undo command, which is a lifesaver if you realize that you have done something wrong only after you hit the Apply button. The Undo command has limitations, though. It will be of use to you only if you have not completed too many other steps since (the number varies by software). And given the nature of editing, it is distinctly possible that you may apply an edit, carry out several other steps, and only then think that you could perhaps have done it differently. The other downside to using Undo in this way is that, of course, if you step back through the steps you have taken to get back your original file, even once you have changed the parameters and applied the edit again, you will then have to redo all the other steps you took, and you may not remember what exactly you did.

Another possibility is for you to manually create a copy of the file and apply the destructive edits to that. By keeping an untouched "original" copy, you know that you are safe, even if there is some problem with trying to undo the changes that you have made. The most useful benefit of working this way is the ability to create different versions of your work at various stages. A typical use might be to have a completely untouched original version, to then remove any background noise or unwanted sections and apply fade-ins and fade-outs to the different regions, and then save to a new file. Then you could create a comped file and save it under a new name. From there you might adjust the timing of a few parts, apply a little pitch correction, and adjust a few levels before creating a final file. The only real downside to this approach is that you can end up with a lot of different versions of files (depending on just how cautious you are). Modern hard drives are large enough for the actual size of all these files to be not much of a worry, but that doesn't allow for the fact that, unless you are very strict with your file naming and general "housekeeping" within your project, you can easily end up very confused if you ever have to open up the project at a later date.

One useful method is to settle on a suffix scheme that you can apply when you name your different stages of edits. In the example given above, you could save the original untouched file as Test-ORIG.wav, the cleaned-up version as Test-CLN.wav, the comped version as Test-COMP.wav, and the final version as Test-FIN.wav. If this is something that you are consistent with, it can greatly help your work flow and make locating different versions very simple.

As you can see, there are a great number of ways we can stop destructive edits being especially "destructive," but in the end, we will come to a point where we are happy with the result and we have committed to it. The edits are then a part of the underlying file, and we know that if we pass that particular file over to somebody else, it will be exactly as we meant it to be on their system as well. In fact, if you are editing files for somebody else, unless you are 100% sure that they have a system that is fully compatible with yours and that they will be able to open any project files that you send them, then you will, at some point, have to bounce/render your edits anyway in order to ensure full compatibility.

Another very important consideration relates to the use of the files/regions. If we commit to a destructive edit, then, because the underlying file has been changed, any instances of (or references to) that file or region throughout the entire project will be changed in the same way. How much of a problem this could be is hard to say conclusively, as it very much depends on how you work and how much copying and pasting there has been of different regions throughout a project (and also how your DAW handles destructive edits, as some will automatically create new files for destructive edits). On the one hand, if there is a particular sound that you have used a number of times throughout a project and there is some need to change an aspect of it, if you make that edit destructively, then you know that all instances of it will be changed, and that method can save you a lot of time. Equally, though, if you don't realize that you have used the same file or region somewhere else in the project, then making a destructive edit could cause problems elsewhere that you don't even realize.

If, however, you have a file or region that is used in multiple places throughout a project and you want to apply a change to it in only one of those places, in that instance, it would be better to apply the change nondestructively to ensure that no other copies are affected. But at the same time, if you apply changes like this nondestructively, then there is always a chance that you might miss one copy of it somewhere in the project.

So there is no conclusive right or wrong approach. Each method has its advantages and disadvantages, and your choice will be largely down to the individual project and your personal preferences and work flow. I would always advise, however, that you check for multiple copies (or aliases) of any file or region before you do any destructive edits, just to make sure that you know exactly what you will be changing.

DIFFERENT TYPES OF LEVEL CONTROL

DESTRUCTIVE EDITS

The most basic of all destructive level control methods is to apply an overall gain change to a file. This is usually a simple case of adding or subtracting a certain amount of decibels consistently over the whole duration of the file or region. However, if too much of a positive gain change is applied, then the resulting file will be clipped and distorted, and this can be a big problem. In order to avoid this happening, if getting the maximum possible level is what you are trying to achieve, there is a dedicated function: Normalize.

What the Normalize function does is to first scan the file to determine the current maximum level in the file and then apply a gain change to the entire file to bring that current peak level up to the maximum possible value. That doesn't mean, however, that all or even most of the file will even be close to the maximum possible level. If that is what you want to achieve, then you will need to consider something like a limiter plug-in, and, to be honest, anything like that should probably be avoided during the editing process and left until the mixing process. In fact, I would probably advise, as a general rule, against even normalizing audio files. When a file is normalized, it means that any additional processing (during either the editing or mixing process) that applies any kind of positive gain change will result in the file clipping (briefly going beyond the loudest possible level resulting in the waveform being limited to the maximum level for the furation of the "clip" and the resulting waveform shape being changed or "distorted) at some point, so anything we can do to avoid that is welcome.

FIGURE 5.1
Normalization is one of the simplest forms of level control in audio editing. The idea is to make a recording as loud as it can be without clipping, as you can see in the example above, where the bottom track is a normalized version of the track above it.

In addition to these constant gain changes, many DAWs and stand-alone wave editors also allow us to apply "gain envelopes." This simply means that the gain we apply isn't constant throughout the whole file or region. We could use this for something as simple as a gradual increase or decrease in gain (similar to fades) or perhaps a situation where gain is applied only to certain parts of the file or where something more complex is required.

Moving on, we will take a look at the many ways in which we can control levels nondestructively. They generally fall into one of three categories—static, variable, and automatic—and we will look at each in turn and look at the pros and cons and discuss possible applications.

STATIC NONDESTRUCTIVE EDITS

The easiest way to achieve a nondestructive gain change, where applicable, is to simply add a gain "offset." This will add (or subtract) a particular amount of decibels to the level upon playback. It should be noted, though, that because this is operating on the file itself, it will change the level of the signal going in to any plug-ins on the mixer channel for that file. If you have any plug-ins that are level dependent (compressors, noise gates, amp modeling, etc.), then changing the gain in this way could change things other than just the volume of the sound. On the other hand, this may be exactly what you are looking for; perhaps one region is being compressed a little too hard, or perhaps one region is a fraction too quiet and isn't opening the noise gate. Using volume changes in this way can help to resolve those problems.

Another option, if it is available, is to use a region volume envelope, which is often drawn directly onto the region in the Arrange window. While these are generally used for more complex and variable changes, you can, if you choose, use them for constant gain changes, but you should once again remember that doing this will also change the volume of the sound before it reaches any plug-ins, so the same caveats apply regarding level-dependent plug-ins.

A third option is to use a gain-changing plug-in in conjunction with track/channel automation. This plug-in will have the same effect as moving the main channel fader but with the additional flexibility that it can be placed anywhere in the signal chain after the audio files itself and before the final channel fader. This has the added benefit that, when you have a gainer plug-in on a channel, it could serve as a reminder you that you have made some level changes on that track. Also, by using a gainer plug-in, you can try moving its position around to precede or follow any other plug-ins that may be on the channel, to determine where the volume change is most suitable. You would need to make any adjustments to the level on the automation for the plug-in, but that is pretty similar to making changes on a region volume envelope, so, with the exception of opening the plug-in and moving its position in the chain, the process is very similar to using the region envelopes, only it offers quite a lot more flexibility.

The final option is to actually automate the level of the main channel fader. If we have no additional processing on the channel (no plug-ins), then all four methods should give us exactly the same result. Any processing, however, would mean that one of the other options might be better. This method, like the others, does have its disadvantages. Although you wouldn't ordinarily do so during an editing process, you might find that you need to make some quick level changes during a later mixing stage, and, in that situation, you might have reverb or delay plug-ins on the channel itself. Using any of the first three methods will allow the reverb or delay effects to tail off naturally, but if you automate the main channel fader to do a quick drop of a couple of decibels on a particular word or note, then you will also be dropping the volume of the reverb or delay tails temporarily, which can sound very unnatural. Therefore, automating the channel fader itself isn't really especially suitable for "spot" changes to volume levels, unless the channel itself has little or no plug-in processing on it, and certainly no reverb or delay effects.

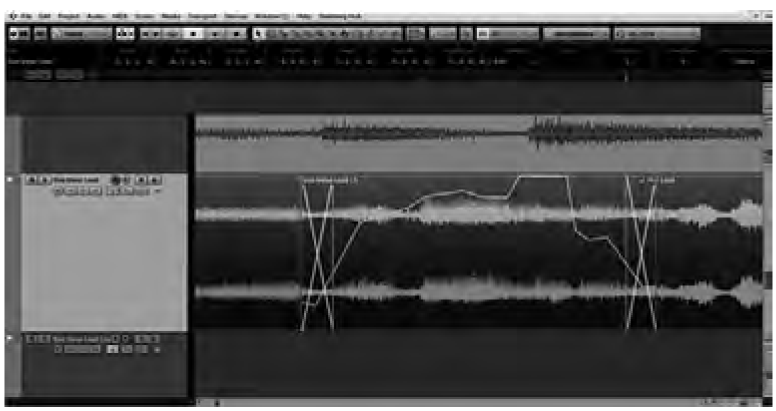
FIGURE 5.2
Volume envelopes (or Clip gain) allows for simple control of a region or part of a region within the Arrange window. This type of level control is independent of any volume automation and will take effect prior to any plug-in processing.

VARIABLE NONDESTRUCTIVE EDITS

Each of the methods that we described above for static level changes can also be used for variable level changes, with the (obvious) exception of the simple region gain offset that we listed first. And each of them has the same benefits and drawbacks associated with them in these situations as well. In fact, the only real difference between the two situations is that of subtlety. The variable level changes that I am talking about here are not aiming to even out the volumes throughout an audio file, as that is more of a mixing task. Instead, what I am talking about is still, in essence, a corrective measure. It would be used in situations where what we are looking for isn't that different from a static change but perhaps with a little extra control.

One example might be in a comped vocal where there is one word from one take in the middle of two words from a different take. The vocalist may have sung the first take a little closer to the mic or with a little more power, so the first take is just a little louder than the second. The word we want to use from the second take starts off at a good level but then tails off slightly in volume. The beginning of the word, therefore, might sound like a good level compared to the first take, but we might feel that the end of the word just needs a little lift to bring it more into line with the word that follows. We are walking a fine line here between editing and mixing, and your own opinion may be that what I have just described is a part of the mixing process, but, even though it is a close call, I would consider something like this to be editing, purely because we are doing it as a part of the comping process, and the use of this slight volume ramp is to make the inserted region the same (dynamically) as the one we are replacing with it. In that sense, at least, we aren't making any major creative decisions. We are just making the word from the second take as close as we can (dynamically) to the same word from the first take.

FIGURE 5.3
In some cases, volume envelopes can allow for far more elaborate control of the recording, as shown in the example above. The degree of complexity here allows for very accurate fine-tuning of the recording, before it is processed by any plug-ins, which can reduce the need for relying on compression to even out volume discrepancies.



Needless to say, if we are going to do that, then there will always be a temptation to just go in and poke around with things a little more. And this is where you will have to learn discipline. If in doubt, just ask how far your client wants you to go with the editing. If it is strictly an editing job, then you probably shouldn't be overly concerned with evening things out too much, as anything beyond simple, corrective level changing is more the realm of mixing and production.

AUTOMATIC NONDESTRUCTIVE EDITS

The final type of level control that we can incorporate into the editing process requires the use of additional plug-ins that will take some of the guesswork out of controlling the levels. In most situations using plug-ins (dynamics plug-ins in this case) is something that would generally come further down the line, but there are a few situations when plug-ins can be used as substitutes for some of the manual techniques we have been looking at.

A simple and fairly noninvasive example is to use a compressor (on a drum sound, for example) with a fairly gentle compression ratio (no more than 4:1 or so) and a Threshold set at 2 to 3 dB above the average level, with Attack and Release setting appropriate for the sound. The overall effect of this compressor should be—nothing! If we have set this up right, then the compressor will hopefully do nothing at all but will always be just on the edge of kicking in, and if it occasionally kicks into action, then it's probably OK. If, on the other hand, it is taking action every couple of bars or even more often, then we might need to adjust the Threshold a little. This may seem pointless, but in fact we are just using it as a safety net in case anything slipped by unnoticed. You could arguably use a peak limiter instead of a compressor, but, in a situation like this, I think that a compressor is a gentler approach, and, seeing as we still have the mixing stage to consider, if we can avoid squashing the sound at any point, it gives the mix engineer the most possible freedom. It should also be noted that, for a task like this, we would want a compressor that is as "transparent" as possible and not one that adds a vintage character or anything like that. That may well be desirable at the mixing stage, but for the purposes that we have here, the cleaner, the better.

HANDS ON

INTRODUCTION

While a large amount of level control will take place at the mixing stage (and rightly so) for a number of reasons that we have outlined in this chapter, there are times when it will be a part of the editing process. With the ease of creating automation data in modern software and the proliferation of level-control plug-ins, many people will use these options, but there are other options available if you are aiming to simply raise the level of an individual region or selection by a static amount.

LOGIC

There are two types of nonautomation level control in Logic, and those are destructive and nondestructive. The choice of which type to use is, I find, largely dependent on what you are aiming to achieve. If you are looking for broad changes to an entire recording, such as bringing up the overall level to compensate for a quiet recording, then the destructive methods will be more appropriate. If, on the other hand, you are looking to make changes to smaller regions or selections in context, then the nondestructive method is usually the best choice, as it gives a very quick and easy way of changing the settings later if you feel they aren't quite right.

In order to apply the destructive processes to an audio file, you will first need to have the *Sample Editor* window open for the file you wish to work on. This is achieved by clicking the *Sample Editor* button at the bottom left of the Arrange window. It is possible to set up a key command to do this, but none is set by default in the Logic preferences. Once the Sample Editor window is open, there are two main options for static level control. The first of these, Normalize, can be accessed by pressing Ctrl + N. When you use a key command to start the normalizing process, you will see a dialogue box that gives you a warning that this is a destructive process and asks if you wish to proceed. If you choose the Normalize command through a menu, you don't get this warning, but that is because it is far easier to inadvertently use an incorrect key command than an incorrect menu choice, so there is this additional safeguard to protect you.

If, instead of normalizing, you want to apply a fixed gain of your choice to the region, then you should choose Ctrl + G, which brings up the Change Gain dialogue box. Here you have choices to set the gain change in absolute (dB) or relative (%) terms. Usefully this box will indicate if the desired gain change will result in clipping before you apply the processing. Once again, if this process has been initiated through a key command, you will receive the warning about the destructive process.

It is worth noting that, with both of these key commands, you will need to make sure that the focus is on the Sample Editor window rather than the Arrange window. If you select an audio file on the Arrange window and then open the Sample Editor window, the focus will automatically be set. You can tell where the focus is, because there is a lighter highlight around the edge of whichever window has the focus. If you wish to simply process the currently selected region, then you shouldn't have any problems. If, however, while the Sample Editor is open, you select a different region in the Arrange window, you will notice that the focus goes back to the Arrange window. If you now try to use your key commands, they won't work, and you will need to click back into the Sample Editor window (click on the menu bar at the top rather than the waveform display) to get the focus back before they will work.

Moving on to the nondestructive level control, there is only one real option available. which is the Gain parameter available in the Inspector panel to the left of the Arrange window. If this isn't visible, then simply click on the Inspector button (a blue circle with a white "i" in it) at the very top left of the screen. In this Inspector panel, you will see a number of parameters that all apply to the selected region(s), but the one we are interested in here is the Gain parameter. Clicking and dragging enables you to create a nondestructive gain offset between –30 dB and +30 dB in 1-dB increments. Unfortunately, as useful as this feature is, it is arguable that it could have been taken further. For many, the minimum increment of 1 dB could prove restrictive, but perhaps more of a potential problem is the fact that the waveform display inside the regions in the Arrange window doesn't change to reflect the change in gain. While any obvious clipping as a result of a gain change applied in this way would be audible, it would still be beneficial if the waveform display did change as any gain offsets were applied, as it does in some other DAWs. At the very least, it would be helpful if each individual region showed, in the region itself, if the gain had been offset and by how much, if only as a time-saver to avoid having to select each region individually and look at the Inspector panel to see what changes had been applied.

Finally, it is worth again referring back to the Hands On section at the end of chapter 3 and the use of the Marquee tool to create automation nodes, because this does give us a way, albeit a little bit of a work around, to actually get a visual display of any gain changes applied. By creating these automation nodes for any regions or selections that you wish to change the gain of, you will have a visual reference of any gain changes when the automation is visible. The biggest caveat to all of this is that, if you apply automation to the volume, it will be *after* any plug-ins, and they may not have the desired effect. If, for example, you wanted to change the gain of a particular word on a vocal that was causing the compressor plug-in to work too hard, changing the level of the volume with automation wouldn't solve the problem. In cases like this, it can be better to load the Logic Gain plug-in into the first insert and then automate this plug-in rather than the main channel volume, as this will affect the level going into any further processing, which, in most cases, would be the purpose of level control at the editing stage.

FIGURE 5.4
Other than a region gain parameter in the Inspector, Logic doesn't have any additional pre-plug-in level control options, but, if it is useful, using the Marquee tool on the channel volume automation track will automatically create four nodes at the edge of the Marquee selection to allow for quick volume changes to specific parts of a region.

PRO TOOLS

Pro Tools is one of the better-equipped DAWs when it comes to nondestructive level control. Of course there are the usual options for volume automation of the channel fader, but, if we are looking to avoid that for the reasons listed in the main chapter, then Pro Tools has the excellent Clip Gain feature that allows us to do a great deal with the level of the region before any plug-in processing takes place. The first thing that we should clear up is a terminology matter. For consistency with the main chapter, I refer to individual sections of audio as regions, but, from Pro Tools 10 onward, these are now referred to as clips. While the name difference may be misleading, the two are actually identical in the sense of what both terms mean. So when I am referring to *Clip Gain*, you can think of it as *Region Gain*.

There are two ways in which we can use the Clip Gain feature, depending on what exactly it is that we are trying to achieve. The simplest of the two, which is perfectly sufficient if we simply want to adjust the gain up or down for the whole of a region, is to use *Clip Gain Info*. We can enable this by pressing Ctrl[Mac]/Start[PC] + Shift + = or by going to View > Clip > Clip Gain Info. When this is enabled, you will see, at the bottom left of each region, an icon that looks like a small fader along with a numerical gain reader in decibels. Clicking on the small fader icon will bring up a miniature fader next to the icon, and you can drag up and down while holding down the mouse button to change the gain. As you do this, the numerical readout will change, and, as stated, the waveform overview will change to reflect this.

At the end of chapter 3, we looked at ways of zooming in to the waveform overview to get a more detailed look at it, and at the time, we stated that this zoomed-in view was not the same as a change in gain. The unfortunate thing is that, at a glance, there is no easy way to tell if a waveform that looks like it will be clipping is as a result of a zoomed-in view or excessive use of the Clip Gain. Of course the numerical readouts will show if you have applied any gain, but it is distinctly possible that you have applied a small amount of gain that hasn't resulted in clipping, but, because you are still zoomed in, it looks like you have. The best way to avoid situations like this is to always reset any vertical waveform zooming to the minimum level once you have done what you need to do. If you follow this routine, then any subsequent gain changes will be reflected accurately in the waveform overview.

FIGURE 5.5
The Clip Gain Line in Pro Tools allows for extensive level control of audio regions. In addition to the flexibility that this gives, the waveform display also updates in real time, allowing you to get good visual feedback on the changes you are making.

If you want to do something more complex than a simple, static gain change for the whole region, or if you want to change the gain of only a part of a region, then you can use the other method by pressing Ctrl[Mac]/Start[PC] + Shift + – (minus) or by going to View > Clip > Clip Gain Line. This will overlay a line, similar to an automation line, on top of the region. At the beginning of each region, you will notice a small circle on the line, which represents a breakpoint. Each breakpoint defines a level and a position. You can adjust the position of individual breakpoints up and down, and you can create new ones to enable more-complex changes. The specific changes that you can make to these breakpoints depend on the tool you currently have selected.

If you have the Smart tool selected, and you move the cursor close to the Clip Gain Line, you will see the cursor change. You can click and drag the line up and down using this cursor, and it will affect all breakpoints within the region equally. The region doesn't have to be selected for this to work, but, if you do have multiple regions selected, this method will apply the same gain change to all currently selected regions. As you are dragging using this tool you will notice that, at the top of the region, you get a numerical readout that shows you the absolute gain change applied (based on the original level being 0 dB) and the relative gain change that you are applying with this use of the tool. If, for example, you had previously applied a gain change of –2 dB, and you now used this tool again to add a new change of –3 dB, then the display would show –5 dB on the left, as this represents the overall change from the original region, and –3 dB on the right, as this represents the change applied at this time.

Another option with the Smart tool is to move the cursor to the top of the region, so that the Selector tool becomes active, and then click and drag a selection within a region. If you then move the cursor over the Clip Gain Line, you will be able to change the gain for only that section. If there is already a gain change happening within that selection, you will be able to move it up or down, whether it is static of changing, using this method. New breakpoints are created at the selection boundaries, and the gain change is applied only to the selection and won't change anything for the rest of the region.

If you have the Grabber tool selected, and you move the cursor close the Clip Gain Line, then you will see the cursor change to a pointing finger with a small "+" sign. Clicking on the line will create a breakpoint at that position, while clicking and dragging will both create the breakpoint and change the gain value. If there is another breakpoint after the one you are creating in the current region, then raising or lowering this new one will cause a ramp from the new breakpoint to the following one as well as from the preceding one to the new one. But if there is no breakpoint after the one you create, then there will be a ramp only from the preceding one to the new one, and then the gain change will stay constant to the end of the region. If, however, you move the cursor over an existing breakpoint, then you will see only the finger icon without the "+" sign, and if you click and drag, you can change the position (both level and time) at this point.

If you need to make only a minor (and consistent) change to a region, then you can use the key command Ctrl[Mac]/Start[PC] + Shift + up or down arrows. Any regions that are selected when you use this command will ha ve their gain changed (nondestructively, remember) by 0.5 dB. This is the default value and can be changed if you wish by going to Setup > Preferences and then choosing the Editing tab and then changing the *Clip Gain Nudge Value* (located under the Clip heading).

THE SEARCH FOR PERFECTION

Comping, or compiling, is the process of putting together one "master" take from a number of alternate takes, using the best parts from each to create one final version. It is a very widely used technique, but possibly the most common application is vocal comping. Because of that we will focus on using all the things, we have learned so far to comp a vocal. Most of the things we cover will be common to comping all different kinds of instruments and voices, but there will be specific issues for certain types of instruments, and there is additional information on the website about these additional considerations.

But first, why do we bother comping? To answer that, we need to accept the fact that, as human beings, we can't achieve 100% perfection for 100% of the time. Even if the performers are well-rehearsed and know what they are doing and are technically competent and artistically expressive, there are still physical limits placed on us. First, even for the most experienced, a studio environment, especially when the "tape is rolling," increases the pressure to get things right. Some people thrive in this environment, while others fall apart. Furthermore, when you are expected to give it everything you have, you can't be expected to give it for too long. Fatigue sets in eventually, and for instrumentalists this can lead to less accuracy, less expression, less emotion, and there is nothing that can be done about it other than resting and coming back later.

Vocalists in particular aren't governed only by their physical abilities to "play" their instrument but are also very much at the mercy of the fact that their instrument is a part of their own body and can be affected by a number of things that are all beyond their control. Trying to push a vocalist too hard for too long will only be a downward slope. Also, most vocalists will, at some point in their career, have pushed too hard and could well have lost their voice, so they won't be in a hurry to do that again. When you couple their understandable resistance to damaging their voice with the increasing pressure as the day wears on, it is completely understandable if the quality of the performance starts to falter eventually.

So we can't reasonably expect a performance to be 100% perfect from start to finish. That's not to say that it couldn't happen, but the chances are incredibly small. Additionally, even if a singer were to be 100% pitch-perfect, with immaculate timing, expansive dynamics, wonderful delivery, and sublime tone, we still have to realize that each performance is just an interpretation. You could have two takes that were both perfect, but, at the same time, different in some respect (emotion, tone, etc.), and you could clearly prefer one over the other just because of the feeling it gave you.

Sometimes, there is a lot more to a perfect take than just timing and pitching.

This gives rise to an interesting question of what would be preferable to us: 50% perfect 100% of the time or 100% perfect 50% of the time? The answer to this is not only very personal but also it depends largely on circumstance. For example, if we are considering a studio recording environment, where we have the ability to perform all kinds of magic on our recordings, we could make do with either of these scenarios. If the singer was 50% perfect 100% of the time, then we have tools to help get that 50% up quite a lot higher. We can adjust timing, pitching, tone (in terms of matching tone between takes), and dynamics. In fact, the only two things we really have no control over are the basic tone of voice and the emotion/ delivery of the performance. Equally, though, if a singer is 100% perfect but only for 50% of the time, then we have the option to comp things together, to assemble a "best of" take and get that 50% much higher. In other words, we can deal with both scenarios.

Having said that, many would prefer, in a studio environment at least, someone who was closer to the 100% perfect but for a lower percentage of the time. This is purely a practical consideration. We have to consider that comping and perhaps moving a few parts around will have less potential risk of artifacts and other unwanted side effects than if we have to start time-stretching, correcting pitch, and matching up tone between takes. In addition, it is a far easier process. So for those reasons alone, it can be easier to work with somebody who is closer to perfect but for less of the time. Of course there is always a balance and a tipping point. It would be much easier to work with somebody who was 75% perfect for 50% of the time than somebody who was 100% perfect but only 5% of the time.

In summary, comping as a skill in general, and in particular vocal comping, is perhaps one of the greatest assets that you can have as an editor. In a high percentage of modern music, the vocalist is the absolute key feature, and the vocals will be riding high above everything else: loud and proud, front and center. So if you can perfect your vocal edits and comping, and in the end have it sound like the vocals were all recorded in one perfect take, then your skills will undoubtedly be in demand.

PUTTING IT ALL TOGETHER

The idea of comping shouldn't be new to us. Even if you didn't know about it before, we have referred to it several times in the earlier chapters, so it should be starting to become familiar now. We have looked at all the types of skills you would need in order to do successful comps, and now we have arrived at the point where we can put it all together into a very useful and often-needed practical use. What we will do now is to walk through the process. We will touch on technical issues but won't dwell on them overly, as most of the deeper discussion has already been done. Instead we will focus more on the process itself: the decision-making process, the things you need to consider in terms of the choices you make, and tips that can help make the whole process a little more intuitive.

The first thing you need to do is make sure that all the takes are lined up and synchronized. If you have done the recording yourself, then they will most likely already be lined up, but if you have received the files from elsewhere, you may have to do this on your own. With DAW software as widespread as it is today, it is fair to assume that most recordings will be made direct to DAW. If that is indeed the case, then the various takes will probably have been lined up in the original project. That doesn't mean, however, that all the files you receive will necessarily line up when you import them. If you perform a basic import of a file into a DAW, then the import process is essentially a "dumb" one. The software doesn't know the start position of the original file. If you are lucky, the files will all have been trimmed to the same start point, so all you would have to do is import them and line up the start of all the regions, and it will all play back in sync. What can sometimes happen, though, is that the original recording engineer will have cut the start point of the audio region in the Arrange window, so that everything lines up nicely, and will then send the files over to you. Sadly, cutting the front of the region in the Arrange window simply changes the start point of the playback and doesn't actually trim the file itself. This is possible and very easy to do from within your DAW, but it is a step that is often forgotten. So if, for whatever reason, your various takes don't line up nicely, then you have to move them by hand until they do.

Once everything is lined up, the next step is a simple one: listen. I know it might sound ridiculous, but a quick listen through each of the takes in isolation, one at a time, just to check that there is nothing unexpected, is a very good idea at this stage, even if just to familiarize yourself with the song and the sound of the vocalist before you really get involved in anything more complex.

MAKING THE GRADE

At this point it is useful to start to grade the performances. It can be very useful to use a color code here to help you identify things later. You could do this by splitting each take into sections (a bar at a time, a line at a time, potentially even a word at a time, but this is probably overkill at this stage) and then listen to each take, one at a time, with the music playing, and color each region according to your initial opinion. You could use green for regions you thought were good candidates for the final comp, yellow for ones that you thought might be usable, red for ones that you really weren't sure of, and black (or muted) for ones that were totally unsuitable or had mistakes. This initial scan-through and grading process can be very helpful, as it reduces the number of options you have to consider later.

Once this is done for all tracks, it's time to begin the comping in earnest. The concept is simple enough: you listen to each take of a particular line (or word, if it comes to that), and you choose which one you want to include in your final comp. This doesn't necessarily mean the line that is best in isolation, because there has to be a flow through the final comped vocal. Because of this, it is useful to not just loop around one line as you are auditioning the different takes but to loop around not only the line you are concerned with but also the one before and the one after. This contextual listening will help you get a better picture of the flow.

As for actually putting the comp together, there are two main schools of thought. The first is that you listen through all the options for a particular line, and then, once you have chosen which one you want, you mute all the others and leave your choice un-muted. The other option is that you create a new channel/track in your Arrange window and move the line from whichever take you have chosen on to this "master track." Both have exactly the same result, and the choice is really down to how you prefer to work with the master track, perhaps having the advantage of keeping things a little easier to interpret.

If you have, so far, separated only each take-out into lines, then you might, at this point, need to further split the lines into words in order to put together the best take. Once you have done this, or if you already did it earlier, you might still find a need to split a take midword (as we discussed in chapters 4 and 5). If this is necessary, I would make a quick note and come back to it once the rest of comp had been completed. Getting a good cross-fade in the middle of a word can take a bit of work, so it is usually more productive to leave that until the end to avoid interrupting the flow of the comping process and also to avoid your getting that one word stuck in your head too much at this stage. The same is true of any level matching or tonal matching. It is probably best to leave both these things until the end of the comping process to avoid getting caught up in the details and losing the feel of the vocal performance.

FIGURE 7.1
When you have multiple takes, it is often useful to either name them or color them based on how good they were. The final comp will most likely include sections from a number of different takes, but an initial grading of takes is often helpful in speeding up the process.

FIGURE 7.2
The traditional method of DAW comping involved cutting out sections from a number of different tracks and manually moving them to a main "comp track". While the basic principles are still the same, all the DAWs featured in this book have developed methods to make this process as automated and intuitive as possible.

If you have gone through all the options for a particular line, and even the best one needs a cross-fade or level matching or tonal matching, and if the difference is enough for it to be really obvious and off-putting, then it might be wise to at least roughly try to remedy the problem. Doing a quick cross-fade, even if it's a little bumpy, or a rough guess at a level match or a quick EQ job, might be enough to make the edit stand out a little less while you finish up the rest of the comping. The whole process can be quite time-consuming and more than a little difficult, as the differences can be very subtle at times, and there is no clear right and wrong: it is often a purely artistic decision (once obviously poor options have been removed). Once you reach the end, it is advisable to take a break, as you will be moving into a different mind-set for the next stage. Having been working with broad strokes, you now have to start filling in the detail, and a break and resting of your ears would be a good choice.

TONAL MATCHING

The obvious thing to do next would be to take care of any midword cross-fades and level matching, but to do this you may need to take care of the tonal matching first. It is important to remember that you need to tone-match any given region to the regions on either side of it. It is hoped that the regions on either side will be fairly consistent with each other, so that whatever EQ changes you make will blend smoothly on both sides. If this isn't the case, then you will need to make additional changes to one of the other regions as well. If you find yourself in a situation where you have too much tonal correction to do, or where you have a number of different takes and each one is tonally different from all the others, then you probably have bigger problems than a simple vocal comp. If there are too many different tones to the takes, then you can either change your choice of take for each section and try to use more parts from each take and (consequently) less different takes, or, ideally, but not especially practical, you can see if it is possible to get some additional recording done to try to get the job done in fewer takes.

When you have matched up the regions from the different takes, you have a couple of options. If the matched region is on a different track, you can't apply a cross-fade as such. Of course, you can fade-out the region on one track and do a corresponding fade-in on the other track, and, if the parameters were comparable, there shouldn't be any difference in sound. It isn't quite as convenient, though, so there is always the option to bounce the tone-matched region to a new file and then drop that new file back on to the main "master take" track and apply the cross-fades there. Which option you choose comes down to the question of convenience vs. flexibility. Keeping the tone-matched region on its own track will allow you to make further adjustments to the EQ of that region individually, but the transition between regions might take a little longer to get right. Try both, and see which works better for you.

At this point, we have reached a very important milestone in the creation of our vocal comp. By now we should have something that is sounding pretty good and quite fluid and natural. There might still be some more we can do to get it as close to perfect as we can, but we will have definitely come a long way. Now you have a choice to make. You can either push ahead and move on to the next stage of polishing, or, if you have the time and the inclination, you can go through the whole process again. The reason for doing it all again is to create another version of the comp using the best of the remaining options to serve as a "double track" vocal to thicken up the lead vocal.

The next thing two things we need to work on are the pitching and timing of the vocal.

PITCH ADJUSTMENTS

Vocal tuning is a very difficult subject. Technically, it is relatively straightforward, but artistically it is a little more complicated. And it is further complicated if we decide to incorporate it into the audio editing process. I mentioned right at the beginning of this book that there are many areas where editing and mixing or production overlap, and this is definitely one of the big ones. If the editing is for a project of your own, then working on pitch correction during the editing stage certainly won't step on any toes, and if you have a backing track already that the vocals were recorded to, then you have a frame of reference to use when tuning. If, on the other hand, your editing work will be passed on to somebody else for mixing and production, then, unless it has been specifically requested, I would avoid doing any kind of pitch correction at this stage—with one small exception.

The only time I would consider pitch correction to definitely be an editing task is to match up the pitches of cross-faded sections. Any need for this would have become apparent during the process of doing any final cross-fades between regions. If the difference is subtle, and if the cross-fade it long enough, then you will probably be OK, as it will sound like a gentle and natural bit of pitch drift, which can be cleaned up during production and mixing if needed. If the difference is large enough, it will need fixing, or you might even need to choose a different take. But let's proceed on the assumption that you had a very good reason for choosing that particular take at that point and need to deal with the pitch difference.

The first thing we need to establish is whether the note is out of tune overall or whether it just drifts out over its duration. If the note is out of tune overall, then it is a fairly straightforward process to correct it. Most DAWs have some sort of included pitch-shifting plug-in, so we can quickly create a new track, move the out-of-tune region onto that track, load up a pitch-shifter plug-in, and then adjust the "fine tune" control until the note is and tune and we are done—right? Actually, I would say no. As good as pitch-shifting plug-ins are, any that work in real time simply won't be of the best possible quality. And audio editing is, after all, about getting the best possible version of the files we are editing! So what we can do is use this pitch-shifting plug-in to figure out how much we need to shift by (in cents), and then, once we have that figure, we can create a new file from this region and return it to its original place on the master-take track and then go into the wave editor in our DAW (which should have a pitch-shifting option). Make sure that only the region we want is selected, and then process this region with the pitch-shifting option. Because this pitch-shifting isn't working in real time, we should get a much better result.

Notes that drift in to or out of tune will require a little more effort, and, potentially, some external software. It may not be the best idea to even consider using any kind of pitch-correction plug-in for a task like this, because, as we have already said, at this stage, we aren't trying to make the tuning more accurate. All we are trying to do is match it up to another sound, so, unless that other sound happened to be perfectly tuned, automatic pitch-correction won't work. What we need is the ability to just change the part that is out of tune with the "reference" part, irrespective of whether that part is perfectly in tune. The two most commonly used tools for this are Antares Auto-Tune and Celemony Melodyne. Both are available as plug-ins, and both are capable of doing exactly what we are looking for (and much more).

Both of them work in a fundamentally similar fashion, in that you have to record the audio in to the plug-in, which then analyzes it before you can perform any corrections. Because the audio has been pre-analyzed, we have a much higher quality than a typical pitch-correction system, and yet we have the ability to make changes to the audio as it is playing. It really is the best of both worlds. Once the audio has been analyzed, we will see an overview of what we are dealing with. Both plug-ins use a similar system—which has some of the attributes of a typical piano roll editor in that each note in the audio is presented on a grid, whereby its vertical position tells us its pitch (to the nearest note) and its horizontal position its time—but it is different in that, instead of the rectangular blocks shown in a piano roll editor, here we have the actual waveform shape shown to us. On top of this, there is usually a line overlaid that will show the exact pitch and any fluctuations from the nearest "perfect" note. This line is what we can use to determine what we need to fix, without worrying about its actual tuning. From here, we can use either of them to manually adjust the pitch of both parts of the comp to bring them into line with each other (although not necessarily perfectly in tune) and, with a bit of luck, end up with something that sounds very natural.

Once any level difference, tonal differences, and tuning differences have been dealt with and any cross-fades applied, it is time to move on to any timing adjustments that are needed. Strictly speaking, when talking about timing adjustments, we are referring to the timing of the start of words, but, given that there will be times when a move of the start point will leave a wrong or uncomfortable-sounding gap, we will also discuss about time-stretching, as the two often go hand in hand during comping.

TIMING CHANGES

With luck, any timing changes that you have to make should be pretty subtle, and as such should be done by ear instead of looking at the position of the waveform relative to the beat grid on the Arrange window. Aligning the start of the audio to the grid is very useful to get things roughly into position if they have to be moved a lot, but the final careful positioning should always be done by ear. Each region should be moved individually, except if you have cross-faded regions, in which case all the regions that have cross-fades should be selected simultaneously and moved by the same amount in order to preserve the cross-fades. There will be times, however, when moving things around isn't enough. This may lead to overlapping regions, or pauses/gaps that are too long, or a number of other reasons why we might want to time-stretch instead of (or in addition to) moving things around.

Before we start, though, let's remember one important thing. In the case of any "lead" instrument, and voice especially, time-stretching should be avoided wherever possible. Although the technology has improved greatly over the years, there are often still audible artifacts to time-stretching that we should try to avoid if possible. So if there is a gap at the end of the region, you should ask yourself if it really is too long or if you are wanting to time-stretch it just to keep the end of the region where it was before. If there genuinely is a need, then time-stretching might be an option.

If, on the other hand, we are dealing with a region that now overlaps with another, then we may have another option. As long as the overlap isn't too long, we can consider using a cross-fade to just gently taper off the end of one word and allow it to flow into the next. This is particularly suitable if the end of one word only slightly overlaps into a following word. But if there is a breath sound prior to the next word, then having the preceding word cross-fade into a breath doesn't sound natural at all. Instead we could try shortening the now too-long region to bring the end of it back and allow enough of a gap before the breath sound. We can then apply a fade-out to the end of the region to make sure that the preceding word tapers off naturally before the breath. The choice of whether to use the time-stretching method, the fade, or the cross-fade methods depends largely on the situation, and it is difficult to know which will be more suitable. It is perhaps easiest to try the fade or cross-fade methods first, as those are probably quicker and don't have the artifacts associated with time-stretching, but if they don't work, there is always the time-stretching method to fall back on.

FIGURE 7.3
Sometimes when you are putting a comp together, you many find that you have gaps or overlaps to deal with. The top track above shows an example of this, which was then cleaned up by adding fades and cross-fades and time-stretching a few parts. The result is a much smoother and more flowing final comp.

There is one final point of note to do with time-stretching that I would like to make, which relates to the choice of exactly what we should time-stretch. Any time-stretching should take into account the preservation of any attack portion of the sound (for transients and subtle pitch variations or bends) along with factoring in the possibility of delayed vibrato, which could sound unnatural if stretched or compressed too much.

As you can see, the process of comping isn't an inherently difficult one (from a technical perspective, at least) but it can be quite a long-winded one. If only there was a way to streamline even a part of the process to help make it more intuitive and a little less "clunky"...

COMPING MADE EASY

In late 2007, Apple introduced Logic Pro 8, and with it they introduced a new feature called Quick Swipe Comping. The idea was a fundamentally simple one: instead of having to manually cut each take into separate parts/lines/words, you could simply choose which take you wanted to use by selecting it with a "swipe" (a click and drag). Only one take could be selected for any given part, so swiping over a line on one take would automatically replace whichever take had been used before. Cross-fades would be applied at the edit points, and you could set up a default cross-fade time and shape as part of your DAW preferences, but this would be applied globally to all edits in the comp. Once you were happy with the result, you could either bounce it to a new file, create a new track with all of the comped regions and still leave the comp track intact, or just remove the comp track, leaving only the comped regions.

Perhaps you might see this as only evolutionary, but for many, it was revolutionary and changed the way we thought about comping. What was previously seen as a hugely time-consuming (but often needed) task had just become orders of magnitude easier. It wasn't perfect in either concept or execution (it was quite "buggy" in the earliest versions), and there were many areas that could, and later would, be improved. Yet in spite of this, it really was a very welcome addition to the Logic toolbox. And because of this, other DAW manufacturers were quick to follow with their own tools to improve the comping work flow.

Pro Tools added a similar (although some might feel slightly less user-friendly and immediate) system to Pro Tools 8 in 2008, while Steinberg's approach didn't really develop fully until Cubase 6 was released in 2011 (even though they had been developing their approach since Cubase 4). MOTU added their own version in Digital Performer 6 in 2008, and PreSonus greatly improved their comping tools in Studio One version 2 in 2011. With this flurry of activity to introduce similar features into other DAWs, Apple quietly improved, both in terms of features and reliability, their own version in Logic Pro 9, and, over the last few years, many people have grown accustomed to this new paradigm. It is true that it doesn't allow us to do anything inherently new, but for people who work in any industry day after day, work-flow improvements that allow them to complete their work more quickly (as long as the quality doesn't suffer) can often be just as welcome and important as completely new features.

While this can be a great way of quickly putting together a comp and keeping everything nicely organized, as well as giving you the option to easily store and switch between alternate comps (depending on which implementation you are using), it isn't all perfect. You still need to carry out the rest of the steps afterward, and doing it this way may not automatically take into account any timing differences between the takes. It may be that the tone and delivery on take three sounds great, but it was a little too late. Some DAWs will allow you to nudge the timing of individual parts of the comp, while others won't. But what none of them will do is automatically align the timing of different takes.

So that's pretty much it for general (including vocal) comping. We have covered all the common comping techniques, traditional and new, and, and, although the examples we have considered have been focused on vocal comps, they are still applicable to comping any instruments. As stated at the beginning of the chapter, some instruments' groups have other factors to consider, so please take a look at the website for more information about these.

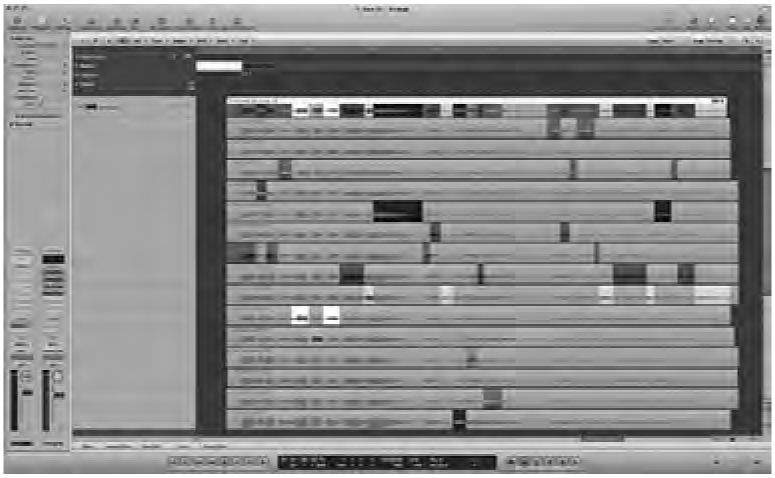


FIGURE 7.4
Swipe comping, as shown here in Logic, enables you to carry out the comping process without having to cut regions and move them manually onto another track. It also allows you to create a number of different alternative comps that can be switched easily, enabling you to try different ideas with a minimum of fuss.

HANDS ON

INTRODUCTION

Comping is one of the most-often carried-out editing tasks in the world of music recording and production. Its use is not as extensive in music for picture and film scenarios, but in any case, it is a prerequisite skill for any editor to have, as it can be seen in many ways as the sum of all the corrective-editing processes. It was traditionally quite a time-consuming task, but the most recent versions of most major DAWs have introduced tools to make the editing process not only quicker but also much more intuitive and flexible. These new tools could quite easily reduce the time taken to carry out a typical comping job by 50% or more. As such, a good look at how we use them is almost essential.

LOGIC

The biggest tool in Logic (and many of the other modern DAWs) for comping is its so-called *Quick Swipe Comping*, which is a very quick and easy way of carrying out comping. Instead of having to manually cut regions from each of the takes and mute them as you decide that they won't make the final comp, and instead of having no simple way of comparing alternate comps, this process gives a huge amount of flexibility and makes the entire process seem far less daunting.

Crucial to the whole idea of Quick Swipe Comping is the existence of *take folders*. These are a simple way of grouping multiple takes of a given part, which, aside from the obvious benefits that we will see shortly, is also a very good way of keeping your Arrange window organized and tidy. If you are recording audio into Logic and have set up a loop of a particular section (a verse, for example) to allow the performer to repeat the section a number of times, then a take folder will be automatically be created for all takes recorded continuously. If recording is stopped and then restarted for exactly the same loop range, then new takes will be automatically added to the take folder, while, if a totally new range is selected, then a new take folder will be created. However, if a recording range is selected that overlaps an existing take folder, then the new takes will be added to an existing folder and the loop range of that folder extended.

On the other hand, if you are given a number of separate takes from a third party that aren't already grouped into folders, all is not lost. In order to be able to carry out Swipe Comping, you will need to get the individual takes into a take folder, and fortunately this is a very easy process. All you need to do is select all the regions that you wish to put into a take folder (either by clicking and dragging or by holding down Shift while clicking on the individual regions) and then right-clicking on any of the selected regions and choosing *Pack Take Folder* from the Folder contextual menu. Alternatively, you can go to the menus at the top of the Arrange window and choose Region > Folder > Pack Take Folder, or, if you prefer key commands, the same thing can be accomplished by pressing Ctrl + Cmd + F. If the take folder was created during recording, then by default all the takes will be visible, while creating one from separate regions will have the individual takes hidden. To change from one view to the other, you simply need to click on the disclosure triangle located at the top left of the take folder.

Once you have your take folder, you can start the actual comping process. The first thing you need to do is make sure that Quick Swipe Comping Mode is enabled. This is done simply by clicking on the Quick Swipe Comping Mode button at the top right of the take folder. This button looks like a rectangle with three small horizontal bars inside. If the rectangle looks transparent, then Quick Swipe Comping Mode is disabled, and if the rectangle background looks solid, then Quick Swipe Comping Mode is enabled. With Quick Swipe Comping Mode enabled, you can begin to make decisions about which parts of which take to use.

The take that is active (being used) will be colored, while the unused takes (or parts of take) will be grayed out. To change which take is being used, you simply move the cursor over one of the inactive takes (the cursor will change to two vertical bars) and then click and drag (or swipe) over the section of the inactive take that you would like to use. At this point, the swiped section (take region) will become colored, and the corresponding part of the previously active take will be grayed out. Fundamentally that is all there is to it. It really is quick.

Of course there is a little more to it than that, as we could well need to fine-tune the sections that we have chosen, and there are two main ways to do this. The first method is to move the cursor over either the left or right edge of a currently active take region, at which point the cursor changes and adds left and right arrows on either side of the two vertical bars. This tool allows us to move either end of a take region. Doing this not only changes the start or end position of the currently active take region but also correspondingly lengthens or shortens the adjacent part of the preceding or following take region. The other option that we have is to move the entire take region. If we position the cursor in the middle of a take region, it will change to just left and right arrows, and clicking and dragging with this tool will keep the overall length of the take region the same but will move it earlier or later. In combination, these two tools allow you to quickly fine-tune the initial swipe-comping that you have done.

FIGURE 7.5
A take folder in Logic expanded to show the different takes and the sections of each take that go into making up the final comp (on the very top track).

If for any reason one of your takes is not aligned with the others properly, then you can move it within the take folder, but to do so you need to turn off Quick Swipe Comping Mode, as described above. Once it is turned off, clicking and dragging on an individual region follows a more normal behavior, where it simply moves the region either earlier or later. You should note that any take regions on the region you are moving will stay in the same place relative to the region and not to the take folder, which could mean that a movement of this kind could mess up any comping you have done so far. As such, it is probably wisest to make sure all takes are aligned using this method before embarking on the comping process. If you find that only a single part of one take is out of time, then you can split that take and move only the required part to bring it into line.

Once you are happy with your comp, you have a number of choices. You can either leave things as they are and move on to the next task, or, if you want a little more control over things, you can export the comp to a new audio track. If you click on the small arrow at the top right-hand corner of the take folder, it will bring up the Take Folder menu, and from this menu, you can choose Export Active Comp to New Track . This will create a new audio track and copy all active take regions into their correct positions on the new track and apply crossfade between the regions. This then allows you to further work on cross-fades and other editing tasks on the individual regions if required. Once this is done, you can mute the audio track containing the take folder, and you are ready to move on.

Alternatively, if you are completely finished with the comping, then you can choose Flatten from the Take Folder menu, which carries out the same process, only it will move the take regions into position on the same track as the take folder and delete the folder, leaving you with just the comp. The only real difference here is a work-flow one. If you are 100% sure that you do not wish to make any more changes, then flattening is the better option, to minimize clutter. However, unless you are 100% sure, I would always choose the Export option, as it allows you to go back in and try something different if you need to.

And speaking of trying something different, the Take Folders allow you to have a number of options you can quickly choose between by being able to store different versions of comps within the folder. Once you are happy with your first comp, you can export it, and then, in the Take Folder menu, choose *Duplicate Comp*. This will create an exact copy of the current comp, which, in the menu, will show up as Comp 2. If you now make changes to your comp, perhaps trying different lines from different takes, those changes will be stored within Comp 2. If you open the Take Folder menu and switch back to Comp 1, you will see that your original comp remains untouched. To make matters easier, you can use *Rename Comp* from the Take Folder menu if you would rather have a reference to each version by name (Master, Double Track, Alternate Lead, etc.). Once you are done, you can again export this new comp to a new track.

These alternate comps make it very easy to create alternate takes, and the fact that you can export each one once you have finished it while still keeping it as a comp within the take folder means that very little has to be committed to that can't easily be changed later. Whether or not this is a good thing depends on how you like to work, but, whatever your views, having options during this often-critical process can only make life easier. Once you are done with all the comping for this take folder you can, of course, delete the whole take folder, but, unless you have a very compelling reason to, I would always suggest muting it and leaving it in your arrangement—just in case.

PRO TOOLS

Comping in Pro Tools is made much easier by the use of *playlists* on tracks. The most simple example of a playlist in Pro Tools is a single region on a single track that starts at a certain point and plays for a certain amount of time. If you add in different audio files or regions, or you split regions and move them around, then the playlist will become more complicated, but it is still counted as a single playlist. Where this functionality becomes much more useful to us for comping is the fact that each track can have multiple playlists stored. Only one can be active at any time for any given track, but multiple alternatives can be stored. This idea could be used to store alternative arrangements of the same regions in different playlists, perhaps to allow different song arrangement ideas to be tried, or it could be used to store different versions of a particular recording—in other words, different takes.

Now the reality is that if you had five alternate playlists with a different take on each, it isn't a massive step away from having five different tracks with a single take on each, but the real benefits come from the work-flow improvements. If you had five different tracks, then you would have to remember to make sure that only one was playing at any given time, and you would also need to duplicate any plug-ins that were being used across all five tracks, and this could potentially place a significant extra load on your computer. And then, if you wanted to used those five tracks to create a comped part, you would probably need to create a sixth track to put the comp together on. You would have to make sure that, as you were moving or copying each section of the comp onto the "master" track, it was copied in the same position (in the time sense) as the take that it came from and, not insignificantly, it would create a massive amount of clutter if you had multiple parts to comp.

Using playlists, on the other hand, keeps things organized and keeps clutter to a minimum by allowing you to easily hide and show the alternate playlists for a given track. It also allows you to quickly switch between alternate playlists if you want to try different things out but be able to get back to where you started quickly, and it also provides a very quick and simple way of copying sections from individual playlists onto another playlist, so that you can put your comp together with as little fuss as possible and as few worries about timing consistency as possible.

If you are going to be recording into Pro Tools yourself, then there is a very simple option that you can set that could greatly speed up the early stages of the comping process. If you go to Setup > Preferences and then go to the Operation tab, you will see, under the Record heading, an option to *Automatically Create New Playlists When Loop Recording*. Turning this option on will mean that any recording done in loop mode will group each pass of the loop (or take) onto the same audio track and will create a playlist for each take. When you do this, the most recent take will be the uppermost; and the earliest take, the lowermost. Recording tracks in this way removes one step of the comping process. Of course there will be times when you simply record straight through a song from start to finish and then, perhaps after making some adjustments, start recording again. In this case, or if you have been given a collection of individual files from a client, you need to do a little bit of work before you are ready to actually start comping.

The simplest way to do this is to create a new audio track by pressing Cmd[Mac]/ Ctrl[PC] + Shift + N or choosing Track > New to bring up the *New Tracks* dialogue. Here you can choose the number of tracks you wish to create along with various options about the type of track you wish to create. Choose *Mono* or *Stereo* as appropriate, make sure *Audio Track* is the selected option, and hit *Create*. The audio track will be created and you can rename it to something meaningful ("Vocal Comp A," for example) by double-clicking on the track name and entering the new name in the dialogue that appears. This track will now function as a container for all the different takes that we have and will also be the eventual location of our comped track.

The next thing to do is to click on the *Track View Selector* (located on the track header underneath the *Solo* and *Mute* buttons) and change the default Waveform to Playlists. As you do this, you will notice that a smaller sub-track appears underneath our newly created track, and this is where we can start to put together our alternate takes. If you already have all the alternate takes in the main arrangement (already synchronized to each other), all you have to do to make them available for our comping process is to drag and drop them onto the smaller track. This will move the region and create a playlist for that region, the track will become larger, and another small sub-track will appear beneath that one. Once again you can double-click on the track name to rename it ("Lead Vocal Take 1," for example), and this is always a good idea, just so that you know which track represents which take at a glance. You can then drag in as many takes as you need, one by one, into the smaller subtracks, and new playlists will be created for each one. Once this is done, we will be at the same stage that we would have been had we recorded the audio into Pro Tools in loop mode, so we can now move on to the actual comping process.

The actual process of comping itself is very easy. You can listen to any individual take (playlist) by clicking on the Solo button on the individual take track and hitting Play. The rest of the tracks in the song will play along with only the soloed take. This allows you to preview each take in context of the song. If no individual take is soloed, then you will hear what is on the main comp track. If you have not yet selected anything to be placed on this track, then you won't hear anything. Once you have auditioned each track and are ready to start putting the comp together, you have a couple of ways of doing it. If you want to move a whole region from any particular take onto the comp track, then you just need to select that region (either with the Grabber tool or the Grabber part of the Smart tool) and then either press Ctrl[Mac]/Start[PC] + Alt + V or press the up arrow located next to the Solo button for that particular take. Doing either of these will move this region to the main comp track/playlist. Alternatively, if you wish to move only a part of a region, you can select it using either the Selector tool or the Selector part of the Smart tool, and then once again either pressing Ctrl[Mac]/Start[PC] + Alt + V or pressing the up arrow located next to the Solo button to move the selection to the main comp track/playlist.

You should note that if you choose a region or selection that overlaps a region or selection that is already on the main comp track, then this new selection that you move will overwrite the existing area of the main comp track. The latest file to be moved to the main comp track is always the one that takes priority. Once the regions or selections have been copied to the main comp track, you can use the Trimmer tool to fine-tune the start or end points of each section and then apply cross-fade to smooth things off if required.

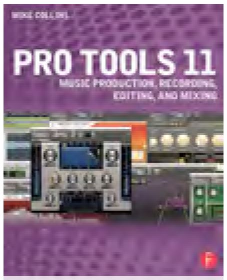
If you wish to create an alternate comp using the same takes, then you can start either by duplicating the existing comp or by creating a new, empty one. To duplicate the existing comp/playlist, you should click on the down arrow to the right of the track name and choose Duplicate from the pop-up and then name the new comp/playlist. If you wish to create a new comp/playlist, choose New instead. Whichever of these options you choose, the newly created comp/ playlist will appear on the main track, and the previous comp/playlist will be moved down to join your individual takes. Now you can repeat the process and create as many alternate comps as you require. You can switch between which comp is active by clicking on the down arrow next to the main track name and choosing one of the other comp tracks from the list. Choosing any other comp moves the previous active one down into the area with the original takes.

FIGURE 7.6
While the naming might be a little different and the mechanics slightly different, the Playlist functionality in Pro Tools brings quick and efficient comping and the ability to have multiple versions of comps stored for quick access.

Once you have completed all your comping, you can either leave the track as it is and hide the alternate playlists by changing the Track View Selector back to Waveform or, if you prefer or have multiple alternate comps, you can move each comp to its own new track by creating new audio tracks and simply dragging and dropping the regions from each comp track onto their own new track.

FOR MORE INFORMATION ON THIS TOPIC, PLEASE CHECK OUT THE FOLLOWING CREATIVELIVE VIDEO COURSE:

Check out Intro to Logic Pro X HERE.



The following is excerpted from *Pro Tools 11* by Mike Collins.

You can purchase a copy HERE.

INTRODUCTION

Getting started with any software environment involves something of a learning curve and, if you are new to Pro Tools, you will find that this is no exception. Even if you have just upgraded from an earlier version, there may be a number of new features that you have not encountered previously. And if you are 'cross-grading' from another platform such as Apple Logic, MOTU Digital Performer, or Steinberg Cubase or Nuendo, you may be looking for features or trying to use keyboard commands that you are used to and wondering where these are in Pro Tools. So boot up your Pro Tools system and spend as much time as possible working 'hands-on' with the software until you know it inside out! Oh, and reading this book will help…

GETTING STARTED

If you are using Pro Tools for the first time, you will need to spend some time familiarizing yourself with the user interface, or with its more recent features if you are upgrading. You will also probably find it helpful to configure the software to suit the ways you like to work. You should take some time out to practice using the system in a non-critical situation first (without impatient clients breathing down your neck) and take the trouble to learn at least a basic set of keyboard commands so that you don't have to use the mouse and menus all the time. You should also learn how to quickly find your way around whichever piece of music you are working on – zooming in and out and navigating along the timeline until you feel comfortable with all this.

HELP

Avid provides excellent manuals for Pro Tools as Adobe Acrobat .pdf document files that you can access from the Help menu. When you open the Pro Tools Reference Manual, for example, the Acrobat application launches and the document file opens into a new window on your computer. You should resize this window and position it in a convenient place on your screen.

It can be very useful to keep the relevant manual open on your computer, but hidden until you need it. You can use the standard menu command or the keyboard Command-H on the Mac to hide the Acrobat application. When you want to reveal the manual again, you can always select it again from the Help menu or use the Show All menu command. One of the quickest ways to do this is to press and hold the Command key, then repeatedly press the Tab key until you see the application you want displayed in the middle of the screen. When the Tab has moved you along the list of open applications to Acrobat, let go of the Command key and this will be shown and brought to the front. Windows users will have their own preferred ways of doing these things.

STARTING THE ENGINES

As with any complex piece of machinery, there are various settings that you may need to adjust each time you want to use it. This is certainly the case with Pro Tools. I recommend that you always check the Playback Engine settings and the I/O Setup at the start of any new session. You may also need to make some changes to the Hardware Setup if you want to hook up additional hardware.

AVID AUDIO ENGINE

If you are using Avid hardware, Pro Tools 11 will use the Avid Audio Engine. This Audio Engine is a real-time operating system for digital audio recording, playback, and processing designed for use with Pro Tools and Avid audio hardware that is automatically installed on your system when you install Pro Tools. If you are using hardware made by another manufacturer, Pro Tools 11 will use Core Audio on the Mac or Audio Stream Input/Output (ASIO) on Windows.

Apple's Core Audio provides the audio connections between software applications, such as Pro Tools 11, Digital Performer, Cubase, Nuendo or Logic, and any audio hardware installed on a computer that uses Mac OS X.

Similarly, Steinberg's ASIO provides the audio connections between software applications such as Pro Tools 11, Digital Performer, Cubase or Nuendo, and audio hardware installed on a computer that uses Windows.

NOTE

Pro Tools software can only record up to 32 input channels of audio or play back up to 32 output channels of audio, when using audio interfaces with the Avid Audio Engine, or supported Core Audio (Mac), or ASIO (Windows) drivers. So if you need more channels for I/O (Input and Output), you will have to use Pro Tools HD software with Avid HDX or Native hardware.

AVID VIDEO ENGINE

Using Pro Tools Avid's Video Engine works with QuickTime video, so you can use any video that you have available in this format. The Video Engine also works with a wide range of Avid HD and SD MXF video formats on Pro Tools video tracks without the need for transcoding these fi rst, including Avid DNxHD. It also lets you monitor Avid HD and SD MXF, and QuickTime media using Avid Nitris DX, Mojo DX, and other Avid qualifi ed third-party video interfaces.

> **NOTE**
>
> If you want to use video in your Pro Tools session, then you will need to tick the box in the Playback Engine dialog to enable the Video Engine – otherwise, the video track(s) in Pro Tools will not work.

THE PLAYBACK ENGINE DIALOG

In the Playback Engine dialog, Pro Tools provides a pop-up selector that allows you to choose the audio 'Playback Engine' for use with your audio interfaces. The available options will depend on which audio interfaces are connected and have compatible drivers installed.

Changing the Playback Engine can be useful if you have two or more audio interfaces connected to your computer with different routing configurations in your studio or if you want to prepare a session for use with a specific interface on a different system (e.g. you might want to prepare a session created on your Avid HDX system for use with the built-in audio on your Mac laptop).

On the Mac, for example, there will always be a 'Built-in' audio interface, and there may be another if you are using an Apple display monitor that has audio input and output capabilities. On Mac systems, you can also select the Pro Tools Aggregate I/O option, which lets you use a combination of built-in inputs and outputs at the same time – see **Figure 2.1**.

You can configure the I/O options for Pro Tools Aggregate I/O using the Audio Devices window from the Mac's Audio MIDI Setup utility software – choosing an appropriate combination of inputs and outputs – see **Figure 2.2**. These inputs and outputs will become available in the I/O Setup dialog if you open this and click on the Default buttons in the Insert, Input, and Output tabs.

> **TIP**
>
> You can open the Audio MIDI Setup utility by choosing the MIDI Studio sub-menu item available from the MIDI menu item in the Setup menu in Pro Tools. This opens the MIDI window by default, but then you can open the Audio window from the Window menu in the Setup utility.

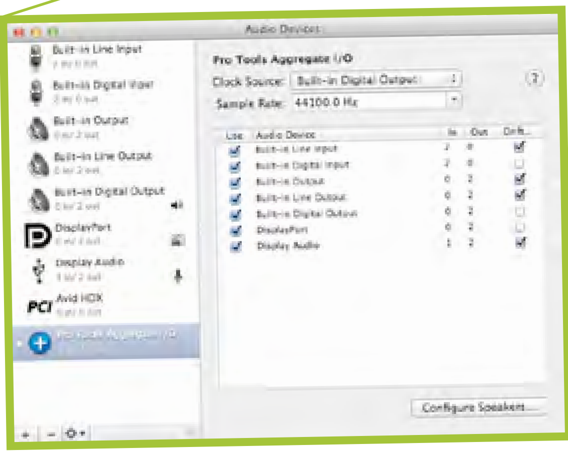


FIGURE 2.2
Configuring Pro Tools Aggregate I/O in the Audio Devices window.

FIGURE 2.1
Choosing Pro Tools Aggregate I/O in the Playback Engine on the Mac.

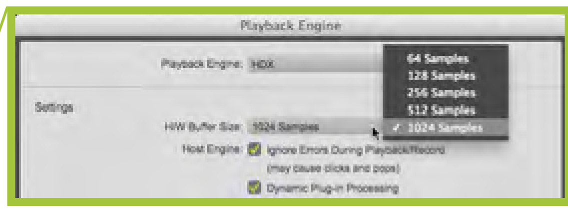


FIGURE 2.3
Hardware Buffer Size and Host Engine settings.

The H/W (Hardware) Buffer Size pop-up selector in the Playback Engine dialog – see Figure 2.3 – lets you choose the size of the buffer used to handle host processing tasks such as processing with host-based, or "Native" plug-ins.

On all Pro Tools systems, lower settings reduce MIDI-to-audio latency – for example, the delay between playing notes on your MIDI keyboard and hearing the audio response from a virtual instrument. Lower settings can also improve screen response and the accuracy of plug-in and mute automation data.

If you are using lots of 'Native' plug-ins, you should choose a higher buffer size to allow for the greater amounts of audio processing required.

The Host Engine settings provide options for error suppression during playback and recording and the option to use dynamic plug-in processing.

There may be times when it makes sense to tick the box to Ignore Errors During Playback/Record. For example, when you are working with several instrument plug-ins and these are stretching the capabilities of your computer's CPU to the point where you are hearing clicks and pops in the audio. In this case, you may choose to work with reduced audio quality, accepting that there are clicks and pops in the audio while you are trying out arrangement ideas. Later, when you want to make sure that you are getting the highest possible audio quality, you can disable this option.

TIP

If you are using a slower computer, you may want to disable the Minimize Additional I/O Latency option to avoid adverse performance.

Ignoring errors requires at least 128 samples of additional buffering on some systems. Host-based Pro Tools systems have an option to Minimize Additional I/O Latency. Enabling this option restricts any additional latency due to ignoring errors during playback and recording to 128 samples. With this option disabled, the buffer used for error suppression will be at least 128 samples or half the H/W Buffer Size – whichever is greater.

NOTE

On host-based Pro Tools systems, lower settings reduce all input-to-output monitoring latency on any record-armed tracks or Auxiliary Input tracks with 'live' inputs. On Avid HDX systems, lower settings reduce the monitoring latency that occurs on tracks that have one or more Native plug-ins. Lower settings can also improve the accuracy of MIDI track timing on systems without a MIDI interface that supports time stamping and on tracks using MIDI virtual instruments that do not support time stamping.

The Dynamic Plug-In Processing option maximizes plug-in counts by dynamically reallocating host-based processing resources as needed, so plugins only use CPU cycles when they are actually processing audio. Normally, you will want to make sure that this option is enabled.

NOTE

The Minimize Additional I/O Latency option is only available if the Ignore Errors During Playback/Record option is enabled and the Pro Tools system you are using requires additional buffering for error suppression, as is the case with the following: Mbox Pro and Mbox 2 Pro on Windows and the Mbox family devices, Digi 002 and 003 devices, Eleven Rack, and Pro Tools Aggregate I/O on the Mac.

HARDWARE SETUP

The Hardware Setup dialog (see **Figure 2.7**) lets you make various settings for your hardware interface or interfaces. On most systems, you will make these settings when you install your Pro Tools system and then leave them alone from that point onwards.

If you are using a third-generation Mbox family audio interface or a third-party Core Audio (Mac) or ASIO (Windows) compatible audio interface, a Launch Control Panel button is provided to launch the control panel for your particular audio interface to configure its settings.

If you are using an Mbox family or a third-party audio interface, there will be either a Launch Control Panel button or a Launch Setup App button (depending on which audio interface you are using) in the Hardware Setup dialog.

When you open this control panel or setup application, you can change the mixer, output, and hardware settings, including the sample rate, hardware buffer size, and sync source.

TIP

You can set the sample rate when creating a new Pro Tools session by selecting a different sample rate in the New Session dialog.

FIGURE 2.7
Hardware Setup with a Focusrite Saffire third-party audio interface.

NOTE

If you have multiple audio interfaces of the same type connected to your system, make sure that you choose the appropriate interface in the Peripherals list when you define its inputs and outputs in the Hardware Setup.

NOTE

On Mac systems using Core Audio, you can select Pro Tools Aggregate I/O as the Current Engine to use the built-in audio inputs and outputs on your Mac computer. You can configure the Pro Tools Aggregate I/O setting in the Mac Audio Setup, which can be accessed from the Pro Tools Hardware Setup dialog. The Pro Tools Aggregate I/O device is intended for use only with the built-in audio on your Mac computer. For best performance, use the default settings.

NOTE

You can also configure similar settings in the Hardware Setup dialog if you are using Avid 003 or 002 family interfaces.

THE I/O SETUP WINDOW

The I/O Setup window lets you label, format, and assign Pro Tools input, output, insert, and bus audio signal paths for individual sessions and for your specific Pro Tools system. Input, Output, and Insert pages display a graphical representation of the signal routing (the Channel Grid) for physical input and output paths for each connected audio interface with controls that let you route physical inputs and outputs on audio interfaces to Pro Tools input and output channels. Controls for PRE (Mic Preamp) signal paths and Delay Compensation settings for hardware inserts are provided on additional pages within the I/O Setup window.

NOTE

When you are working in the Mix and Edit windows, signals are routed to and from tracks, sends, and inserts using track Input, Output, Insert, and Send selectors. When you click any of these selectors, the paths created and defined in the I/O Setup are what you will see listed there.

A logical grouping of multiple inputs, outputs, inserts, or busses that has a single name and (channel) format is referred to as a signal 'path' in Pro Tools terminology. These paths can include main paths and sub-paths. An example of a main path would be a master stereo output path with its left and right channels. A sub-path represents a signal path within a main path. For example, a default stereo output bus path consists of two mono sub-paths, left and right. Mono tracks and sends can be routed to either mono sub-path of the stereo output bus path. Multichannel bus paths can have any number of sub-paths.

NOTE

For Pro Tools systems such as the Mbox Pro and the 003, and for HD MADI, physical outputs are fixed. For third-party and built-in hardware, click the Launch Setup App button in the Hardware Setup for available configuration options.

The I/O Setup dialog window has six pages that you can access by clicking on the 'tabs' that run across the top of the window, marked Input, Output, Bus, Insert, Mic Preamps, and H/W Insert Delay.

INPUT

The Input page of the I/O Setup allows you to create and assign Pro Tools Input channels to receive audio from the physical inputs of your audio hardware – see Figure 2.13.

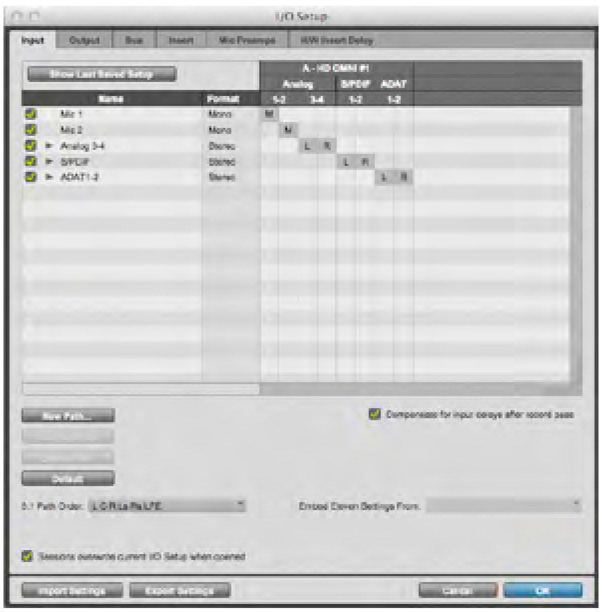


FIGURE 2.13
I/O Setup Input Page.

NOTE

Most of the time, you won't need to change anything in the I/O Setup unless you change your system hardware or you want to customize your I/O paths.

If you double-click on the name of an input path, you can rename this with something more meaningful if you prefer, such as AKG C12 instead of Input 1. Also, under the column headed Format, pop-up selectors let you choose mono, stereo (or multichannel with PT HD) formats.

TIP

If you want to return the I/O settings to their initial states, re-setting all the names, formats, and assignments, you can go through the different tabs for Input, Output, Insert, Bus, and so forth, clicking on the Default button for each I/O Setup section to reset these labels to the defaults for your system.

For example, I named Pro Tools input and output pair 9-10 'S/PDIF' to make it clear that these were connected to the S/P DIF I/O on the hardware. It is always a good idea to use descriptive names for the Inputs, Outputs, Inserts, and Buses, as this makes things much easier when you are setting up your tracks and mixer configurations during your Pro Tools Session.

NOTE

With an Eleven Rack connected to your Pro Tools system, Pro Tools lets you embed the current Eleven Rack Rig settings into audio clips as you record them, so that you can retrieve these for further use later. This can also be helpful when collaborating or bringing sessions or clips to a different system using Eleven Rack, because your settings travel with the files. Here's how this works: set the Embed Eleven Settings from pop-up selector to the input you plan to record through, such as Eleven Rig L/R; create an audio track and set its input to the corresponding input on your audio interface; and record-enable the track and record your part. The settings of the currently loaded Rig will be embedded in each audio clip that is recorded from the input you selected. To help you identify these, Audio clips with embedded Rig data are marked with a small Eleven Rack logo icon in both the Edit Window and the Clip list in Pro Tools.

OUTPUT

The Output section in the I/O Setup dialog (see **Figure 2.14**) lets you write your own names for the output signal paths, choose the formats for these, and choose which physical outputs of your audio hardware these output paths are assigned to.

Depending on your system hardware, there are various other settings that you can make. With the HD Omni, for example, you can also choose which physical outputs on your interface are to be used for the Output Meter Path, the Audition Paths, the Default Output Bus, the After Fader Listen (AFL)/Pre Fader Listen (PFL) path, and the AFL/PFL Mute. You can also choose a Default Monitor Format and a 5.1 Path Order for multichannel operation.

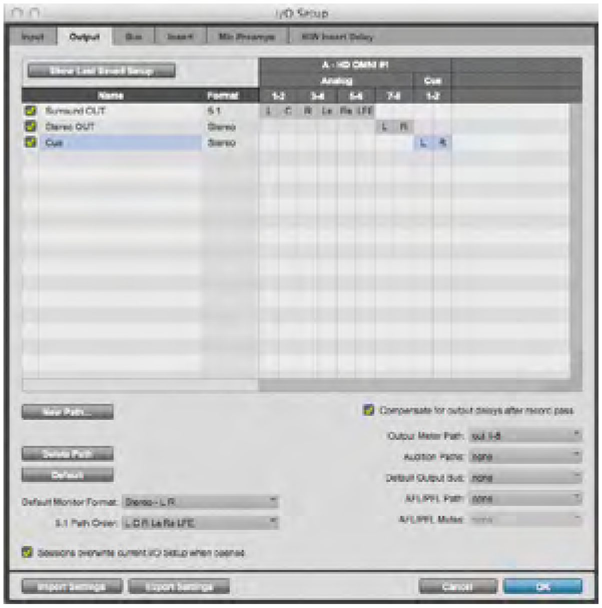


FIGURE 2.14
I/O Setup Output Page.

NOTE

AFL is the acronym for After Fade Listen and PFL is for Pre Fade Listen. PFL is useful for monitoring channels with the faders all the way down so that you can listen for a noisy microphone coming through a channel without hearing this in your mix. You could send the PFL channels to a headphone amplifier, for example, so you can independently monitor these channels to make sure they are OK before feeding them into the mix that you are sending to a PA system or for broadcast. When you monitor using AFL, the volume level of what you hear depends on where the fader is set. So if you pull the channel fader all the way down, you won't hear anything through this channel when you are using AFL.

The Output Meter Path selector lets you choose which output or bus paths will be monitored by the Output Meters in the transport.

TIP

You can configure these output paths – naming, formatting, and assigning them to physical outputs – in the Output page of the I/O Setup.

BUS

The Bus page of the I/O Setup (see **Figure 2.15**) lets you type in your own bus path names, choose the formats, and map any main bus path to any of the available output paths of the same channel width or greater. So, for example, a mono bus can be mapped to a mono output path, a stereo bus can be mapped to a stereo output path, and a 5.1 surround bus can be mapped to a 5.1 surround output path. The bus page also lets you map output busses to output paths.

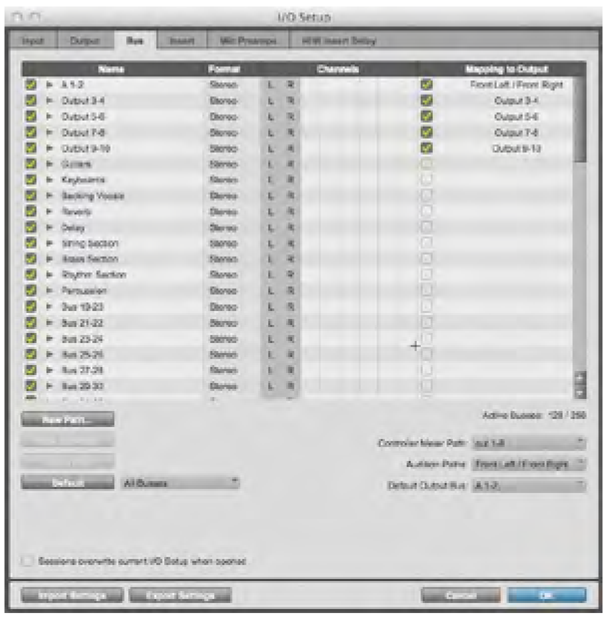


FIGURE 2.15
I/O Setup Bus Page.

Pro Tools provides up to 256 internal mix busses that can be used to route audio from track outputs and sends to other track inputs or to plug-in sidechains. Busses are typically used with the Sends in the Pro Tools Mix window to route audio from one or more audio tracks to an Auxiliary Input track used as a Reverb or Effects return (to the mix) channel. Another popular use is to create sub-mixes of groups of tracks containing Backing Vocals, Strings, Brass, Guitars, Keyboards – or whatever you find useful – by routing the outputs of these tracks via internal busses to Auxiliary Input tracks. Busses can also be used to route audio via Auxiliary tracks to physical outputs on an audio interface to use as Cue or Headphone mixes.

INSERT

You can create and edit hardware insert signal paths for the Pro Tools mixer using the Insert page of the I/O Setup, (see Figure 2.16), naming these appropriately and choosing which pairs of inputs and outputs to use for these on your audio interface. So, for example, if you want to be able to use some external 'outboard' effects devices, such as a Lexicon 224 reverb or even a vintage EMT 140 'echo' plate, you would connect the inputs and outputs of these devices to available inputs and outputs on your Pro Tools audio interface.

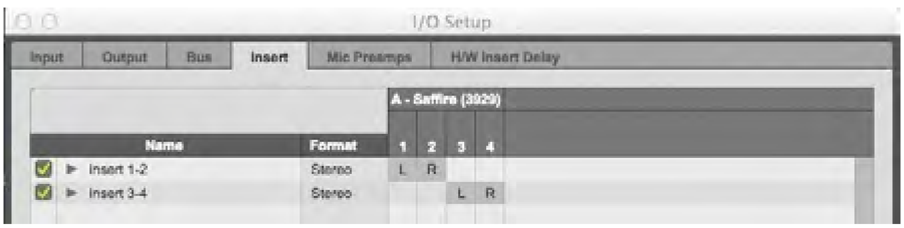


FIGURE 2.16
Part of I/O Setup Insert Page.

H/W INSERT DELAY

If you are using any external hardware devices, such as a reverb unit or a compressor, you can set a specific amount of Hardware Insert Delay Compensation, in milliseconds, for each external device to compensate for the delay (latency) that will occur between sending audio from Pro Tools to the device and Pro Tools receiving audio back from the device – see Figure 2.17. When the hardware insert is in use and Delay Compensation is enabled, these delay times will be used by the Delay Compensation Engine to time-align the input paths.

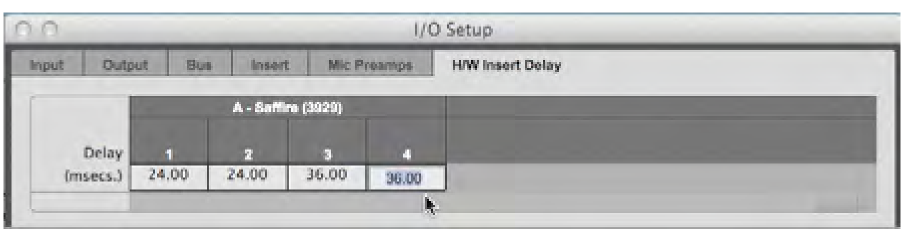


FIGURE 2.17
Part of I/O Setup H/W Insert Delay Page.

I/O SETUP OPTIONS

Pro Tools systems have several additional I/O Setup options depending on which page of the I/O Setup you are viewing. These include default signal routing for metering and auditioning and default track layout for multichannel mix formats.

I/O DELAY COMPENSATION

Pro Tools|HDX and Pro Tools|HD Native Systems provide two options for compensating for input and output latency (due to any inherent latency in the analog-to-digital and digital-to-analog converters of the audio interface) after recording.

A 'Compensation for Input Delays After Record Pass' option is available in the I/O Setup Input page – see Figure 2.18. This provides automatic compensation for any analog or digital input delay with Avid HD interfaces, so you should enable this option whenever you are recording.

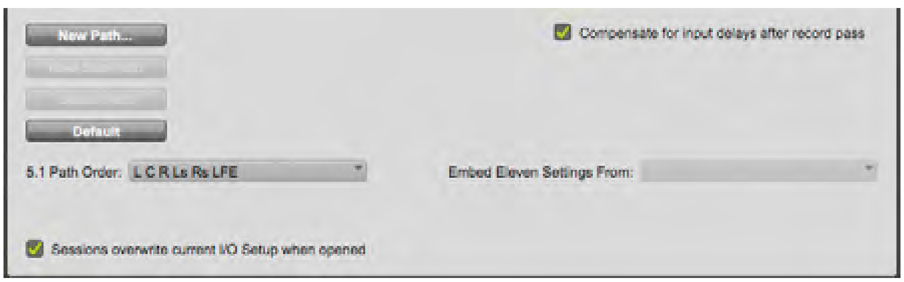


FIGURE 2.18
Compensation for Input Delays After Record Pass option in the I/O Setup Input page.

Similarly, in the I/O Setup Output page, there is a 'Compensation for Output Delays After Record Pass' option that provides automatic compensation for any analog or digital output delay with Avid HD audio interfaces – see Figure 2.19. You should enable this option whenever you are synchronized to an external clock source.

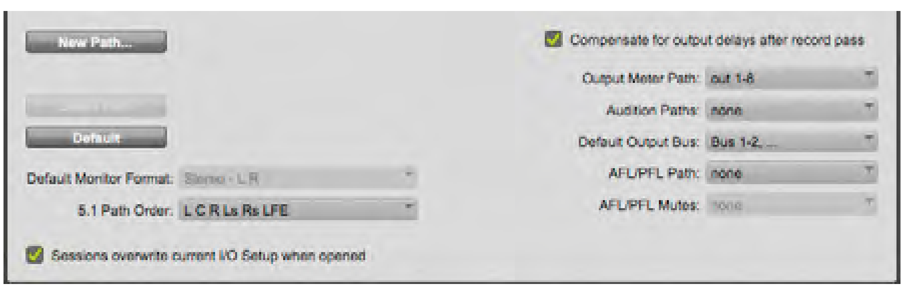


FIGURE 2.19
Compensation for Output Delays After Record Pass option in the I/O Setup Output page.

NOTE

When recording from a digital source, both the Compensation for Input Delays After Record Pass and the Compensation for Output Delays After Record Pass options must be enabled.

AUDITION PATHS

You can specify the output path to be used as the 'Audition Path' to playback files and clips when you audition or 'preview' these from the Clip List or Workspace browsers, and when previewing AudioSuite processes. Pro Tools assigns as default Audition Path, the first available main Output path of the corresponding format. If you prefer, you can select a specific Audition Path using the Audition Paths pop-up selector, which is available in the Output and Bus pages of the I/O Setup.

DEFAULT OUTPUT BUS

You can also specify the default output bus path assignment for new tracks, in each available format using the Default Output Bus selector, which is available in the Output and Bus pages of the I/O Setup. The Default Output Bus can also be set for internal mix bus paths.

AFL/PFL PATH

When you are using Pro Tools HD with Avid HDX or HD Native Hardware, there is an AFL/PFL Path selector available in the Output and Bus pages of the I/O Setup. Tracks soloed in AFL or PFL Solo mode are played back via the current AFL/PFL Path that you have set using the AFL/PFL Path selector. Be aware that if you select 'None' as the AFL/PFL path, this disables the AFL and PFL solo modes.

NOTE

In the Mix or Edit window, you can set separate master playback levels for all AFL and PFL solos. Here's how this works: choose the AFL or PFL solo mode from the Options menu Solo Mode sub-menu first. Next, Command-click (Mac) or Control-click (Windows) a Solo button on any track in the Mix or Edit window. Adjust the AFL/PFL Path fader then click on the new fader position (or press Esc) to close the fader display. To set the AFL/PFL Path level to 0 dB, Command-Control-click (Mac) or Control-Start-click (Windows) on any Solo button.

AFL/PFL MUTES (OUTPUT PATH)

The AFL/PFL Mutes selector becomes available in the Output and Bus pages of the I/O Setup when you are not using a D-Control or D-Command work surface. This allows you to mute the normal Pro Tools output path when you send a signal to the AFL/PFL Path. To set which output path will be muted when tracks are soloed in AFL or PFL Solo mode, select a path from the AFL/ PFL Mutes (Output Path) pop-up selector.

I/O SETUP RULES

Path configurations, Input, Output, Insert, and bus names, and channel widths in the I/O Setup are saved as I/O settings with both the session and the system and can be recalled from either.

When you take a session from one system to another, track and send assignments are maintained within the session and, where possible, Pro Tools automatically remaps the session's output busses to the output paths of the system on which the session is being opened.

A checkbox near the bottom of the Input, Output, Bus, and Insert pages of the I/O Setup window lets you choose whether the I/O settings saved with the session will overwrite the I/O settings saved with the system when you open the session. When this option is enabled, which is the default, Pro Tools recalls these settings from the session rather than the system. This option is the best choice when exchanging sessions with systems running Pro Tools 8.0.x and lower. When this option is disabled, Pro Tools recalls these settings from the system, which is the best choice when exchanging sessions among different systems running Pro Tools 8.1 or higher.

When you create a new session, you can specify which I/O Settings to use in the I/O Settings pop-up menu in the Quick Start and New Session dialogs. For example, you can use the factory installed default settings, the 'Last Used' setting, or one of any available custom I/O settings files.

If any changes are made to the I/O Setup, these changes are automatically saved to the I/O Settings folder as the Last Used settings file when the I/O Setup is closed.

Custom I/O Settings files can be created by changing I/O Setup settings and then exporting the settings. These I/O settings can then be restored by importing them into a system.

I/O settings can be imported and exported for use with sessions shared between systems, and these can be imported either before or after you open a session.

I/O Settings are only imported for the current page of the I/O Setup – helping you to avoid overwriting any I/O settings you have made on the other pages. When you export I/O settings, on the other hand, all the pages of the I/O Setup are exported, so that all the latest changes you have made are preserved.

GETTING STARTED

THE QUICKSTART DIALOG

When you launch Pro Tools, the QuickStart dialog appears (unless you have previously deselected this option in the Warnings & Dialogs section of the Display Preferences or in the Quickstart Dialog itself). Using this dialog, you can create a blank session, create a new session from a template, open any of the last 10 previously opened sessions, or open any existing session.

TIP

If you click on the Session Parameters arrow in the QuickStart dialog, you can choose a different audio file type, bit depth, or sample rate for your new session.

A selection of templates is provided in the QuickStart dialog (see **Figure 2.20**) containing useful Session configurations – and you can easily create your own if you prefer. Simply create a new Pro Tools session, configure it however you like, and choose 'Save As Template' from the File menu. For songwriting, you might just have one mono track for guitar, a second mono track for voice, a stereo Instrument plug-in for drums or percussion, an Auxiliary Input with a reverb plug-in inserted, and a stereo Master Fader.

FIGURE 2.20
The Quick Start dialog.

TIP

You might prefer this second method if you want to keep the template with the project that you are working on and intend to move this to another system at some point during the project.

In the Save Session Template dialog, you can choose 'Install Template In System', which installs the template file in the system folder referenced by the Pro Tools Session Quick Start dialog.

Alternatively, you can choose 'Select Location For Template', which lets you select any other location, in which case the session template won't appear in the Pro Tools Session Quick Start dialog – but you can simply open this file to start a new session from this template.

When the Include Media option is enabled, any audio, MIDI, or video media in the session is included in the template. This is useful if you want to use some standard media elements, such as a particular sound effect or drum loop, in a series of related Pro Tools Sessions.

TIP

There is a convenient keyboard command that lets you quickly re-open the most recently opened session: Command-Shift-O (Mac) or Control-Shift-O (Windows). This is very useful if you have just closed this file and want to immediately open it again, perhaps because you forgot to do something.

OPENING SESSIONS WITH PLUG-INS DEACTIVATED

If you are using a lot of plug-ins in a particular session, these can take a long time to load. If you simply want to open the session to check if this is the one you want to work with, or to make some changes that you will not need the plug-ins for, Pro Tools provides a convenient way to open sessions with all of the session's plug-ins set to inactive.

From the File Menu, choose Open. When the Open Session dialog appears, first locate and select the session you want, then Shift-click Open. The session will now open with the plug-ins deactivated.

If you decide that you do want to work with this file with the plug-ins loaded and activated, you can either choose Revert To Saved from the File menu or choose Open Recent from the File menu and select the most recent session in the sub-menu.

TIP

You can always make changes to a session and save those changes in a new session file to create multiple versions of a session or to back up your editing and mixing work.

PRO TOOLS SESSIONS

So what is a 'session'? Well, this is the name given to the project document that Pro Tools creates when you create and configure a new session. This document, which has the file extension '.ptx' when it is saved to disk, contains all your edit and mix information, and it references any audio, MIDI, video, or other files associated with this project. The session file does not actually contain any audio or video media files – these files are quite separate – and only one session file can be open at any one time.

When you have created or opened a Pro Tools session, you can work on your recording, editing, mixing or post-production session, importing, exporting, creating or referencing audio, video, or MIDI 'media' files. MIDI data can be created directly within the session and stored in the session document, or can be imported or exported as standard MIDI files, while audio and video files recorded or imported into a Pro Tools session are always stored separately from the Pro Tools session file in dedicated Audio Files and Video Files folders within the Pro Tools Session folder.

NOTE

Audio files are listed in the Pro Tools Clip List and can appear in an audio track. A section of an audio file can be defined as a clip. Video files can be created in (or copied to) the Video Files folder in the session folder. However, in most cases, Pro Tools references video files that have been captured by another application, such as Avid Media Composer.

TRACKS

Within a Pro Tools session, video, audio, MIDI, and automation data can be recorded and edited on video, audio, MIDI, or Instrument 'tracks' that can be viewed in the Mix and Edit windows.

Auxiliary Input tracks can be used route additional external inputs (or internal sources entering via internal busses) out to the various outputs, or out via internal busses to other destinations. These Auxiliary Inputs are often used to create submixes or used as Effects Returns with reverb, delays, or other effects inserted.

Master Faders are normally used to control the level and panning of audio being routed to physical outputs, although these have other uses that will be discussed later in this book.

Pro Tools HD also has VCA Master tracks that allow you to control other tracks in a Mix Group that has been assigned to the VCA Master.

TIP

Pro Tools 11 offers a great new way to create new tracks by double-clicking in blank space below tracks in the Edit window, or below or to the right of tracks in the Mix window, or in the empty area below any current tracks in the Tracks list. Double-clicking in any of these blank areas will add a new track of the same type (Audio, Auxiliary Input, Instrument, VCA Master, MIDI, or Master Fader) and channel width as the last new track that you added. If there are no tracks in the session yet, a stereo audio track will be created by default.

If you want to ensure that the track type added will be an audio track of the same channel width as the last new track, then hold the Command key (Mac) or Control key (Windows) as you double-click.

If you want to specifically create a new Auxiliary track of the same channel width as the last new track, hold the Control key (Mac) or Start key (Windows) as you double-click. If no tracks exist in the session, a stereo Auxiliary Input track is created by default.

To add a new stereo Instrument track, hold the Option (Mac) or Alt (Windows) key.

To specifically add a new Master Fader track of the same channel width as the last new track, hold the Shift key as you double-click, and, again, if no tracks exist in the session, a stereo Master Fader track will be created by default.

CLIPS

A segment of audio, MIDI, or video data is referred to as a 'clip'. So, a clip could contain a drum loop, a guitar riff, a verse of a song, a recording take, a sound effect, and some dialog – or an entire sound file. A clip can also have associated automation data. Video tracks (on Mac or Windows) let you work with QuickTime movies or VC-1 video files (on Windows), or with Avid video – although you cannot mix these formats on a single track.

In a typical Pro Tools session, clips are created from audio files or MIDI data, and arranged in audio and MIDI track 'playlists' along the timeline in the Edit window. Clips can also be grouped (to form a 'clip of clips') and they can be made to act like 'loops' by being repeated end-to-end along the timeline in the Edit window for as long as you would like the 'loop' to last. Using these techniques, you can re-arrange sections or entire songs, and assemble tracks using material from multiple takes.

PLAYLISTS

A 'playlist' is simply a sequence of clips arranged on an audio, MIDI, or video track, and tracks can have both Edit playlists and Automation playlists. For example, on an audio track, you might have recorded a guitar part that lasts for four bars of an eight bar intro, followed by a second guitar part that plays in the bridge sections, with a third guitar part that plays during the end choruses. These parts may have been played by three different guitarists and recorded at different times during your recording project, and the three parts will each be contained within separate Clips. When you come to mixing the track, you may decide to have copies of the guitar part from the end choruses play during the earlier choruses and maybe you will decide to add a chorus effect to the guitar parts that play in the bridge sections. All this can be done using a single Edit playlist with an Automation playlist to control the effects. Because you can use a copy of the audio clip to play one guitar part in different locations, this doesn't use any additional disk space, as the copy of the clip simply plays the original guitar in the new locations. It is also very easy to apply different effects to an original piece of audio, and then play this back in different places.

Perhaps the most powerful feature of playlists is that you can create any number of alternate playlists that you can choose from within each track. An example of how you might use these within a single audio track is when you are recording, say, a saxophone solo. You could record the musician playing several different versions of the solo into different playlists within the same track, then choose between these later – maybe using part of one 'take' followed by part of a different 'take', until you have created the perfect combination of notes.

NOTE

Each audio, Auxiliary Input, Instrument, Master Fader, and VCA track also has a single set of automation playlists. Automation playlists can include volume, pan, mute, and each automation-enabled control for the insert and send assignments on that track. MIDI controller data on Instrument and MIDI tracks is always included as part of the track playlist.

CHANNELS VS. TRACKS

In Pro Tools terminology, 'tracks' are used to record audio (or MIDI) to your hard drive. These tracks appear in the Edit window and in the Mix window 'channel' strips.

Rather confusingly, what you might expect would be referred to as 'channels' in the Mix window are referred to as Audio Tracks, Video Tracks, MIDI Tracks, Instrument Tracks, Auxiliary Inputs, VCA Master, and Master Faders in Pro Tools.

Also, Pro Tools uses the term 'channel' to refer to an actual physical input or output connection on whichever interface or interfaces you are using, so the documentation talks about an interface having 16 channels of audio input and output, for example.

All this is straightforward enough to understand – but it is a little unusual when you are used to a tape recorder having tracks and a mixer having channels. It can be a little 'jarring' mentally to talk about adjusting the mixer's 'track' controls rather than the mixer's 'channel' controls until you get used to the idea. But the documentation still talks about 'channel' strips in the Pro Tools Mix window – each of which corresponds to a track in a Pro Tools session.

VOICES AND 'VOICEABLE' TRACKS

The concept of 'voices' in Pro Tools needs to be thoroughly understood. The number of playback and recording 'voices' is the number of unique simultaneous playback and record tracks on your system – a bit like the concept of polyphony in a synthesizer, which dictates the maximum number of simultaneously playable notes.

Every Pro Tools system has a maximum number of voices, depending on the hardware it uses, that defines how many audio tracks you can play back or record at the same time. Pro Tools terminology, somewhat confusingly, also refers to 'voiceable' or 'virtual' tracks, when describing the maximum number of audio tracks that the software can support in a session.

For example, a Pro Tools HDX system with one HDX card can support up to 768 'voicable' or 'virtual' tracks (using Avid's terminology for these) – but you cannot record to these all at once. And you can't play all these back at once either. You can only record and play back up to the limit of the available 'voices' within your system, and this depends on the type of card or cards you have in your system and the sample rate you are working at. An HDX system with one card, for example, provides a maximum of 256 voices at 48 kHz and the older Pro Tools HD cards supported a maximum of 192 voices at 48 kHz – yet Pro Tools HD software running on either of these cards supports up to 768 voiceable tracks! To put this another way: the total number of 'voiceable' tracks is the maximum number of audio tracks that can share the available voices on your system.

Are you still feeling confused? Well, here's the thing: the Pro Tools software lets you create many more audio tracks than there are available voices, and save these in your session, but you just cannot record to or play back more tracks than there are available voices at any one time. With an HDX card, you could record up to 256 tracks at once, then record a further 256 tracks, and then a further 256 tracks for a total of 768 tracks. When you playback this session using just one HDX card, you would only hear the 256 tracks with the highest track playback priority – the others would not be heard. But if you have three HDX cards, you could hear all 768 tracks!

NOTE

By way of comparison, standard Pro Tools systems let you play back up to 96 simultaneous stereo or mono tracks, but can only record up to 32 tracks simultaneously.

Here are some relevant facts and figures to further clarify the situation: Pro Tools HD can open sessions with up to 768 audio tracks, but any audio tracks beyond that system's voiceable track limit will be automatically set to Voice Off. Each HDX card lets you record and playback up to 256 tracks at once at 44.1/48 kHz, you can use up to 3 cards in a system, and each HDX card supports a maximum of 64 channels of I/O. If your requirements are somewhat humbler, an HD Native system may suffice. HD Native hardware lets you record and play back up to 256 tracks at once at 44.1/48 kHz and supports a maximum of 64 channels of I/O, but, unlike the HDX cards, doesn't have on-board DSP processing. How many tracks you can actually record at once will also depend on how many inputs you have available via your audio interface or interfaces. For example, Avid interfaces offer a range of configurations: HD OMNI has 8 inputs and 8 outputs, HD I/O has 16 inputs and 16 outputs, and HD MADI has 64 inputs and 64 outputs.

A big advantage that Pro Tools HDX cards have over the previous-generation Pro Tools HD cards is that they use dedicated Field Programmable Gate Arrray (FPGA) chips to provide the 256 voices on each card in addition to the DSP chips used to support plug-in processing and mixing. As a consequence, the number of tracks that you use in a session does not reduce the amount of DSP available for plug-in processing and mixing and the maximum voice count is always available.

On Pro Tools HDX systems, the number of available voices will depend on how many of these cards you are using in your system – the more cards you have, the more voices you can use. On all other Pro Tools systems, the maximum number of voices may be limited by the host-processing power of your computer – so you will need to use a powerful computer for optimum results.

USING UP VOICES

Each audio channel for each track in your Pro Tools session uses a single voice. So, for a mono audio track, a single voice is used; stereo and multichannel tracks take up one voice per channel. However, you do need to bear in mind that if you are using Punch Recording, two voices are needed for every single audio channel (one for playback and one for recording on punch in and out).

Also, with Pro Tools|HDX systems, when you insert a Native (host-based) plugin, this may cause additional latency and will take up two additional voices per channel (one voice for input and one voice for output) when inserted on an Auxiliary Input or Master Fader track; or when inserted on an Instrument track that does not contain an instrument plug-in; or when inserted after a DSP plug-in on any kind of track.

For example, the initial insert of a host-based plugin on a mono Auxiliary Input track uses two voices (one channel with two voices), while the initial insert of that plug-in on a stereo Auxiliary Input track uses four voices (two channels with two voices each). Subsequent host-based plug-ins on the same track do not take up additional voices unless a DSP plugin is inserted between other host-based plug-ins.

NOTE

You should always avoid inserting DSP plug-ins between host-based Native plug-ins on any kind of track as this will cause unnecessary voice usage and may cause additional latency.

There are various other scenarios in which additional voices are used: when you select multiple track outputs for a track, one voice is used for each output; and when you select an AFL/PFL Path output in the Output tab of the I/O Setup dialog, one voice is used for each channel. Also, when you use the external key side-chain of a host-based plug-in on a track, one additional voice is used.

RE-CLAIMING VOICES

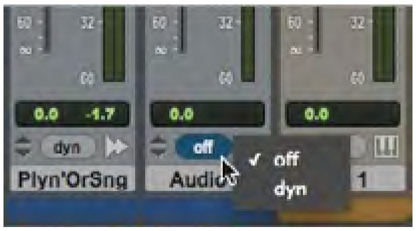


FIGURE 2.21
Setting the Track Voice Selector to 'Off'.

So what happens when you run out of voices, for instance, if you open a session created using a large Pro Tools HD/HDX system on a laptop using Pro Tools? Well, Pro Tools will automatically make tracks inactive as necessary to allow sessions to be opened. So if the session contains more than the number of tracks supported on your Pro Tools system, audio tracks beyond the system's 'voiceable' track limit will be automatically set to inactive.

However, although your Pro Tools hardware only allows a fixed number of voices, Pro Tools software will let you work with many more audio tracks in a session – you can record new tracks, import audio to new tracks, edit any of the existing tracks, and so forth. It's just that you cannot simultaneously playback all the tracks beyond the voice limit.

What you can do is to adjust the playback priorities of the audio tracks in these larger sessions so that the most important tracks can be heard. The track priority depends on the order of the tracks in the Mix and Edit windows – with the leftmost track in the Mix window and the topmost track in the Edit windows having the greatest priority. The order of priority for track playback runs from left to right in the Mix window and from top to bottom in the Edit window. So if a track will not play back because you have run out of voices, you can simply drag the track to the left in the Mix window or higher up in the Edit window until it does play back. Of course, you will lose playback on the higher priority track that you have displaced to a lower priority, so you will have to make decisions about which tracks are most important for you to hear.

FIGURE 2.22
Clicking on the track type icon to access the Make Inactive pop-up.

Another way is to free up some of the voices being used by less important tracks, at least temporarily while you work on the tracks that you need to hear. Pro Tools will always free up a voice if you un-assign the track's output and send assignments, or you can set the track's Voice Selector to 'Off' – see Figure 2.21.

Perhaps the best way is to make the less important tracks completely inactive by clicking on the track type icon that you will find to the right of the Voice Selector on each mixer channel strip and choosing 'Make Inactive' from the popup selector that

TIP

You can use a keyboard command to toggle the track between active and inactive: Command-Control-click (Mac) or Control-Start-click (Windows)on the Track Type indicator in the Mix window.

FIGURE 2.39
Pro Tools Mix window with mouse pointing to the Group settings pop-up selector. The pop-up selector for an adjacent track is open in front of the Mix window.

appears – see Figure 2.22. To give you visual feedback, mixer channels turn a darker shade of grey and tracks in the Edit window are dimmed when these are inactive.

Making some of your less-important, or unused, tracks inactive is a great way of freeing up DSP resources and voices for use elsewhere, as all the plug-ins, sends, voices, and automation on inactive tracks are disabled. This feature also allows you to open Pro Tools sessions on systems with less DSP resources than were available when the sessions were created.

THE MAIN WINDOWS

The Edit and Mix windows are the main Pro Tools work areas. Depending on which phase of your project you are in or what type of project you are working on, you may prefer to work with just the Mix or just the Edit window.

You can use the View Menu to choose what will be presented to you in the Edit and Mix windows. So, for example, if you reveal the Instruments, I/O controls, Inserts, and Sends in the Edit window, you can mostly work in just the Edit window, which some people find easier than working in two windows.

ORGANISING THE MIX WINDOW

In the Mix window, tracks appear as mixer channel strips with controls for signal routing, input and output assignments, volume, panning, record enable, automation mode, and solo/mute.

From the View menu, you can choose whether to display the Mic Preamps, Instruments, Inserts, Sends, Meters & Faders, Delay Compensation, Track Color, or Comments in the Mix window.

THE GROUP SETTINGS POP-UP SELECTOR

Directly above the Pan controls at the top of each fader section, there is a pop-up selector that lets you control any Group settings that you have made for the track – see Figure 2.39.

Using this pop-up selector, you can select or hide all the tracks in the group, show only the tracks in the group, delete, duplicate, or modify the group, and see which tracks and attributes are included in the group.

INSERTS

Pro Tools lets you use up to 10 Inserts on each Audio track, Auxiliary Input, Instrument Track, or Master Fader. Each insert can be either a software plug-in or an external hardware device. There are two sets of Inserts, labelled A–E and F–J, so you can conserve space onscreen by only displaying one set at a time.

SENDS

You can also use up to 10 Sends on each track. Sends let you route signals across internal buses or to audio interface outputs, so that one plug-in or one external signal processor can be used to process several tracks at once.

There are two sets of sends (labeled A–E and F–J) that can optionally be displayed in the Mix window. You might use the first set to send to effects (such as reverb) that you wish to apply to several tracks and use the second set to send cue mixes to musicians – routing these from your Pro Tools hardware interface to suitable headphone amplifiers.

If you are not using the second set of sends, or the mic preamps, or you don't need to see the delay compensation controls or comments, or whichever, then you should hide these so that they don't distract you from the controls you do want to use and so that the Mix window takes up less space on your computer screen.

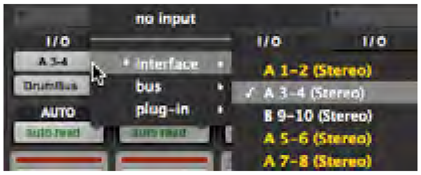


FIGURE 2.40
Track Input Selector pop-up.

I/O SELECTORS

Track Input and Output Selector pop-ups are located below the Inserts and just above the Automation controls on the channel strips. The Track Input Selector pop-up (see Figure 2.40) lets you choose input sources for Audio tracks, Instrument tracks, and Auxiliary Inputs. Track input can come from your hardware interface, from an internal Pro Tools bus, or from a plug-in.

The Track Output Selector pop-ups let you route the audio from each track to your choice of available outputs or bus paths.

ASSIGNING MULTIPLE TRACK OUTPUTS AND SENDS

Pro Tools Audio tracks, Instrument Tracks, and Auxiliary Inputs can have multiple track output and send assignments chosen from the actual paths and resources available on your system (although Master Faders can only be assigned to a single path). Assigning to multiple paths is an efficient way to route an identical mix to other separate outputs, for simultaneous monitor feeds, headphone mixes, or other situations where a parallel mix is needed.

FIGURE 2.41
Making multiple assignments for a track's Audio Output Path.

FIGURE 2.42
Multiple assignments for a Send's Audio Output Path.

To assign an extra output, hold down the Control key (Mac) or Start key (Windows), open the Output Selector, and select your additional output. A '+' sign is added to the Output Selector legend to remind you that this track has more than one output assigned (see Figure 2.41), and you can add as many additional outputs as are available on your system. If you also hold the Option (Alt) key at the same time as the Control (Start) key, the additional output will be added to all tracks (apart from Master Faders and MIDI tracks, of course).

NOTE

You can use the same procedure to add additional output assignments to track Sends – see Figure 2.42.

ORGANISING THE EDIT WINDOW

The Edit window provides a timeline display of audio, MIDI data, and mixer automation. As in the Mix window, each track has controls for record enable, solo, mute, and automation mode.

Using the View menu options, you can also reveal the input and output (I/O) routing assignments, the Inserts, the Sends, the Instrument controls, the RealTime properties, the Comments, or any combination of these – which makes it possible to work with just the Edit window for most of the time.

The Edit window lets you display the audio and MIDI data in a variety of ways to suit your purpose, and you can edit the audio right down to sample level in this one window.

THE TOOLBAR

Understanding the area at the top of the Edit window is an essential part of learning how to use Pro Tools. This area, called the Toolbar, contains the Edit Mode buttons, the Zoom buttons, the Edit Tools, the Location indicators, various other displays and controls, and the Grid and Nudge controls.

Pro Tools lets you customize the Toolbar in the Edit window (as with the MIDI and Score Editor windows), by re-arranging, showing, and hiding the available controls and displays. To show or hide the various tools, you can use the pop-up Toolbar menu located in the upper-right corner of the Edit window.

Here you can choose to display the Output Meters and/or the Zoom Controls, the Transport, MIDI, and Synchronization controls in the Toolbar if you wish. You can also show or hide the Track List or Clips List, reveal the Universe view at the top of the Edit window, or reveal the MIDI Editor at the bottom of the Edit window.

With all these showing in the Edit window, this allows you to work almost entirely within the one window. The disadvantage is that the Edit window then starts to look very cluttered – which is why you are given the option to hide whichever of these elements you are not using (see Figure 2.43).

FIGURE 2.43
Toolbar showing the basic set of controls and displays with the zoom controls.

REARRANGING TOOLBAR CONTROLS AND DISPLAYS

To rearrange controls and displays in the Edit (or MIDI or Score Editor) window toolbar, simply Command-click (Mac) or Control-click (Windows) on the controls or displays that you want to move and drag them to the right or to the left to reposition them along the toolbar. So, for example, if you prefer to have the Counters and Edit Selection indicators repositioned to the right of the Transport controls in the toolbar, it will only take you a moment to do this.

TIP
To enable Shuffle, Slip, or Spot while in Grid mode: Shift-click the Shuffle, Slip, or Spot mode button. Alternatively, press F1+F4 to enable Grid and Shuffle mode, press F2+F4 to enable Grid and Slip mode, and press F3+F4 to enable Grid and Spot mode.

THE EDIT MODES

At the left of the Toolbar, there are four buttons to let you select the Edit Mode (see Figure 2.44):

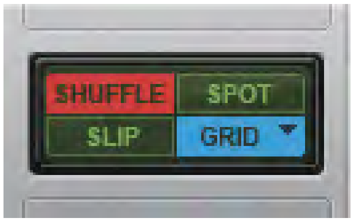


FIGURE 2.44
The Edit Mode Buttons.

- Slip Mode is the basic mode to use by default. In this mode, you can freely move clips forwards and backwards in time in the Edit window.
- In Shuffle Mode, when you move a clip, it will automatically snap to the clip before it – perfectly butting up to this.
- In Spot Mode, a Spot Dialog appears whenever you click on a clip. This lets you specify exactly where the clip should be placed on the timeline – ideal for 'spotting' effects to picture.
- In Grid Mode, movements are constrained by whichever Grid settings you have made – such as Bars:Beats or Mins:Secs.

Pro Tools lets you Shift-click to enable Grid Mode while in Shuffle, Slip or Spot mode. You can also use the keyboard command Shift-F4 if you prefer.

FIGURE 2.45
Zoom buttons and presets

With Grid Mode active, placing the Edit cursor and making Edit selections is constrained by the Grid, while clip editing is simultaneously affected by the other selected Edit mode. For example, in Shuffle mode, with Grid Mode also active you can make a selection in a clip based on the Grid, cut the selection, and any clips to the right of the edit will shuffle to the left.

THE TOOLBAR ZOOM BUTTONS AND PRESETS

To the right of the Edit Mode buttons, there are various arrow buttons that let you zoom the display vertically or horizontally – see Figure 2.45. The horizontal zoom arrows work for both audio and MIDI clips. To zoom audio clips vertically, use the first pair of up and down arrow buttons. To zoom MIDI clips vertically, use the second pair.

Underneath these zoom controls, there are five small Zoom Preset buttons that store preset zoom levels. You can use these as handy shortcuts to particular zoom levels: just set the zoom level you want, then Command-click (Mac) or Control-click (Windows) on any of the five buttons to store the current zoom level.

To recall these zoom presets, you can either click on the numbered buttons using the mouse, or press the numbers 1, 2, 3, 4, or 5 on your computer's QWERTY keyboard (not on the numeric keypad).

FIGURE 2.46
Horizontal and vertical zoom buttons

THE EDIT WINDOW ZOOM BUTTONS

In addition to the Zoom controls in the Toolbar, Pro Tools provides horizontal and vertical zoom buttons in the lower-right corner of the Edit window – see Figure 2.46. The Vertical Zoom buttons zoom the track heights proportionally in the Edit window. The Horizontal Zoom buttons zoom the Timeline, just like the Horizontal Zoom controls in the Edit window toolbar.

FIGURE 2.47
Audio and MIDI Zoom In and Out buttons

Pro Tools also provides Audio and MIDI Zoom In and Out buttons in the upper-right corner of the Edit window – see Figure 2.47. These controls function exactly the same as the Audio and MIDI Zoom controls in the Toolbar, and let you zoom in and out vertically on audio waveforms and MIDI notes, respectively, although MIDI Vertical Zoom only affects tracks in Notes view.

THE TOOL BUTTONS

Located by default to the right of the Zoom buttons, you will find six Tool buttons.

ZOOMER TOOL

The first button is the Zoomer Tool. You can use this to zoom the display either vertically or horizontally. If you click and hold this, you can select the Single Zoom mode. With this selected, you can click in the Edit window to zoom the display one time, then when you let go of the mouse, the Zoomer Tool is de-selected and the Selector Tool becomes active.

TRIMMER TOOL

The next button is the Trim tool. If you click and hold this, you can select various modes using the pop-up that appears. You can use the standard Trim tool to lengthen or shorten clips or notes. The Time Compression/Expansion Trim tool (TCE Trim) lets you apply Time Compression/Expansion directly in the Edit window. The Loop Trim tool lets you create or trim looped clips. Pro Tools HD also includes the Scrub Trim Tool. This lets you drag in a track to hear the audio, then trim at a specific location by releasing the mouse button.

SELECTOR TOOL

To the right of the Trimmer Tool is the Selector Tool that lets you use the cursor to select areas within the Edit window. With this tool selected, the mousepointer changes to an insertion cursor that you can use to drag across andselect clips in the Edit window or you can simply point and click at a particular location in the Edit window to position the insertion point (and update the Location Indicators) at that location.

TIP

Whenever you are not specifically using one of the other tools, the Smart Tool is the most useful (and safest) tool to leave selected.

GRABBER TOOL

To the right of the Selector Tool, you will find the Grabber Tool – the one with the 'hand' icon. You can use this to move clips around in the Edit window. You can also use the Grabber to automatically separate an edit selection and move it to another location or another track using its Separation Mode.

The Grabber Tool offers a third mode – the Object mode – that you can enter using the pop-up that appears when you click on the Grabber Tool. This 'Object Grabber' lets you select non-contiguous clips on one or more tracks. Just take a look at **Figure 2.48** to see an example of a 'non-contiguous' selection of two clips, all on different tracks. By the way, 'non-contiguous clips' in this context basically means clips that are not next to each other on the same track, or are on different tracks.

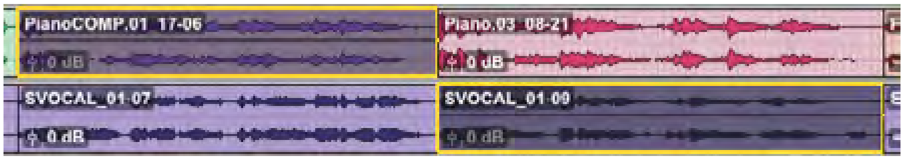


FIGURE 2.48
The Object Grabber Tool being used to select non-contiguous clips in the Edit window.

FIGURE 2.49
The Smart Tool.

FIGURE 2.50
Zoom Toggle button.

SMART TOOL

The Trimmer, Selector, and Grabber tools can be combined using the Smart Tool to link them together – see Figure 2.49. The Smart Tool button is positioned both above and to either side of the Trimmer, Selector, and Grabber tools. To activate it, simply click in this area above or to either side of these tools. With the Smart Tool activated, then, depending on where you point your mouse in the Edit window, one or other of these tools will become active – saving you having to click on these tools individually when you want to change to a diff erent tool.

SCRUB TOOL

To the right of the Grabber Tool is the Scrub Tool that lets you 'scrub' back and forth over an edit point while you are trying to hear the exact position of a particular sound – rather like moving a tape back and forth across the playback head in conventional tape editing.

PENCIL TOOL

The sixth tool is the Pencil that you can use to redraw a waveform to repair a pop or click. Alternate Pencil modes are available that constrain drawing to lines, triangle, square, or random shapes.

THE MODE BUTTONS

Running underneath the Tool buttons, there are six more mode buttons.

ZOOM TOGGLE

The leftmost button controls the Zoom Toggle – see Figure 2.50. This feature lets you define and toggle between zoom states in the Edit window, storing and recalling the Vertical Zoom, Horizontal Zoom, Track Height, Track View, and Grid settings. When Zoom Toggle is activated, the Edit window changes to the stored zoom state, and when Zoom

TIP

When Zoom Toggle is enabled, you can cancel it and remain at the same zoom level by pressing Option-Shift-E (Mac) or Alt-Shift-E (Windows).

Toggle is disabled, the Edit window reverts to the previous zoom state. Any changes made to the view while Zoom Toggle is enabled are also stored in the zoom state.

FIGURE 2.51
Tab to Transients button.

TAB TO TRANSIENTS

The Tab to Transients button (see Figure 2.51) lets you automatically locate the cursor to the next transient while editing waveforms. With this activated, just press the Tab key on your computer keyboard to jump to the beginning of the next transient (e.g. at the start of a snare hit) in the waveform that you are currently editing.

FIGURE 2.52
Mirrored MIDI Editing button.

MIRRORED MIDI EDITING

The third button from the left (see Figure 2.52) lets you enable or disable Mirrored MIDI Editing. Mirrored MIDI Editing is useful when you edit a clip containing MIDI notes and you want these edits to apply to every MIDI clip of the same name.

FIGURE 2.53
Automation Follows Edit button.

AUTOMATION FOLLOWS EDIT

When Automation Follows Edit is enabled, (see Figure 2.53 above), automation events are affected by edits to audio or MIDI notes.

LINK TIMELINE AND EDIT SELECTION

The Link Timeline and Edit Selection button (see Figure 2.54) lets you link or unlink Edit and Timeline selections. With the Timeline and Edit selections unlinked, you can select different locations in the Timeline at the top of the Edit window and in the clips within the Edit window. With the Timeline and Edit selections linked, whenever you make a selection inside the Edit window, the same selection is automatically made in the Timeline at the top of the Edit window.

FIGURE 2.54
Link Timeline and Edit Selection button.

LINK TRACK AND EDIT SELECTION

The Link Track and Edit Selection button (see **Figure 2.55**) also does what it says: with this highlighted, when you select a clip in the Edit window, the track becomes selected as well. If you then select another track, Pro Tools selects the corresponding clip (to the clip selected in the first track) in this other track – because the Track and Edit selection features are linked!

FIGURE 2.55
Link Track and Edit Selection button.

INSERTION FOLLOWS PLAYBACK

The Insertion Follows Playback button (see **Figure 2.56**) lets you enable or disable the 'Timeline Insertion/Play Start Marker Follows Playback' option in the Operation Preferences and also provides a visual indication of whether this option is on. Previously, this option was only available in the Operation Preferences.

FIGURE 2.56
Insertion Follows Playback button.

When this button is highlighted, the 'Timeline Insertion/Play Start Marker Follows Playback' option is enabled, and both the Timeline Insertion and the Play Start Marker will move to the point in the Timeline where playback stops. When deselected, the Timeline Insertion and Play Start Marker do not follow playback, instead they return to the location where playback began.

TIP

There is a useful keyboard command that lets you switch this preference on and off: press Control-n (Mac) or Start-n (Windows) or, with the Commands Keyboard Focus enabled, simply press the 'n' key.

The following is excerpted from *Digital Audio Editing* by Simon Langford.

You can purchase a copy HERE.

INTRODUCTION

In this section of the book, we are going to start to look at editing tools and processes that can not only serve a corrective purpose but also can be put to more creative uses if the need or opportunity arises. And as much as some of these techniques cross the line between corrective and creative uses, equally some of them cross the lines between editing, production, and sound design. So, first up, and following on from our last chapter on transient detection, it makes sense to start off with a pair of related but very different ways of using those transients that the software kindly worked so hard to figure out for us.

Beat-mapping is the process of analyzing an audio file—often longer files or even full songs—and using the transient markers created during the analysis as a way to subtly (or not so subtly) alter the timing of the piece while keeping it fluid and recognizable. Recycling is now a generic term for a process of cutting up an audio file—normally one that is quite short in length—into "slices" and then creating a sampler patch and associated MIDI file based on the timing of the slices. The name originates from the Recycle software released in 1994 by Propellerhead Software, but it has become such a well-used process that now, even though other tools, software, and methods have been developed to do the same process, the whole idea of loop-slicing has now become synonymous with the term "recycling."

Although both processes rely on transient markers, that is where the similarity ends. Each of the two has its strengths, its weaknesses, and its common uses.

BEAT-MAPPING

As mentioned briefly above, beat-mapping is set up more as a corrective tool. It will analyze an audio file and create transient markers throughout the whole length of the file. These transient markers can then be moved and the resulting groove and feel of the track changed. Crucially, though, as each transient marker is moved, the audio between it and the following marker is time-stretched, so that there are no gaps or overlaps between the markers. The whole thing is very fluid and consistent. This makes it ideally suited for quickly and easily changing the groove of a long audio file by choosing how the timing should be changed (usually quantizing) and then letting the software do the rest.

Another very useful consequence of the way that this system works is that, once the transient markers are in place, the whole track will then be easily mappable to a standard "bars and beats" format instead of being purely based on absolute time.

Instead of starting at 1:02 (minutes and seconds), the chorus can now be thought of as starting at bar 33. This shift from minutes and seconds to bars means that, even without actually changing the groove or quantizing, the file now has the ability to be tempo-independent, as any changes in tempo will mean that, instead of the software having to calculate that an event at 1:02 now has to occur at 1:05, and doing so for every "event" in the file, the software knows that what happens at bar 33 stays at bar 33.

Changing the tempo of an audio recording has been possible for a very long time, so being able to speed up or slow down an audio file isn't new, but the ability to do so without changing the pitch only comes with time-stretching. Equally, time-stretching itself isn't new but traditionally would require a file to be processed, and, once the destination tempo had been chosen, the only way to change it again would be to reprocess the file. Furthermore, any gradual changes to tempo would be impossible with regular time-stretching, so this method of actually having the tempo completely free to be set at whatever tempo you choose or even to change midway through a file represents not only a huge workflow improvement for static tempo changes but also a whole new range of possibilities for changing the tempo as the song progresses.

One of the other things that beat-mapping allows us to do from a corrective perspective is to make tracks that were not recorded to a click track—and consequently have tempo variations throughout their length—conform to a consistent tempo.

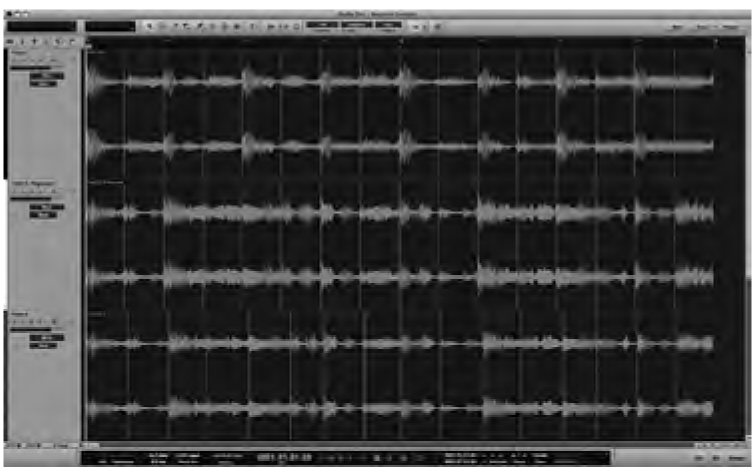


FIGURE 10.1
This is an illustration of the power of beat-mapping. The top two tracks show two different songs that would play back nicely in sync. The bottom track shows the original timing and positioning of the transient markers of the second track. There is quite a noticeable difference, and this would allow us to play the two tracks together without them sounding "messy".

There is an argument that doing this is actually taking some of the "life" out of a song, and I would be inclined to agree, but there is one situation where this would be extremely beneficial and one group of people who, as a result, have become very fond of beat-mapping.

DJs, at least in the case of DJs who mix records in to one another to achieve a constant flow rather than just having a very short cross-fade between one track and the next irrespective of tempo, have for a very long time been able to deal with mixing tracks together that are different in tempo. By adjusting the speed of the record (or CD), they can make the tempo match, but for them to be able to mix the records properly, the tempo throughout the song needs to be consistent. It is one thing to adjust the pitch/tempo of one record to allow it to be mixed in to another, but if there were constant tempo variations throughout the song, then, no matter how adept DJs were, they simply wouldn't be able to keep up with the changes, and the resulting mix between the two would be constantly drifting in and out of sync. By being able to remap the timing of a song so that it is at a consistent tempo, even if that is at the expense of a little bit of the feel and life of the track, they now have the ability to use that song in a predictable and reliable way in their mix sets.

QUALITY LIMITATIONS OF BEAT-MAPPING

Before we move on to look at some creative ways in which we can use beat-mapping, it is probably best to consider the quality issues of what we are trying to do. The biggest problem we face with this beat-mapping technique is that no time-stretching system is totally without artifacts. Depending on the time-stretching algorithm, the actual content of the file being stretched and the amount of stretch can range from almost unnoticeable to a very clear change. The most common artifact is a so-called smearing effect, which is perceived as a loss of definition and clarity and the tendency for the component parts to blend with one another.

There are several different methods used to achieve time-stretching (see chapter 12 for more information), and each of these methods/algorithms has strengths and weaknesses. Therefore, it is always advisable to try out any different options that you might be offered in terms of different algorithms. Sometimes these are named according to what they might be suitable for, and, given that the people who come up with these algorithms generally know what they are talking about, they are often correct. However, there might be times when a file that you have doesn't fall neatly into one of these descriptions or when you just have the time to try different things out.

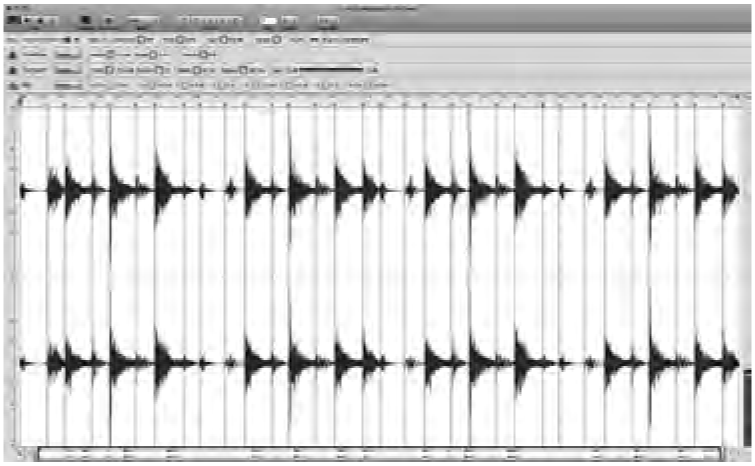
If we try to look at ways to reduce the number and severity of these artifacts, then we have to consider one important fact. In general, with any complex process, there is a trade-off between speed and quality. If we want to do something in real time, then we almost always have to compromise on the quality of the result. And time-stretching is a very good example of this. Computing power has increased rapidly in recent years, and it is only because of that that we can even consider doing things like time-stretching in real time. But processing power has yet to reach a point where the best quality that we have today for the stretching algorithms can be achieved in real time. And even if we reach a point where computers are powerful enough to do that, by that point there may well be even more advanced algorithms, which need even more processing power. So it is easy to believe that real-time processing will always be a second-best option when it comes to quality.

What would be a better option is to have the real-time processing, as we do now (which is actually pretty good), and all of the inherent flexibility of it, but then in addition have an option to bounce or render the file with the best-quality algorithms, once we have decided that we don't want to change it any more. There are a few synth plug-ins that are offering this option now (more-efficient CPU use during normal playback but a "high quality" mode to be used during bouncing), so there is definitely the ability to do it within the coding of the software, and, given that quality is always a concern when dealing with processes like time-stretching, it would be a good way to reduce that trade-off at the cost of an additional step of processing.

RECYCLING

Recycling uses a very different principle and is designed for a very different purpose. It still uses transient markers as a means of identifying timing references in the source material, but instead of manipulating the length of the audio between these markers, it simply cuts the audio file at these points and changes their timing from absolute (minutes and seconds) to "song position" (bars and beats). The original Recycle software (by default, anyway) created one container file that contained all of the audio "slices" and the timing information. There was, however, an option to export the audio information and timing information separately. The audio would be converted to a sampler patch, where each slice was mapped to a particular MIDI note, and the timing information would be created as a standard MIDI file, where each slice was triggered by the respective MIDI note at the correct time.

FIGURE 10.2
Propellerhead's Recycle software was the progenitor of many of the techniques we take for granted today. While it might seem quite basic compared to some of the technology that we have available today, it is still used a great deal, because it does a simple job very well, and sometimes there is no need for added complexity.



Given the explosion in popularity that Recycle experienced, it was inevitable that other software would come to the market that offered similar facilities. One of the limitations (if you would consider it that) of Recycle was that it was purely designed for slicing and then exporting the results for loading into another program or plug-in for playback. Some of the alternatives that followed combined the slicing features of Recycle with a playback system and wrapped the whole package up into a plug-in format. This meant that the whole process could be completed inside the plug-in, which greatly simplified the workflow. Given the importance of saving time without compromising on quality of results, clearly this all-in-one solution was a much better way to do things.

So, however we arrive at our destination, let's assume that we have our recycled file ready to use. What can we do with the file now that we have it? The main reason that we have recycled files is to allow us to change the tempo and timing of the files. In this respect it is actually very similar to beat-mapping but with one crucial difference. Because each "slice" isn't time-stretched in any way, there are no artifacts to worry about. Each slice sounds exactly the same after recycling as it does before. On the other hand, because the slices aren't stretched, there will be some potential problems if we move them or change the tempo.

When we move the slices further apart, there might be audible gaps between them. Just how much of a problem this will be depends on the source file. If we have a file with legato sounds or sound with long decays, then there is a chance that the sound would naturally run into the next slice if we create too many slices. Equally, at the beginning of any given slice, there could be the end of sounds from the previous slice in addition to the sounds occurring at the start of this slice. If the sounds in the loop have a very tight sound (closed hi-hats and a short-sounding snare, for example), then we might not have any problems.

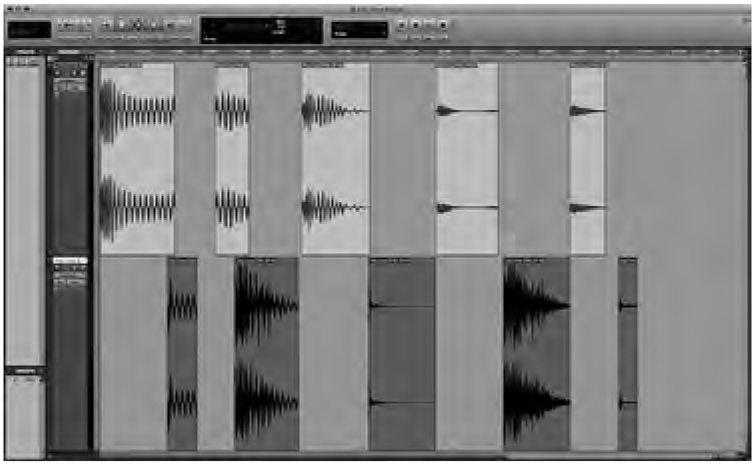


FIGURE 10.3
When an audio file is split at transient markers and then re-quantized, there can often be gaps or overlaps. The "slices" have been separated onto two tracks in the image above to make it easier to see.

The opposite is true if we move the slices closer together. The natural decay of the sounds in each slice might be cut short by the start of the next slice. There is perhaps more scope to deal with this issue, though, because you can just play the slices back polyphonically (more than one at once). In the case of a sampler playing back a .rx2 file, you would simply have to set the sampler's polyphony to a value of higher than one to allow more than one slice to play back simultaneously. In the case of the individual slices being on an audio track, any overlapping slices can be moved to a newly created (and identical in terms of plug-ins, routing, and settings) audio track to allow the overlap to be dealt with naturally.

You could also make use of fades to deal with both these situations. In the case of gaps between slices, it might be an option to apply a fade-out to the end of each slice that was followed by a gap. This could solve the problem of the slices cutting off abruptly, but what it wouldn't deal with is the presence of the end of those sounds at the start of the next slice. If this is the case, then you always have the option of selectively time-stretching certain slices to avoid this. The problem with this approach is that, if you are time-stretching only certain slices and applying fade-outs to others, the final result can sound a little disjointed. In this situation it may be a better option to either beat-map the sample or to manually stretch all of the slices to allow consistency across the length of the file.

In the case of overlaps, the situation with fades is a little easier, because all we would be doing is returning the level of each slice to zero before the start of the next region. It probably wouldn't be necessary anyway, as we have already seen that it is very easy to play the file polyphonically, but simple fade-outs at the end of each slice might help to clean things up if for any reason you can't utilize any of the polyphonic options.

EXTENDED USES OF RECYCLED FILES

So far you might be thinking that beat-mapping sounds like a better option, and you would probably be correct if all you wanted to do was clean up the timing of a particular sample. But the possibilities of recycled files go beyond simple timing correction. One of the other key benefits of recycled files is the ability to not only change the timing of each slice to make the timing more accurate but also change the order in which the slices play back. If you are working with a MIDI region and a sampler, then it is simply a matter of moving the notes to the new positions that you want them in. The fact that a MIDI note triggers each slice means that you can actually play a new groove on a keyboard to figure out what you want to change before actually moving the notes around. You can achieve the same end result if you are just dealing with audio regions, but obviously the process is a little less straightforward.

As well as rearranging the pattern that the file plays, there is another potentially very useful thing that we can do with recycled files. If we are using recycled files within a MIDI region and sampler setup, we can use the fact that each slice is independent as a way to change the dynamics of the pattern. Most samplers will, by default, have some mapping between velocity of the MIDI note and the volume of playback of the sample. You can always change this if you don't require it, but it can be useful in our situation. By changing the velocity of the individual notes within the created MIDI region, we can change the volume of each slice as it plays back. This won't give us complete flexibility, because if there are, for example, a kick drum and a hi-hat occurring at the same time and contained within a single slice, then changing the velocity of the relevant MIDI note will cause both the kick drum and hi-hat to change volume. So while recycling can certainly help to either increase or decrease the dynamic range of a loop without using any kind of compression or expansion plug-ins, it doesn't give us as much flexibility as we would have if we were dealing with a multi-track drum recording.

MORE CREATIVE USES

BEAT-MAPPING

So far we have looked at the ways in which beat-mapping and recycling were originally intended to be used. Like so many technological advancements, though, it doesn't take long for creative people to come up with new and interesting ways of using the underlying principles, and this holds true for both beat-mapping and recycling.

The main use of beat-mapping that we looked at earlier was to take an audio file and change the timing of that file to either change the tempo, make it more consistent, or radically change the groove of it. If we use it for that purpose, it can work incredibly well. But what if we really like the feel of a certain piece? Perhaps it has a certain "swing" to it that we like, and we want to take the other parts of our song (audio or MIDI) and make them fit to this feel rather than the other way round. Fortunately, we can do exactly that.

Because the beat-mapping process creates transient markers, and these transient markers exist in relation to the timing grid, we can use that information to create a quantization "groove" that, instead of simply moving everything to the nearest sixteenth note, will instead look at the timing of the audio file that we have beat-mapped and analyze the timing offsets between a theoretically perfect timing and the actual timing of the markers. Obviously these offsets would be defined in relative terms (0.03 of a beat late, for example) and could therefore be applied to any MIDI region to change the timing of that MIDI region to match the audio file.

If we want to change the timing of one audio file to match another, then we have a couple of options. If it is a relatively simple audio file (drum beat, bass line, rhythmic guitar part, etc.) then we might be able to recycle that audio file and then just apply the quantization groove we have created to the MIDI region of the recycled file. If it is something more complex, then we may need to approach it in a different way. To get the result we want here, we would need to go through a few different steps. We would beat-map the source file and create the transient markers as usual. We would then beat-map the second file to create the transient markers that we wanted to move. So far so good, but the difficulty lies in how we actually make them both line up. One way would be to cut the second audio file at the transient markers, then move each one manually to the position of the respective one in the first file (they are usually overlaid on top of the waveform display, so this isn't too hard to do), and then manually timestretch each split region to fill the required space. Basically, we are doing what beat-mapping does (moving transient markers and time-stretching to fill the gaps) but in a manual way. This will achieve what we want to achieve, albeit in quite a time-consuming manner.

Fortunately there is an alternative, and that comes in the form of elastic audio. This is a huge step forward for audio editing and production and is, in fact, such a big deal that there is a whole chapter dedicated to it that will be coming up shortly. I have mentioned it here because it uses the same fundamental principles of beat-mapping as we have described here but gives us, the end-users, much more control over how the audio is processed. It is certainly a very creative use of the basic beat-mapping principles, though, and is something that many people find indispensable nowadays.

Moving on from this, we can look at another related use of the beat-mapping ideas. In addition to the subtle variations of timing within a single bar that often exist with live performances, there can also be greater and more obvious tempo changes throughout a piece. These accelerandos and ritardandos can be very effective at adding to the emotion and feeling of a song. Choruses of songs not played to a click track are quite often at a slightly faster tempo than the verses, and the final chorus may be slightly faster still. There may be a subtle drop in the tempo at the end of a verse or a chorus as well to build a slight sense of anticipation. In fact, sometimes these changes aren't subtle at all and can be deliberately exaggerated. Beat-mapping can handle all these with relative ease and allow us to get rid of them if we need to.

In certain circumstances (a club DJ needing consistency of tempo, for example) this can be very useful, but, again, what if we want to work the other way round? What if we have an audio file that has an inconsistent tempo, but, instead of getting rid of the changes, we want to make sure that the tempo of the rest of the song follows the changes in this file? Once again, beat-mapping comes to the rescue. We can use these transient markers to create a tempo map that can then be applied to the current project. Once we have done this, any MIDI regions will automatically follow the tempo changes, but we should be aware that audio files, unless prepared specifically to do so, will remain at their original tempo.

To make sure that the audio files follow the tempo changes, we can either recycle them (if they are shorter files and are not likely to suffer from audible gaps or overlaps) or beat-map them. Once this is done, then everything should follow the tempo changes that you have created from your source file.

RECYCLING

When it comes to being creative with recycled loops, it is fair to say that your options are probably greater in number but less drastic in effect. As with beatmapping, you can use the location of the individual slices to create a quantization groove that can then be applied to other parts within the project in the same ways that we looked at for beat-mapping. If you have a MIDI and sampler setup for the recycled file, then it could, potentially, be even easier. You already have a MIDI region with the notes in the right place (in terms of the timing). If you look at that MIDI region, you will see that, unless you have already moved things around, each subsequent MIDI note will be one note higher up the keyboard, and each note length will fill the whole time until the next. This, as it stands, isn't very useful, but we can take those MIDI notes and, keeping their timing position the same, move them up and down the keyboard to create melodies and chords. We can change the lengths as appropriate, and we can mute any notes that we don't need. By doing this you will have a melodic MIDI part that is perfectly locked to the timing of your recycled part.

FIGURE 10.4
Building on the original Recycle concept, there are now a number of self-contained plug-ins that will not only analyze and split an audio file but also have a number of different options for playing back that file, with each "slice" having its own parameter set in addition to a number of master parameters.

Another thing you can do with recycled files is to modify the amplitude envelopes of the individual slices. This is considerably easier if you use one of the self-contained recycle-style plug-ins, as they generally offer more synth-like control of each slice (individually or globally) with amplitude envelopes (and sometimes filter envelopes too) rather than relying on fade-ins or outs. You can achieve similar results with fade-ins and -outs on audio regions, but it can take quite a bit longer, so this might be a consideration. The aim of this, whichever method you use, is to deliberately shorten the length of each slice or region with a fade-out or decay on the amplitude envelope. With drum loops, this can have the effect of making the whole pattern sound much tighter and more snappy and can be very useful to create a little more space in particularly busy arrangements without having to lower volumes or remove parts completely.

If you are using one of the Recycle-style plug-ins, then you could also have the option to have a particular loop very tight and snappy during a verse and then open it back out to its full sound in a chorus. This is simply a matter of adjusting a global amplitude envelope and setting the sustain level to zero and adjusting the decay time to give the required level of tightness. Then, when you want to open the loop back out to its full sound, you simply automate the sustain level from zero up to full. If you have set up amplitude envelopes on each individual slice, then it is also possible to open up certain elements within the loop individually. You could have one version of the loop where everything is tight, then a second version where you opened up only the snare drum hits, and then another version where the whole loop was opened back up to its full sound. Using these techniques can actually give you a lot of dynamic variation from a single source loop and could certainly be useful if you need to create interest from a limited selection of source files.

A variation on this technique is to adjust the Attack times (or fade-in times) on each "slice" or region to soften the individual hits. Again this is a way to introduce a variation to an otherwise static loop without changing its fundamental character. Simple changes like this can be very effective, as you can move a particular drum or percussion loop a little into the background by taking away the initial attack of the sound and then bring it back to the front when you need to. In many ways this technique is complimentary to the one above, and between the two you can do an awful lot to change the dominance of any given drum loop.

It's also possible to transpose either the whole loop or individual slices within the loop. Because the pitch changing in samplers or the Recycle-style plug-ins is generally based on speeding up or slowing down the sample rather than actual pitch-shifting, any change in the pitch will result in the length of each slice changing, which has all the same issues as we have already seen. But, in the event that the loop is suitable, some interesting effects can be had from playing around with individual pitches within a loop. If the loop is of a musical nature anyway, then changes to the melody of the loop might be possible in a very quick and easy way.

QUICK AND EASY MULTI-SAMPLES

While we are on the subject of melodic parts, you can also use Recycle to create quick "multi-sampled" patches for your sampler. If you have a particular combination of sounds from various synths that you like to use together, perhaps with associated compression and EQ, then you can quickly turn this into a (basic) multi-sampled sampler patch. First create a MIDI region for each of the synths that plays—in sequence and for an equal amount of time—each note, from the lowest to the highest that you want to include. The MIDI notes should have a long-enough gap between them, so that the release phase of the previous note is completely finished. Next you can bounce the combined result of all the synths playing this part. This resulting audio file is then loaded into Recycle or a Recyclestyle plug-in, and, as long as the sound doesn't have a very slow attack, the software should pick up the start of each note as an individual transient. Subsequent notes will then be mapped to subsequent keys, and, perhaps with a little bit of setting up which MIDI notes the exported MIDI region should start on, you have a playable multi-sample of your combined sound. This method isn't anything approaching the best way to do this. But in instances where you need to do something quickly, it is always good to have a few little time-saving tricks up your sleeve.

RHYTHM FROM ANYTHING

The last thing we are going to look at here is a way of creating rhythmic parts out of sounds that are not percussive in nature. This isn't something that you will do all that often, and it is perhaps more limited in its usefulness compared to the other things we have looked at, but, given that it uses a similar process to the one we have just described for creating multi-samples, it seems worthwhile mentioning it here. The basics are very simple: you take an audio file (one with sounds with sharp attacks and gaps between the sounds will work best here) and then recycle it and have each separate note/word/sound mapped to a different key. It doesn't particularly matter if the key that it is mapped to is the same actual note as the pitch of the sample, as we aren't intending to use these slices to play conventional melodies. Once this is done, we now have a multi-sampled patch with different notes, words, or sounds on each key. We can now adjust the amplitude envelope to have a sharp attack, fast decay, and zero sustain, so that each sound has the kind of envelope you would expect from a drum or percussion instrument.

Then it's really up to you how to use it. You can either just play along to your rhythm track and get a feel for a groove by playing random notes, or you can take the MIDI region from another recycled file, copy it to this instrument, and then move the MIDI notes around to create a new groove that matches the other one in timing but is composed of pseudo-random snippets of vocals or other musical sounds. It may be that you want to change the pitch of some of the sounds to help them sit better melodically, or you may wish to shorten the decay time down even further, so that the pitch becomes less relevant and obvious.

As you can see, there are many ways in which you can use these tools outside of the obvious. Beat-mapping applications tend to be wider in scope and have more of an effect on the feel and groove of a song as a whole, whereas the creative uses of recycling tend to be more effective on a smaller scale. Using beat-mapping and recycling in combination, it is possible to really change the feel of a track in quite natural-sounding ways and ways that may have your clients wondering how on earth you managed to get it sounding like that.

HANDS ON

INTRODUCTION

In addition to the benefits of having transient markers in place for general editing, they also serve as a basis for a number of other more-creative editing tasks. Two of the most obvious of these are beat-mapping and recycling. In this section we will take a look at the ways in which each of our DAWs uses the transient markers to help with these increasingly useful tasks and what tools are available to help us get the most out of our audio files.

LOGIC

One of the most common uses of the transient markers within audio files is to enable quantization of audio files. These transient markers, in conjunction with *Flex Mode*, allow audio regions to be treated in the way that MIDI regions have been for a very long time and have a simple quantization parameter applied. Once the transient markers have been placed in the Sample Editor, you can turn Flex Mode on for the region in question, and the *Flex Markers* should appear on the waveform display of the region in the Arrange window. It might be that some of the transient markers (shown as pale vertical lines in the region in the Arrange window) haven't been converted to Flex Markers (brighter lines), and, if this is the case, you simply need to click on the transient marker to convert it to a Flex Marker. Equally, if there is a Flex Marker that you don't wish to have, you can double-click on it to remove it. Once all the Flex Markers are in place (in the sense of the markers for the transients you wish to quantize) then you can adjust the quantization simply by changing the *Quantize* parameter in the *Inspector* panel. You may need to try out different *Flex Algorithms* in order to get the best result, but the options are quite self-explanatory, and the best choice is usually obvious.

If, on the other hand, rather than quantizing a particular region to a fixed quantization feel, you wish to extract a quantization feel (or *Groove Template*, in Logic terminology) you can use the Flex Markers to do this. Select the region that you wish to use as the template, and then, in the Inspector panel, click on the Quantize drop-down list. At the bottom there will be an option called *Make Groove Template*. This extracts the timing subtleties from the currently selected region and saves this as a *Groove Template*. If you then choose a different region that has Flex mode enabled and then go to the Quantize drop-down list, you will see the name of the previous region below all of the default quantize options. Select this option, and the current region should have the feel of the previous region applied to it. The exact quality of the results depends on the initial similarity of the two regions. If the source is a busy sixteenth-note hi-hat pattern, and the target is a more sparse and loose tambourine pattern with a triplet feel, then the results might not be 100% as hoped, but, in the majority of cases, this is a quick and easy way to match the feel and timing of two separate audio parts. It is worth noting that this Groove Template can also be applied to MIDI regions in order to really lock together the groove of, for example, a drum loop and a bass line and can be incredibly effective at tightening up the timing of a rhythm section. Finally it is also worth noting that in order for the Groove Templates to work, the regions that they were taken from have to remain in the arrangement (even if they are muted).

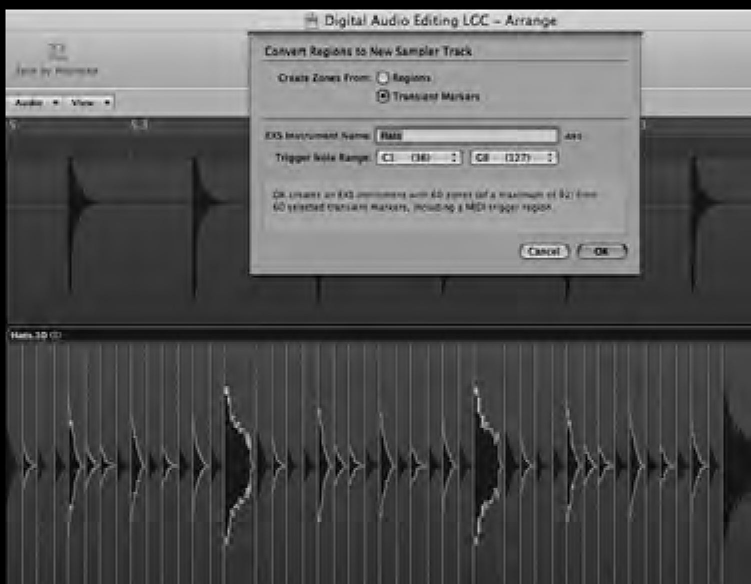


FIGURE 10.5
Logic makes it extremely easy to take an audio file and create a recycled version of it. Pretty much any audio region or file can be analyzed and split and a sampler instrument created and ready to play in just a few mouse clicks.

Another very useful feature of Logic is the ability to automatically recycle audio files based on the transient markers created either in the Sample Editor window or automatically, using Flex Mode. Once the markers are in place, you can convert any audio region into an *EXS-24* sampler instrument and companion MIDI region. Assuming you have an audio region with transient markers in place already, and you have made sure that markers exist only for the slices you wish to create, all you need to do is make sure that audio region is selected in the Arrange window and press Ctrl + E, and the *Convert Regions to New Sampler Track* dialogue box will appear. You are first asked to make a choice as to whether the *Zones* (what we have been calling slices) are created using regions or transient markers. In the context of a single audio region with transient markers, we would obviously choose *Transient Markers*, but, if you have already split a region into individual parts, you could select Regions instead. You then have the option to enter a name for the *EXS-24* sampler instrument, but the default value is the name of the audio file being converted. Finally you have the option to choose a MIDI note range that you wish to use for the resulting MIDI file. Unless you are creating a chromatic instrument, there is no real benefit to changing the default values. Once you click on OK, the audio region will be analyzed and each section between transient markers will be allocated to a single MIDI note in the EXS-24 sampler instrument, a MIDI file created with the timing for each of the zones/slices and the original audio region muted.

Using this method you can, in just a few clicks, convert an audio region into a recycled file, and this has a huge number of uses, as we have seen in the preceding chapter. It isn't all good news, however, as there are compromises made in doing so, depending on your intended use. As we saw in the main chapter, a recycled file is great if you wish to create new variations on the original audio part by changing the order or timing of the MIDI notes, but it is only partially tempo-independent. Of course the initial timing of any MIDI notes will follow the tempo of the song but, unlike Flexed audio, the duration of each slice will stay the same, which can lead to overlaps (which are fairly easy to deal with) or gaps (which are not) depending on which way the tempo changes.

With this in mind, I would suggest that any tempo changes should be figured out before doing any recycling, but of course this isn't always practical. You also get all the other advantages of recycling (ability to change pitch and envelopes, etc.) so it is really just a question of deciding the best time to switch from Flexed audio to recycled in order to achieve what you need to. If it really becomes a problem, then you can always make the changes to the MIDI region to create the variations that you want at the original tempo and then quickly bounce the result using *Bounce In Place* (Ctrl + B) back to a new audio file, which can then be Flexed and follow any tempo changes.

PRO TOOLS

For Recycle-like beat slicing, Pro Tools comes equipped with the very flexible *Beat Detective*. If you wish to create a "sliced" audio file, then you start by choosing a region that you want to analyze and then open Beat Detective by pressing Cmd[Mac]/Ctrl[PC] + Num Keypad 8 or going to Event > Beat Detective. The Beat Detective tool has a number of different options, depending on what you wish to achieve, but the relevant options here are *Groove Template Extraction, Clip Separation, Clip Conform*, and *Edit Smoothing*, and we will look at each of these in order, but before we do we need to put the markers in place that Beat Detective will use for the first two options.

You first need to choose either Groove Template Extraction or Clip Separation from the *Operation* heading, and then, if you preselected a region before opening Beat Detective, the *Start Bar* and *End Bar* under *Selection* should match the region you selected. If you didn't choose a region, or if you wish to process a different region, simply select the region and then click on Capture Selection at the bottom of the Selection section, and this will update the range above.

FIGURE 10.6
Beat Detective is a great option if you wish to re-quantize audio files or create Groove Templates. There are a number of different pages that are chosen on the left of the window, with each one relating to a different part of the whole process.

Moving on to the *Detection* section, we find three main controls. The first of these, *Analysis*, has three different options. Both *High Emphasis* and *Low Emphasis* focus on these specific frequency ranges, while *Enhanced Resolution* uses a different analysis method and in most cases is the better option. Once you have chosen an option here, click on the *Analyze* button to calculate the transient positions. Sensitivity is a simple slider that will increase or decrease the number of transients.

Below this are the *Resolution* controls. Choosing *Bars* will place markers on beat one of each bar and nowhere else. Equally, *Beats* will place markers at the start of every beat, but the nuances of the timing between beats won't be marked. Finally we have *Sub-Beats*, which will place markers all the way down to sixteenth notes. You can make further adjustments by manually deleting the markers by making sure the Grabber tool is selected, then moving the cursor over the marker you wish to delete, and holding down Alt while clicking. You can also reposition markers by using the Grabber tool and then clicking and dragging the markers to the new location.

Now that the analysis is done and the markers are in place, we can move on to the different processes that Beat Detective can offer. The first option, *Groove Template*

Extraction, uses the position of the markers and creates a "timing map" that can then be applied to the timing of other regions to enable us to lock the timing of two separate regions together. Making sure that Groove Template Extraction is selected from the *Operation* heading, simply press the *Extract* button in the bottom right-hand corner of the window, and you will be asked to confirm the length and time signature of the extracted groove; have a text box where you can enter comments. You can then either *Save to Groove Clipboard* or *Save to Disk*. Saving to the clipboard is a temporary option where saved groove will be overwritten next time you carry out the process or lost when you close Pro Tools. Saving to disk, on the other hand, allows you to specify a file name and location to store the template for future use at any time.

The next major task that Beat Detective can perform is to separate an audio region into "slices" like Recycle does. In fact, if you have already worked on getting the markers in the right place as described above, then the only thing you need to do to carry out the separation is to choose *Clip Separation* from the *Operation* heading and then click on the *Separate* button in the bottom right-hand corner. Your one audio region will now be separated at the marker points into a number of individual regions.

If you then click on *Clip Conform*, you will see, under *Conform* on the right of the window, four controls. *Strength* allows you to vary the degree by which your regions are conformed (quantized). A value of 0% will leave them in their current position, while a value of 100% will move them to be on the quantization grid. The *Exclude Within* slider allows us to exclude certain regions from the conforming process. If a region is close to its ideal position, then it can be excluded from the process to make sure that only the most out-of-time regions are moved. The Exclude Within slider controls the amount of variation from the ideal that is allowed before confirming will take place. Finally, the Swing slider allows us to add a variable amount of swing to the basic conforming process, and we can select either eighth-note or sixteenth-note swing using the two buttons located below the slider. Clicking on the *Conform* button in the bottom right-hand corner will apply the process and move the regions into their new positions.

Finally we have the *Edit Smoothing* option. When you separate a region, as we have done here, and then move (conform) individual slices within that region, it is quite possible that there will be audible gaps between the end of one slice and the start of the next. Pro Tools has included the *Edit Smoothing* option (under the *Operation* heading) as a way of trying to deal with situations like this automatically. Fill Gaps, the first option under the *Smoothing* heading, works by moving the start point of any regions that are preceded by a gap back to the end of the preceding regions. In the event that there are still issues with the smoothness of the changeover points, then you can change to the *Fill And Crossfade* option, which will carry out the same process but additionally will apply cross-fades at the junctions of all regions. The duration of the cross-fade can be set in the *Crossfade Length* box.

Moving on from Beat Detective to *Elastic Audio*, we find that one of the most useful things it allows us to do easily is to quantize an audio file or region. First, we need to select all regions that we wish to quantize, make sure that Elastic Audio is active and that the markers are in the right places (as per the previous chapter), and once that is done, we open up the Event Operations dialogue by pressing Alt + 0 or Event > Event Operations > Quantize. *What to Quantize* should be set to *Elastic Audio Events*, and we can leave this as it is and then move on to the actual *Quantize Grid* options. The main drop-down list gives options for all the standard note divisions as well as dotted notes and triplets, but in addition has options for *Groove Clipboard* and a list of any saved templates. If you choose one of the standard note divisions, then you also have options for *Randomize* (which will quantize notes but then apply a degree of randomization to the timing to try to maintain an element of human feel), *Swing, Include Within, Exclude Within*, and *Strength* (which are all similar to the options in Beat Detective). If, alternatively, you choose a groove template, you have options for Pre-Quantize (which applies a standard quantization before applying your groove template), *Randomize, Timing, Duration*, and *Velocity*. The last two of these are relevant only to MIDI regions, where they can match both note durations and note velocities as a part of the quantization process, but the *Timing* slider determines the strength of the quantization process, with higher values having a stronger effect. Any change in parameters will require you to hit the Apply button again, but it is generally quite a quick process, so this shouldn't slow you down too much.

One final way in which Elastic Audio can help us to create pseudo-Recycle files is by setting the *Track View Selector* to *Warp* and then selecting a region and using the Tab to Transient feature to move through the region one transient at a time, and then, at each new transient, use the Separate at Selection command (Cmd[Mac]/Ctrl[PC] + E or Edit > Separate Clip > At Selection). This is an alternative to the Beat Detective method of splitting a region into several smaller slices.

USING RECYCLED LOOPS

One additional use of recycled files that we haven't looked at so far is the possibility to take the timing information that exists within a recycled file and use that to trigger other sounds to reinforce or replace individual sounds within the file. Naturally, we could do this with a normal audio region as well. We could identify the regions or places within regions where a certain sound occurred and then create a new track and copy the replacement sound into each place that it was needed. But over the course of an entire song, this could be a very laborious process. Recycle and its ability to create MIDI regions with individual notes separated and placed at the correct times go a long way toward automating this process.

If we have a drum loop that we like but that we feel has a snare drum that is a little weak, then we simply recycle that drum loop and drop the MIDI file onto a new sampler instrument track. We can then load up a snare drum sound that we want to use to layer (or replace) the weak snare drum with and assign it to a MIDI note of our choosing (let's say we choose C3). We then locate the notes in the MIDI file that correspond to the weak snare drum, move these notes to the correct note (C3 in our case) to trigger the new snare drum in the sample, and then mute all the remaining notes. We will now have our new snare drum layered with our original one.

We may still need to do a little fine-tuning in terms of the timing if we want to layer the two sounds, simply because the transient marker that we used to "slice" the original file when we created the MIDI region may not have been 100% at the start of the snare drum. Or perhaps it was, but the start point of the new sample isn't 100% at the start of the sound. In either of these situations, there may be a slight timing discrepancy between the two. If this happens the easiest solution is to apply a MIDI timing offset to the new snare part (either a positive or negative offset, depending on what is needed) to get the two sounds to work well together. You might also need to consider the phase relationship between the two sounds. Even though they won't be identical and therefore will never completely cancel each other out, there could a great deal of cancellation between the two sounds, so either inverting the phase of one or the other or very subtle shifts in position can fix that.

This system works well if you want to work with short loops and layer an additional sound or two on top, but if, on the other hand, we are dealing with a multi-track drum recording with each drum recorded individually and with variations throughout its length rather than being a constant repeating loop, then things get a little more complicated—not so much complicated in principle but more in execution. We can still use the same principle in that we can still recycle a file that is an isolated snare drum, and we can use a file that is longer than a bar or two as the source, so that isn't a problem. The real diffi culties lie in getting a "clean" snare drum (in this case) track so that we get only transients at the start of each snare drum hit and also in the fact that, given that recycle creates MIDI files for the notes and the MIDI specification allows a maximum of only 128 notes, a recycle file can never contain more than 128 separate "slices" and associated notes.

The first of these, the potential problem with getting a clean track, can be dealt with, and we have already looked at ways in which we can do this, so it's quite possible that you have already been through this process, and, having done so, have come to the conclusion that the snare just isn't right. If that's the case, then half the problem is already solved. If you haven't reached that point yet, then you can refer to the earlier chapters on how to do this. Once you have the snare drum in a reasonably clean state, you also have the sensitivity control to adjust to try to make the slices occur on only the snare drum hits and not because of any residual noise or spill.

The other issue—the 128-slice limitation—is more of an inconvenience than a problem. The solution is to create multiple recycle files to cover the length of the song. If we assume a very "straight" drum pattern with two snare drum hits per bar (on beats two and four), then a single recycle file could cover a maximum of sixty-four bars, which is quite a substantial amount of time and would mean that you would most likely need only a couple of recycle files to cover the whole song. Now of course, the more snare drum hits you have per bar (with a more intricate snare-drum pattern), the smaller number of bars a single recycle file will cover. But in all but the most consistently complex of snare patterns, you shouldn't need more than a handful of separate recycle files to cover the full song.

In the event that you do need to create multiple recycle files, you would still need to create only one new sampler instrument, as the same sound would be used by all the recycle files. You would just take the MIDI region from each recycled file and move it to the same sampler instrument track. This would be true whether you needed one recycle file or twenty. The only time you would need different sampler tracks would be if you wanted to use different sounds to layer with the original sound in different sections.

Although I have used the example of a snare drum, this technique is equally applicable to all drum and percussion sounds (all sounds, in fact, but it is most commonly used with drums), but consideration will need to be given to the number of individual slices you will end up with. A snare drum (or kick drum) with a few hits per bar could mean only a handful of recycle files to cover a whole song, but a busy hi-hat pattern could result in a single recycle file covering only eight bars or so, which means a lot more recycling to cover the whole song. Even in this case, though, it still might be far quicker than manually layering sounds.

Using this method can be quite a lot quicker than manually placing new audio files to layer with existing ones, but it also gives a much more-simple way of controlling the volume of those sounds. Now, when I say "volume," I am referring to the volume dynamics over a bar (or two or ten). Being able to use MIDI note velocity to change the level of the layered sound rather than having to draw in volume automation for each individual hit is a great time-saver. And, as we are using sampler instruments, it is also possible to map MIDI controllers to adjust the volume as well, so a modulation wheel on an attached MIDI keyboard could be used as a volume control for the layered snare drum. This might end up being a more natural way of controlling the dynamics, because it could be done by "feel" rather than just drawing lines or entering numbers. And as well as this, there is still the option to control the attack or decay times of the sampled sound, as we looked at in the last chapter.

By using this technique we can (relatively) quickly layer up a drum sound and have a good amount of control over that layered drum sound. With the right samples used, it is certainly possible to create some very natural-sounding results, but it might take a little time to record MIDI controller movements to re-create the dynamics and tonal changes of the original sound. Clearly it would be useful if we could remove some of the manual work from this to speed things up.

DEDICATED DRUM REPLACEMENT TOOLS

Given that we have a means of detecting when a transient has happened, a way of converting that into a "trigger" signal of some kind, a method of measuring a peak level of a given part of an audio file, a way of loading a sample (or velocity-switched group of samples) into a sampler, and a means of using a trigger signal to sound the sample(s) we have loaded, we already have everything that we need to make the process described above much easier. We have achieved most of these steps with the method described above, but it would be nice to be able to make the process somewhat more intuitive (and quick) by combining the various stages into one single tool.

Most DAWs have transient detection built in as a part of some other process (beat-mapping, time-stretching, etc.) but not all of them use this in the same way to aid with drum replacement. So in this chapter we will look at some third-party tools that are going to be DAW-independent and would be available to everybody. All these drum replacement tools work on the same principle of analyzing an audio file, detecting when a trigger situation has happened, and using that to trigger an alternative sound within the plug-in itself. There are often options to create MIDI tracks from them as well, but, for the purposes of this discussion, I want to look at the all-in-one solution. If you were to use one of these drum-replacement plug-ins and generate a MIDI region from it, then the resulting MIDI region would be used in much the same way as the MIDI regions created from recycling, so we would have all the same options and possibilities as we have already looked at above. But let's see what these all-in-one solutions can do for us and our workflow.

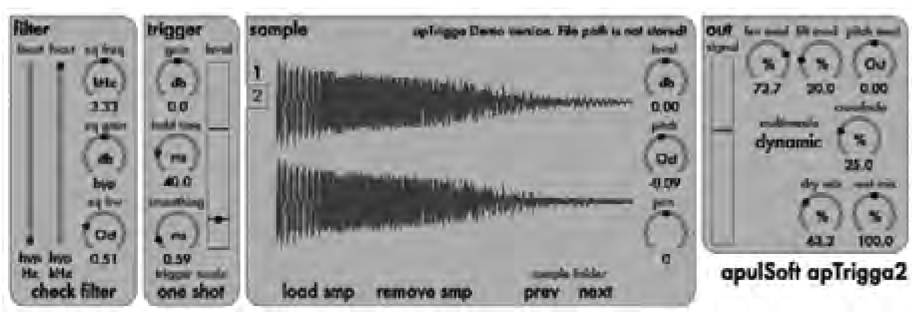


FIGURE 11.1
apulSoft's APTrigga is a simple yet effective drum-replacement plug-in that wraps up detection tools and sample playback into a single plug-in.

The first thing we need to consider, before we get into the details of how these drum replacement plug-ins work, and what they can do for us, is the fact that, unlike the recycled file solutions that we have just looked at, these plug-ins operate in real time. They analyze the audio as it is playing, and, when they detect a transient, they trigger the replacement sound. The recycled file version analyzes the file ahead of time and creates its transient markers once the analysis has been done. Because of this, the associated slices or MIDI notes will be 100% at the same time as the transients occur. With the drum-replacement plug-ins, there will always be a latency (or small delay). It doesn't matter how fast or powerful your computer is—this delay cannot be avoided, only minimized. Naturally, the more powerful your computer is, the less this delay will be, but, if you aren't actually replacing sounds and are layering them instead, we have already seen that even the smallest delays between sounds can cause phase-cancellation issues, so this could cause more problems than just a slight "looseness" in timing.

There are two simple ways to deal with this problem, and both relate to shifting the timing of the replaced drum sound relative to the position of the original sound. The first, and the simplest, is to simply work with the slight timing lag until you are happy with the replacement sound (timing, dynamics, tone, etc.), and then bounce the replaced sound to a new audio file and nudge the timing of this bounced file a little earlier, until it is in time. This is an entirely workable solution if you are actually replacing one sound with another, because there will be no issues of phase cancellation between the two sounds, so, at worst, the timing will be a fraction off while you are still adjusting parameters prior to bouncing the replacement sound.

FIGURE 11.2
If the detection process means an unacceptable latency, then you can always offset the track used to trigger the detection. Here we can see the original snare drum track (at the top) and the offset copy, which is moved a few milliseconds earlier to counteract the delay in the detection process.

The second option is to actually offset the timing of the replaced sound before it is triggered. This might sound a little contradictory, but it is possible with a little bit of work. We have seen that there is an inherent delay in the transient detection and triggering process, so, if we are aware of this, we can get around it by creating a new audio track, copying the audio file or region that we are processing with the drum replacement plug-in to this new track, loading the drum replacement plug-in and setting everything up as we need it, and then, once we become aware of just how much the delay is, we can move the actual "source" file a little earlier to compensate for the processing delay in the plug-in. If you are doing simple drum replacement, then you should mute the original, unmoved "source" file. But if you are layering a new sound, then this method allows you to move the timing (by moving the copy of the source file used to trigger the drum replacement) of the layered sound in real time without having to bounce.

The actual amount you have to move it by will vary from plug-in to plug-in, from computer to computer, and even from one replacement sound to the next. There is no arbitrary "global" offset that you can just apply and know that it will work. It is a matter of experimentation, but, given the ease of selecting an audio file and fine-tuning its position, it isn't something that should take a huge amount of time to get right. And if, as mentioned above, you are doing full replacement of a sound rather than layering, you will need to make sure the timing is accurate only to a couple of milliseconds rather than potentially having to move the "source" file by a few samples at a time to correct a phasecancellation issue. Now that we have that disclaimer out of the way, it's time to look at how these plug-ins work and how we can use them to our advantage.

DRUM REPLACEMENT TOOLS IN USE

There are two main parts to plug-ins of this type: the detection and the replacement. The specific features available are slightly different on each plug-in, but they share at least a few common characteristics. The main controls on the detection part of the plug-in are an input gain control and some kind of sensitivity control. The input control is used to make sure that we can maximize the dynamic range of the input sound as much as possible. Most often this gain control will be used to increase the level of the input, but, on occasion, it might be necessary to reduce the gain of an input if, in addition to the underlying audio file (which can never go above 0dB), there has been additional processing that has actually pushed the input to this section above 0 dB. In this case, having the level set so that it is clipping would actually move into the area of actually reducing the amount of available dynamic range. Once the gain has been set correctly, we can move on to the sensitivity control.

In order to set the sensitivity accurately, it always helps to have a fairly "clean" signal, for the same reasons that it helps in recycling files, and to that end some drum replacement plug-ins offer additional controls to help achieve this. These could take the form of simple high- and low-pass filters to tune the frequency used to detect the transients, or could be more advanced, such as a transient designer that can help to emphasize the attack and suppress any background noise or spill or just make the overall sound more snappy and tight. Of course, even if the drum replacement plug-in doesn't actually have any features to clean up the input signal, or if the features present still aren't tightening it up sufficiently, you always have the option of additional editing or processing prior to the input of the plug-in.

Adjusting the sensitivity can be quite deceptive. After all, with very minimal controls, how hard can it be? The answer to that question is that it isn't hard, but it can be time-consuming. It can take a lot of adjusting back and forth between those relatively few parameters to get something that captures the subtlety of the drum performances but doesn't trigger notes in error from background spill. To help with this, there is sometimes an additional "re-trigger" control. This is very simply an amount of time that must pass once a trigger has happened before the next will be allowed. If you set it to forty milliseconds then, no matter what happens to the input signal, there will not be a trigger signal created until at least forty milliseconds have passed. If you know you have a pretty simple pattern with no intricate "ghost" notes, "flams", "drag rolls", or super-fast playing in general, then you can probably set this much higher to decrease the chances of false triggers, even in a recording with quite a lot of mess and spill.

FIGURE 11.3
The SPL DrumX-changer is a very well-specified drum replacement plug-in that includes not only some very advanced detection circuits but also SPL's Transient Designer technology on both the original and replaced parts of the sound and a library of replacement sounds to get you started.

Moving on to the actual replacement part, we can see that things, however they are laid out or implemented, are basically quite self-explanatory. Fundamental here is the ability to load in samples to be triggered, and all the plug-ins allow you to do this. Many will come with their own library of sounds covering all of the common drum kit pieces, and many of these are multi-sampled, so that the sampled sound will not just simply change volume but will have some of the tonal change that can happen with changes of loudness in a real drum. It would be unrealistic to expect a static collection of samples to fully reproduce the tonal and dynamic range of a real drum, especially a snare drum, but as technology is advancing, larger and larger sample collections are becoming the norm, and the results are getting harder and harder to distinguish, especially if the original performance doesn't utilize a huge amount of the tonal and dynamic spectrum of the drum.

The specifics of how the samples are loaded, whether they are multi-sampled, whether you can load in more than one layer of sounds, whether there is any volume envelope editing, and whether there are then further processing effects within the plug-in will vary from product to product, and, if a particular feature is important to you, then that could be a determining factor in your choice as to which to use. Most of them do, however, allow you to use your own samples (or those from another sample library or source), but, as I have already mentioned, don't expect to get results that sound as good if you load in a single snare drum sample and expect it to represent the snare drum accurately across all volume levels.

One very clever feature is the use of "round robin" sample playback. This entails the use of multiple samples for each velocity level, meaning that, in addition to the sound of the sample changing—not only in volume but also in tone as the volume/velocity changes—subsequent triggers, even if at exactly the same level, will cycle through these alternate samples, so that no two hits in succession are ever exactly the same. Although the differences between these roundrobin samples can be very subtle, the overall effect, over the length of a song, is an indefinable sense of added realism, and, best of all, all of the hard work is done for you in the creation of these sounds, so you can just set up, sit back, and enjoy! Of course things are never quite that simple, but we are getting closer every day.

Once you have chosen the sound you want to use, you may need to revise your settings for sensitivity and other parameters, but one thing that you will almost certainly want to do is adjust the dynamics of the new sound. The dynamics, in this context, are the rate at which the volume of the new sound increases in proportion to changes in the volume of the source sound. By reducing the dynamic correlation, you can set things up so that, even if your original source sound has a lot of variation in volume, this new sound always hits pretty hard. Alternatively, you may wish to take an original source sound that didn't have a great deal of dynamic variation and try to actually increase that. You might want to simply try to keep a linear relationship between the two, or it might be anywhere along this spectrum. The ability to actually change the dynamic correlation between the source sounds and the new sound means that, if you are mixing the new sound with the source sound, you can set things up so that the new sound really becomes apparent only on harder hits but is still there on the quieter hits; it is just that the new sound is more "quiet" than the source.

This brings us on to the final selection of parameters: those concerning the overall balance between the original source sound and the new sound, and the overall output level of the plug-in. Almost all the plug-ins offer a balance control that is continuously variable from only the original source sound through to only the new sound. If you have created a copy of the source sound and applied the plug-in to that track to give you the ability to nudge the timing backward or forward (as discussed above), then you can set the balance within the plug-in to be only the new sound, then adjust the balance between the source and replacement sounds in your DAW and apply separate processing or effects to each sound (something you can't do if you mix the two sounds within the plug-in itself).

That pretty much covers how drum replacement tools work from a functional point of view. Whether you would want to use these tools is really down to how you like to work and just what scope your editing job covers. There are some pretty big names from the music world endorsing some of these plug-ins, and they can certainly be a great problem-solver if you have some truly shocking recordings to work with. Or perhaps the recordings you have really aren't all that bad at all, but either the drum kit itself or the recording equipment use has just resulted in a drum track that, though perfectly competently performed, just lacks any real depth and punch. In cases like this, it helps to have a working knowledge (at least) of production techniques, as it will give you a better idea of what might be fixable by working with the original recordings and what just needs a little help and support tonally.

HANDS ON

INTRODUCTION

In the main chapter, we have looked at the ways in which recycled files and drum replacement plug-ins can be used to either reinforce or completely replace a drum sound that, for whatever reason, isn't right. Many of the drum replacement plug-ins have a number of advanced features, and many also come with custom sample sets that can hold a lot of appeal. If you don't have to do drum replacement that often, you might not feel justified in investing in these plugins, so we will take a look at the options available in each of our DAWs and compare them to using a third-party drum replacement plug-in.

LOGIC

Logic's built-in drum replacement tool, while fairly basic, can often be sufficient for simple drum replacement duties. Like all drum replacement tools, it will work much better when there is a relatively clean recording (not too much of the sound of other drums in the mic), but, because the detection options aren't as complex as some third-party tools, it is perhaps a little less forgiving. Nonetheless, it is a great place to start and may well be all you need if drum replacement is something that you do only occasionally.

The first thing to do is solo the track you are going to be working on and then go to Track > Drum Replacement/Doubling. Alternatively, you can use the key command Ctrl + D to open the dialogue box. As soon as you select this, you will notice that an Instrument track is created below your audio track and an *EX S-24* sampler loaded, ready to serve as the source for the replacement/doubling sounds. The first option that you are presented with is a choice of *Instrument*. The main effect this has is to change the default MIDI note that is used for the MIDI region. If you choose Kick, the note will default to C1, Snare will default to D1, Tom will default to A1, and Other will default to C3. Therefore, if you choose the wrong option, or you are replacing a different drum not listed, it isn't a huge deal, as you can change the MIDI note either later in the process or even after the MIDI region is created. It should also be noted, however, that it does also have an effect when you are using the *Prelisten* option (more on this below).

Next up is the *Mode* option, which can be set to either Replacement or Doubling. The only difference here is how Logic deals with your source audio track once the process is complete. If you choose Replacement, then the source audio track/regions will be muted, while if you choose Doubling, they will, obviously, remain active. But once again, whichever option you choose can be amended very easily after the process is completed by muting/un-muting the source audio track as required.

The next option, *Relative Threshold*, is the most important one, as this is what Logic uses to determine what is an actual hit that needs replacing/doubling and what is unwanted noise/spill. You can make adjustments either by using the slider, the up and down arrows on either side of the threshold numerical display, or by double-clicking on the threshold number and typing a value. It is useful to know that the up and down arrows can work with different increments. By default they will increase or decrease the threshold by 1 dB at a time, but if you click on a single digit of the threshold reading, it will become highlighted, and a small arrow will appear underneath it. Once this small arrow is visible, the increment of the up and down buttons will change, and only this digit will increase or decrease by a single number each time. For example, if there was a current threshold of –13.2 dB, and you clicked on the "2," then the up and down arrows would increase or decrease the threshold by 0.1 dB at a time; if you clicked on the "3," then it would be 1 dB at a time; and if you clicked on the "1," it would be 10 dB at a time.

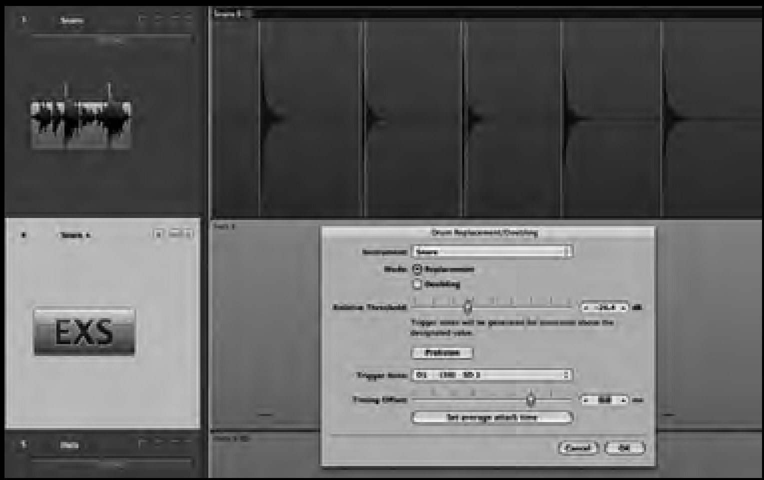


FIGURE 11.4
The built-in drum replacement tool in Logic can be quite effective, but care must be taken with "double triggers," as shown in the example above. Adjustment of the threshold level may not be able to remove these, in which case you can go into the resulting MIDI region and delete them manually.

As you adjust this threshold, whichever method you choose, you will start to see vertical yellow lines overlaid on the waveform display in the Arrange window. These lines represent the transient markers that the process will use. Lowering the threshold will create more markers, and raising it will create fewer. Many times a careful adjustment of this threshold level will create markers in all the places that you need them, but it isn't perfect. Sometimes markers won't be created where you expect them, and adjusting the threshold won't have the desired effect. In many ways it is a shame that you can't use the transient markers that you may have already placed by hand, because these would be in the right places. Instead you have to rely on the threshold method and hope that it works accurately enough.

In the event that no amount of tinkering with the threshold gives you everything that you need, don't worry, because all is not lost. You can always continue with the process and manually place additional notes in the MIDI region to cover any missed notes or delete any unnecessary notes. Of course you will have to estimate the velocity value, and having to do this manually defeats the purpose of a drum replacement tool, but at least it is a work-around if you are not inclined to purchase a more advanced third-party tool that has more options and may be able to do a better job.

Once you think that you have got the threshold level set correctly (or at least as good as it can be), you can click on the *Prelisten* button, which will allow you to audition the settings that you have chosen and the effect they will have. Once clicked, both the source audio and the replaced/doubled MIDI tracks will play, allowing you to hear if the settings you have chosen are appropriate. And here the *Instrument* setting that we mentioned above comes into play. If you chose Kick, then a kick sound will be loaded by default (and likewise with Snare and Tom), but in any case you can navigate through the default Logic library or, indeed, load any *EXS-24* settings of your own choice if you have your own sounds you would like to use. If you wish to change the MIDI note that will be used for the MIDI region, you can do so using the *Trigger Note* drop-down box, and this may prove useful if you are loading your own *EXS-24* instrument that doesn't conform to the default note choices mentioned earlier.

Finally we have the *Timing Offset* adjustment. This is included to allow minor changes to the timing of the MIDI notes. Sometimes a sample may not be perfectly truncated, and the actual sounds may start a few milliseconds after the MIDI note event triggers it. This control allows for any fine-tuning to be made (in 0.1 ms increments) prior to the creation of the MIDI file. Alternatively, you can use the Delay parameter in the *Inspector* panel, once the MIDI region has been created. You may wish or need to do this if you decide to change the sample(s) used after the initial MIDI region creation. While this Delay parameter doesn't offer adjustment in milliseconds, it does offer a resolution of 960 ppqn, which means 960 pulses per quarter note. In real terms this means that however long the duration of one beat is (60/tempo), the minimum increment for changes using this method is 1/960th of a beat.

Once you click on OK, the dialogue box will be closed and the MIDI track created using the selected MIDI note for the triggers and creating note velocity values based on the actual peak level of the individual hits in the audio track. The audio track will also be muted if you chose to replace the sound and the MIDI part ready for further editing if necessary.

PRO TOOLS

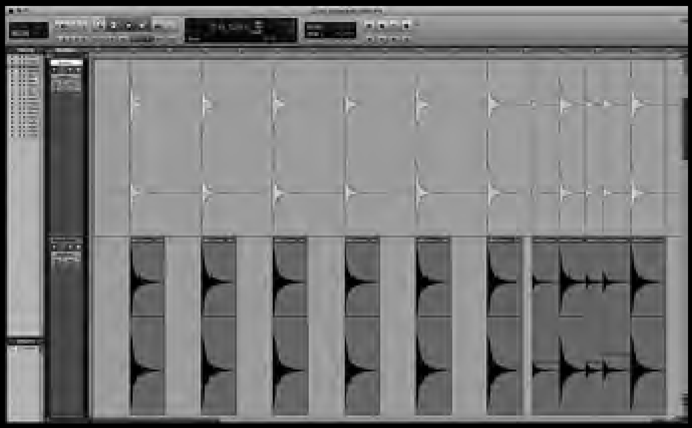
Unlike some other DAWs, Pro Tools doesn't feature any built-in drum replacement tools so, if you don't have access to any of the third-part drum replacement tools, you have to do things the manual way. The essence of this technique is using the *Tab to Transient* features of Pro Tools and then pasting either an audio region (sample) or a MIDI note at each transient. The key to doing this quickly and efficiently is to have the drum you are trying to replace recorded (and a fairly clean, minimal-spill recording at that) on its own track. It doesn't mean that it isn't possible to do it on a more complex track, but you certainly need to be a lot more selective with what you do, and the reasons for this will become apparent shortly.

The first step is to create a new, empty track (we will start by looking at the process for using an audio track and then discuss any differences and additional steps for a MIDI track afterward), which we will use for our replacement drum sound. For this process it makes things a lot easier if this new track is located directly below your source track for ease of navigation. Once this is done, you need to make sure all the transient markers are in the right place for the source track if you haven't done so already (see end of chapter 9 for details), because we will be relying on the accuracy of these transient markers to achieve what we want to. And finally you need to decide on the sound you wish to use to replace the drum. You should import it into the project, place a single copy on the destination track (at the very beginning is helpful), copy it, and then mute it.

You need to activate *Tab to Transient* mode (Cmd[Mac]/Ctrl[PC] + Alt + Tab or the *Tab to Transient* button, located below the edit tool buttons at the top of the Edit window), so that you can use this feature, and you need to select a region in the source track that you want to carry out the drum replacement on.

The final thing you need to do in preparation is turn on *Keyboard Commands Focus Mode*. This is a special mode in which many of the normal modifier keys (such as Cmd[Mac]/Ctrl[PC], Alt, Ctrl[Mac]/Start[PC], etc.) are not required, and the main QWERTY keys become a very useful group of "one touch" commands. When you have to do very repetitive tasks such as drum replacement, the existence of single-key commands that achieve what we want them to do, without having to use modifier keys, right-clicks, or menus, makes life so much easier, and the benefit to you simply cannot be overstated. There are a few different places in which we turn on Keyboard Commands Focus Mode, and one of those, the one we are interested in, is for the main edit window/arrangement. To activate this mode, you need to look for a small square icon with an "a" in one corner and a "z" in the other, which is located at the top of the vertical scroll bar on the right of the window. When this mode is activated, the icon will have a yellow/orange highlight to it, and then you are ready to go.

FIGURE 11.5
There is no dedicated tool for drum replacement in Pro Tools; nonetheless, it can be achieved pretty easily using the Tab to Transient features and a bit of copying and pasting. Unfortunately this method doesn't follow the dynamics of the source audio, so it might be necessary to use the Clip Gain control on individual drum hits to replicate the dynamics of the source, as seen in the example above.



The four keys that you are going to need for this process are Tab to move forward through the transients (and Alt + Tab if you need to go to the previous transient if you missed one), V to paste the copied replacement sound on the new track, ; (semicolon) to move down from the source track to the replacement track, and P to move back up to the source track once you have pasted the replacement sound. From here the process is extremely simple and simply involves a very particular key sequence that you will master very easily, and you will probably be very surprised at how quickly you can progress through a track. That sequence is Tab > ; > V > P, which translates, in real terms, to

Move to next transient > Move down from source track to destination track > Paste previously copied replacement sound > Move back up to source track.

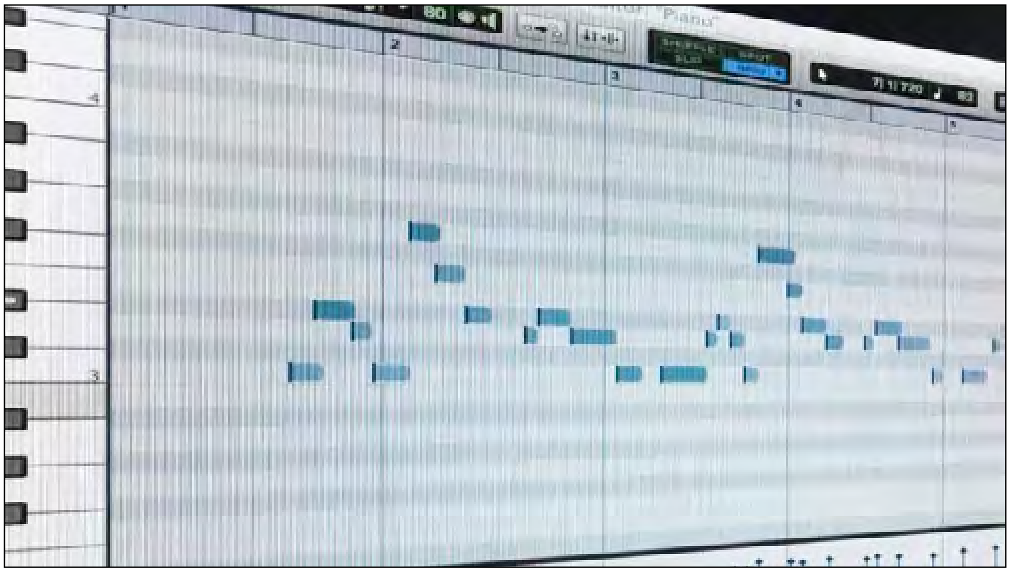
When you reach the end of a region, you may find that two presses of Tab are required to get to the next transient, as a region start is also considered a "transient" when using Tab to Transient, so you can't fully go into automatic mode yourself and will need to pay a little attention, but on the whole it is a very easy process.

As stated at the beginning of this chapter, having a recording with lots of spill or, indeed, a fully mixed stereo bounce of a drum take doesn't mean that the process is impossible, but it means more complications. As we have seen, the transient markers are the key, so, if you have a more difficult recording to work with, it will simply require you to do more work at the transient detection stage and try to make the transient markers only occur (by sensitivity adjustment or by manually placing or removing markers) when the drum you wish to replace is sounding. If you had a very complex recording, there are things you can do to help. You could, for example, make a copy of the recording and then use some plug-in processing on it (EQ, compression, expansion, and/or gating) to try to focus in on the particular sound you want to replace. You could then consolidate these regions and use transient detection on this new, bounced file and perhaps have a better chance of being able to get the transient markers in place with a simple sensitivity adjustment. There are usually ways to achieve what you want to, but some methods and situations are easier than others, so you might just need a little patience.

The biggest problem with the method described above is that each drum hit will now be *exactly* the same: the same sound, the same tone, the same level— everything the same. Unfortunately this is inherent in the process of repeatedly pasting the same file. If you want a little more flexibility, then creating an empty MIDI track (or Instrument track, if you plan on using a plug-in sampler to generate the replacement sound) might be the solution. Once the track is created, use the Pencil tool to create a note on the new track. The actual note (C, D, E, etc.) isn't important, because we can move this later, and in all honesty the note length isn't especially important either, but it makes sense to keep the note length fairly short, so that you can see at a glance each individual note rather than having longer, overlapping notes that are harder to make out. Once that note is created, simply cut it, and then repeat the process described above for samples. Instead of moving to the next transient and pasting an audio region (sample), you are simply pasting a MIDI note that will be used to trigger a sample when you are done.

There are two advantages to using the MIDI method. The first is that, because you are pasting a MIDI note, you can, if you wish, make adjustments to the note velocity on a note-by-note basis once the first stage of replacement is done. This would mean that, assuming you had a snare drum sample with multiple layers and velocity sensitivity, you could program variations in the velocity to try to mirror the dynamics (and often tonal change) of the source track. This is much easier to do if you have a fairly clean track to work from, because you will be able to visually identify louder and quieter hits and adjust note velocities to match. The other advantage is that it allows you to very quickly switch to a different replacement sound simply by loading a new sample, so you don't really have to worry about which sound you want to use before you start.

FOR MORE INFORMATION ON THIS TOPIC, PLEASE CHECK OUT THE FOLLOWING CREATIVELIVE VIDEO COURSE:



Check out Digital Drums with EZDrummer HERE.

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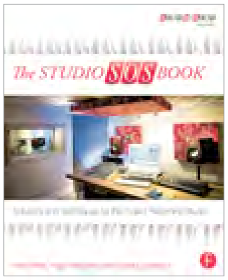


The following is excerpted from *The Studio SOS Book* by Paul White, Hugh Robjohns and Dave Lockwood.

Our *Studio SOS* visits don't just involve improving the acoustic environment of the studio, as we are often also asked to diagnose problems with on-going mix projects. Our popular Mix Rescue column goes into forensic detail on a specific track each month, but there are some basic principles that are worth observing when mixing that we can pass on in this chapter, along with many practical tips and techniques.

Recording and mixing can be difficult for musicians without much studio experience, as the way a song is arranged for recording can be very different from the way it might be treated for a live performance. Commercial recording projects are most often set up with a producer directing the engineer, but in the project-studio world most of us have to double as engineer and producer, and quite often as the performer too. To get a good finished product we need to know a bit about arranging and producing, as well as engineering and performing.

Mixing is a skill that requires experience to develop, so if your first attempts are disappointing, you shouldn't start to think that maybe you can't do it – regard them as learning experiences rather than failures. You can learn a great deal about arranging, recording and mixing simply by listening more carefully to your record collection – all the great secrets of music production are there for the taking! As soon as you start to listen analytically, as opposed to just enjoying the music, you'll start to be able to pick out the different elements and be able to see how they're balanced and arranged, what effects are being used, and what the tonal qualities of the individual parts are. You can then apply these ideas to your own mixes.

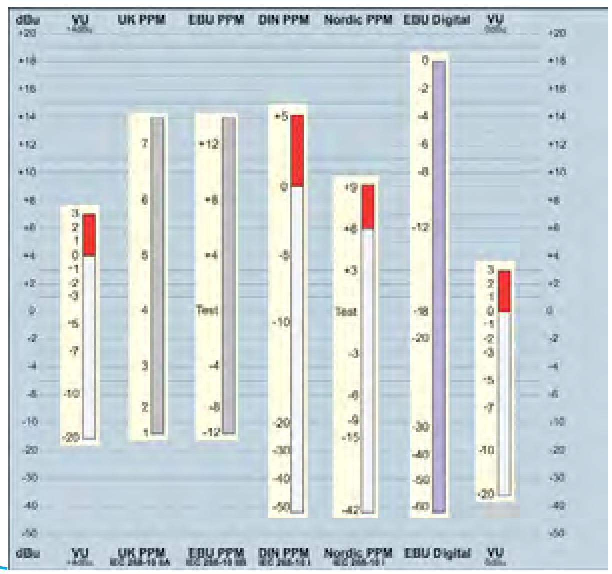
>

Mixing a multitrack recording requires a unique blend of artistic and technical skills – it's no surprise that it takes most people some time before they start feeling confident about their abilities.

The key to a good mix is a bit like the key to good cooking – get good ingredients, use the appropriate amounts, and process them correctly. And, as with cooking, too many flavours can confuse and compromise the end product, so question each element of your musical arrangement to ensure that it is there for a specific purpose. The mixing process will be much less challenging if you have managed to keep spill between instruments to a minimum – if you record one instrument at a time, of course, that won't be a problem – but sometimes spill helps to gel everything together nicely too, so it shouldn't be seen as necessarily a bad thing! It also helps to have left plenty of headroom while recording each source so that nothing clips if somebody sang or played louder than they did during the sound-check. Any hum or interference problems really should have been dealt with at source, before recording, rather than hoping you can sort it out later. Any instrument rattles, buzzes, humming or distortion (other than by intent, of course) that gets onto your original tracks will be very difficult, and sometimes impossible, to remove during the mix.

TIP

Those brought up on analogue gear often get into the habit of pushing the recording-level meters into the red, as analogue equipment has a big safety margin (headroom) built in above the point where the meters start saying 'enough'. Digital gear has no built-in headroom, so you have to create your own by leaving space between your loudest recorded peaks and the top of the meter, indicating 'digital full scale', usually abbreviated to 0dBFS.



GETTING STARTED

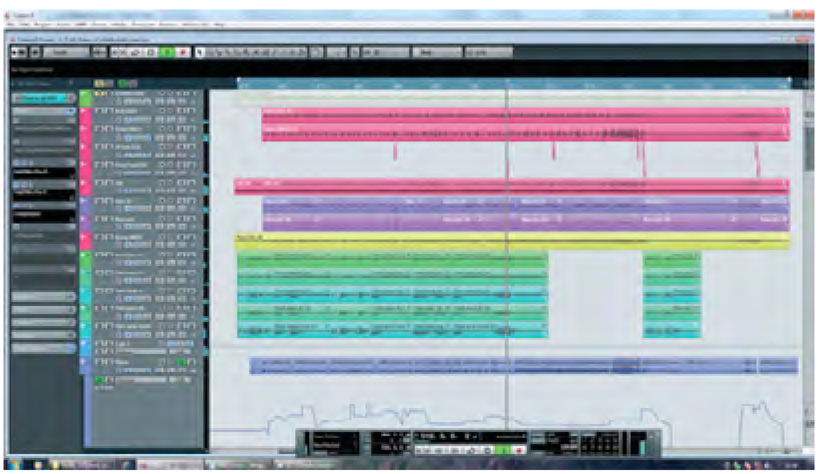
In the era of tape recording with analogue mixers and hardware outboard the detailed mixing work would only really start when all recording was complete. However, Digital Audio Workstation (DAW) software saves every parameter of your session, and whenever you open the file again it's back exactly as you left it. Some people therefore like to build up the mix as they record, making incremental tweaks throughout the process until the entire track is finished. With this approach there is no actual 'mixing stage' as such, just a continuous process of creation and refinement – by the time

TIP

Try to develop a consistent system for labelling projects and versions and use a logical folder system for archiving them. We suggest including the date as part of a file name as the date assigned by the computer may change if you change or update the file.

you record the last part and apply any processing it requires, the mix is effectively finished. The way a part sounds in a track always depends on what else is playing at the same time, but because each part is heard fully in context as it is added under this scenario, there is perhaps less of a tendency to create parts that don't fit or that have to be radically processed or re-recorded later.

Again, back in the analogue world the studio's 'tape op' would make up a track sheet detailing which instrument was recorded on which track of the tape, along with start and end times. Computer-based DAWs allow you to work without paper (although a notebook is still always valuable), but that doesn't mean you don't need to spend some time labelling things. In addition to naming individual source tracks, you may also wish to keep additional notes using the notepad facility of your computer, and maybe even use a digital camera to capture the control settings on any outboard equipment, such as analogue mixers or processors. If you save these files in the same directory or folder as the audio and project files you'll always know where to find them. In some DAWs you can also assign graphic icons and colours to the tracks to aid navigation around the arrangement page – for example, colour all the vocal tracks yellow, drums brown, all guitars red, keyboards blue, and so on.



>
Working with analogue-tape multitracks, it was always essential to keep an accurate log of where everything was recorded using a track sheet (opposite). On a DAW, you can see which tracks have been used and are far less likely to make a costly error, but it is still worth naming tracks, channels and buses as you go along, especially if you never want to have to try rebuilding a corrupted session where all the files are labelled 'Untitled Audio'…

TIP

Don't rely on backing-up files to a different partition on the same hard drive as you can still lose everything if that drive fails – back-ups must be saved to a physically separate drive. In addition to backing-up to hard-drives, also consider using recordable DVDs, local network storage systems (NAS), or online 'cloud' storage. Apple's Time Machine or one of the Windows automatic backup systems can also save the day when things go badly wrong.

PRELIMINARY HOUSEKEEPING

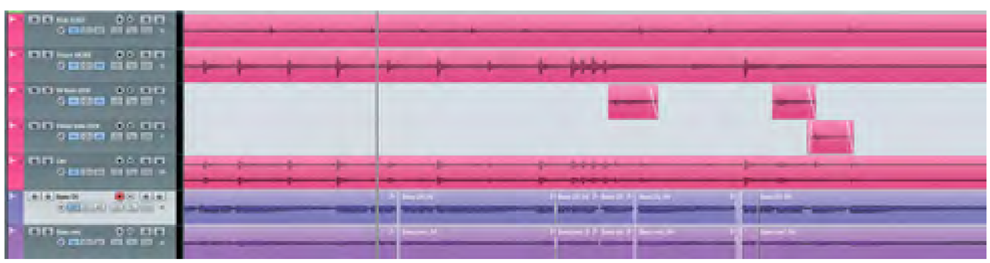
Before starting the creative part of the mix – what some people might call 'right-brain' activity – it helps to get as much of the more technical 'left-brain' activity out of the way as possible. A touch of paranoia doesn't go amiss at this stage, either! As firm believers that digital information doesn't really exist unless copies are stored in at least three physically separate locations, we urge you to back-up any important projects before you start the mix: hard drives can and do fail, usually when it will cause the most grief! This double back-up approach also enables you to go back to the original files if any destructive waveform edits you perform go wrong. Whether your system is computer-based or a hardware digital recorder you should back-up the files as soon as you can, and update the back-ups regularly as you work through the project. Most computer systems have the ability to allow you to back-up your project using just the specific audio files actually used in the current arrangement, and it makes sense to use this approach to save drive space – there's little point backing up discarded takes unless you think you might change your mind.

> Back up your session data, and then back it up again! Hard drives--even good ones like these--can, and do, fail without warning.

If you're working on a collaborative project, ensure where possible that you have the uncompressed WAV or AIFF files to work with. MP3s are fine for sending test mixes back and forth, but you should avoid using MP3 audio files as part of a final mix as their quality has already been compromised by the MP3 data reduction process. This may not sound obvious at the time, but your finished mix will inevitably be converted to an MP3 at some point if you want to make it available as a download, and a double dose of MP3 data reduction can have a very noticeable effect on audio quality.

TRACK 'CLEANING'

Before starting the mixing process it's worthwhile going through the individual tracks one at a time, checking for unwanted noises or unintentional audio picked up before the playing starts or after it finishes. Unwanted sounds can be permanently silenced in the waveform editor or turned down using either fader or mute automation. Most of the more sophisticated hardware recorder/workstations include fader and/or mute automation so you can generally do this quite easily. It doesn't matter how you decide to tidy up any unwanted noises as long as the job gets done, although a non-destructive method, such as fader automation, affords the luxury of undoing any mistakes. It is also common practice to mute audio during sections of the track where the instrument or voice isn't playing: for example, muting the vocal track during instrumental bridges or solos, or cleaning up spaces in electric guitar tracks, as these are often quite noisy. With live drums we have already recommended (in Chapter Ten) that you mute the tom tracks between hits, and again it is often more reliable to do this manually using fader automation or destructive editing rather than using gates. It may also be beneficial to gate the kick drum to reduce spill and to cut off any excessive ringing – but we are choosing a gate in this instance as a more pragmatic solution since the kick drum is generally playing throughout the track, whereas the toms are employed only sporadically.



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Cleaning up your tracks before you start trying to mix them will allow you to concentrate on the creative process, rather than constantly being distracted by the need for remedial action.

You might also want to check and adjust the audio levels of each individual track to make sure nothing was recorded at too low or too high a level. It is usually easiest to mix a song where the individual tracks peak between –18dBFS and –10dBFS as this leaves adequate headroom for the level to rise as the tracks are combined into your final stereo mix, without any significant system noise (assuming your interface is working with 24-bit conversion).

We see many cases where as soon as you start to build the rhythm-section mix it peaks close to (or even above) the maximum full-scale level, which means the output of the mixer will inevitably clip once other tracks are added (if it isn't clipping already). In theory, the floating-point arithmetic used in typical DAW mixers makes them almost impossible to overload internally, but the output D–A converter will still clip unless you pull down the output fader to reduce the peaks levels at the output below 0dBFS. However, if you have any plug-ins on your master bus, they may still be clipping, even if the output isn't. All in all, it just makes far more sense to work in the same way as we used to in the analogue domain, with sensible headroom margins. That means moderate source track levels and mix levels well short of digital maximum – and there is no compromise in removing any redundant headroom afterwards in mastering.

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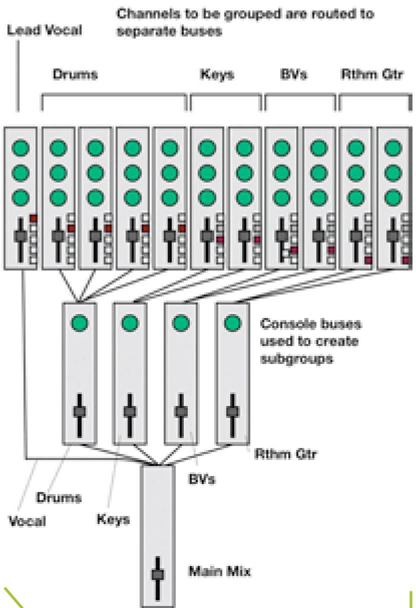
If you are recording your source material at 24-bit resolution, there is no advantage to be gained by keeping levels high, in fact you may well be clipping a plug-in somewhere without realising. It makes far more sense to turn up your monitoring, work with plenty of headroom in your software mixer, and make up the final level in the mastering.

INITIAL BALANCE

A question we are often asked is how to start balancing a mix. The answer seems to depend on who you talk to! There seem to be two main approaches to getting an initial balance, and the leading engineers and producers we interview seem to be pretty evenly split over which method is best. One method is to start with the rhythm section and then build up the mix one instrument at a time, while the other is to push up all the faders to the unity position and then balance everything in context. Others balance the rhythm section and the vocals (as the two most important elements of the mix) before bringing in the other instruments. In our opinion, for those with less experience, the 'one section at a time' option (starting with drums and bass) might be safest.



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Routing logical sub-sections of your mix to buses can make a complex mix containing a large number of source tracks much easier to manage, reducing your final balancing process to just a small number of faders.

BUSES AND SENDS

Having worked on a considerable number of Studio SOS and Mix Rescue projects where DAW bussing either wasn't used at all, or wasn't used effectively, we thought it might be useful to go over the basics of this relatively straightforward but nevertheless important subject. Routing every source track in your mix directly to the stereo mix-bus makes complex mixes harder to manage, especially if your project contains a large number of source tracks. Effective grouping allows you to control and balance your mix more efficiently using fewer faders.

A related topic, but equally important in mixing, is that of effects buses which are used to allow a single effect plug-in to be shared amongst multiple source tracks. Most DAWs handle bussing in a similar way to their hardware mixer counterparts, so perhaps the place to start is to explain the term 'bus'. In 'mixer speak', a bus is the place to which a number of separate audio signals can be sent to be mixed together – hence the term 'mix-bus'. In a hardware mixer, a mix-bus usually takes the physical form of a length of electrical conductor running the full width of the mixer, with each channel contributing some of its signal to the bus, as required. Just think of it as being like a river (the mix-bus) with streams feeding into it (the mixer channels or DAW tracks), terminating in an estuary where the river flows into the sea, which is analogous to the output that carries the mixed signal.

In a very simple mono mixer there may only be one mixbus – the single main output mix – and the channel fader determines the channel's contribution to the mix bus. In a basic stereo mixer there will be at least two buses, one for the left output mix and one for the right output mix, with the channel fader setting the overall contribution level and the channel pan-pot deciding how much of the signal goes left and how much goes right.

A larger mixer may have more buses than simply the left and right stereo mix. In the hardware world these additional buses are often called 'sub-groups' and usually feed separate physical outputs so that you can feed different signals or combinations of signals to a recording device or other destination. A popular mixer format might, for example, be 24:8:2, which means the mixer has 24 input channels, eight sub-group buses, and two main outputs (the stereo mix). The individual channels can usually be routed to any desired combination of the eight groups and stereo main outputs, and the groups themselves can also be routed to the stereo mix bus. For example, if you have ten separate channels of drums you can route them all to a sub-group bus (usually a stereo group bus), and then route the output from that mix-bus into the main stereo mix. The drum signals still get to the stereo mix, but as they now go via a sub-group bus the overall level of the drum mix can be controlled using just the bus fader. The advantage of this approach is that once you've balanced the various drum tracks contributing to the sub-group bus, you can then adjust the level of the entire drum kit within the overall song mix using just one fader – and that's a whole lot easier than having to adjust ten faders while trying to keep their relative levels intact whenever you want to turn the drum kit level up or down!

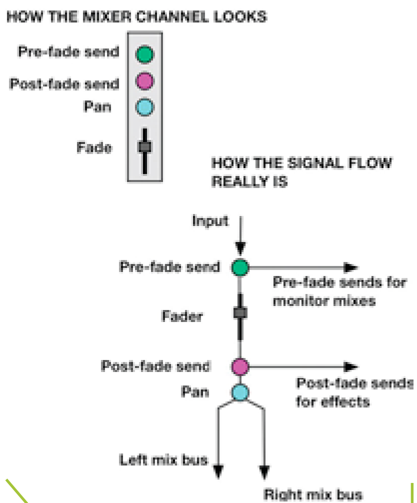
In the software DAW world it is common to have more mixbuses available than we'll ever need, and DAW buses can usually be configured for mono, stereo and even multi-channel applications. DAW mix-bus outputs can usually be routed directly to a physical interface output, or into other mix buses including the main stereo mix. On a hardware mixer this would be done using physical routing buttons, but in a DAW it's usually done via a pull-down menu of possible destinations, and you can think of this almost as setting up a mixer within a mixer.

Using sub-group mix-buses you can set up any number of logical collections of source tracks to allow them to be balanced against other grouped instruments using their respective bus faders. For example, you might put all your drums in one group, all the pad or supporting keyboard parts in another, and all the backing vocals in yet another. Similarly, multiple supporting guitar parts can be sent to their own bus. Main vocals and solo instruments are not normally sent via group buses as these usually require independent control so they can go directly to the main stereo mix bus.

A further advantage of creating sub-groups using bussing is that most DAWs allow you to insert plug-ins into busses, just as you can into channels or into the main mix output. That allows everything sent to one group to be processed together, for example, adding a little overall compression to a multi-layered backing vocal mix to help glue it together, or perhaps some overall EQ to alter the general tonality of the sub-mixed parts.

AUXILIARY SENDS

Aux sends provide another way of sending signals from a channel to more dedicated mix-buses, and most mixers and DAWs offer the choice of whether the 'send' is derived 'prefade' or 'post-fade'. As the names suggest, a pre-fade send picks up its signal before the main channel fader, so its level won't change as you adjust the channel fader. Conversely, a post-fade send is sourced after the channel fader, so as you pull down the channel fader the amount of send signal reduces correspondingly. In a hardware mixer the aux sends usually have a master output level control and appear as a physical output on the connection panel. DAWs are typically more flexible, and these aux or send busses can usually be routed anywhere you like! As



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Pre-fade sends take their signal from a point before the channel fader and are best used for application such as monitor feeds – the main mix can then be altered without risk of upsetting the performer(s). Post-fade sends are generally used for effects: with the feed to the effect being taken after the fader, the effect/dry ratio will remain the same, wherever the fader is positioned.

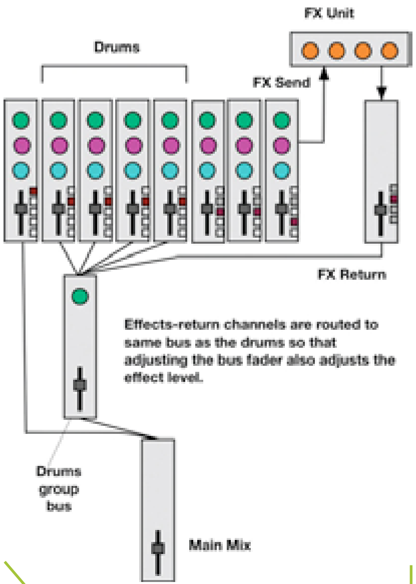
with the sub-group buses there is usually a main aux bus level fader so that the overall bus signal level can be changed as necessary.

So, why 'pre-fade' and 'post-fade'? Pre-fade sends, being independent of the channel fader position, are ideal for setting up monitor mixer for the performers. The aux or send bus signal would, in this case, be routed to a physical output on the audio interface (assuming your interface has more than two outputs), and sent from there to a headphone amplifier for the performers to hear the backing tracks. The pre-fade aux send controls in each mixer or DAW channel allows a complete mix to be set up separately from the main stereo mix, allowing different sources to be emphasised or reduced as necessary to help the performer, without upsetting the actual recorded stereo mix. Whereas a small hardware mixer may have only one or two of each kind of send per channel, most DAWs are fairly generous in this area and the number of physical outputs on your audio interface is more likely to set a limit on the number of different monitor or 'foldback' mixes you can set up.

Post-fade sends are usually used to feed effects processors, like reverberation, so that if a source channel is faded out it no longer contributes to the reverb effect. In a hardware mixer the effects units are usually external processors, with the aux send connected to their input(s), and their outputs returned to the stereo mix bus via either dedicated 'effects return' channels, or other spare input channels. When using a DAW, the post-fade sends could be routed to physical outputs on the audio interface to feed external signal processors, but most people use software plug-ins to provide the effects instead. A typical scenario might be to use one of the postfade sends to feed signals from each channel to a bus into which a reverb plug-in (set to 100% wet, 0% dry) has been inserted, with the bus output (which carries just the reverb signal) being routed into the main stereo mix. The various mainstream DAWs handle this kind of application in slightly different ways, but few stray far from the hardware mixer paradigm. Once set up, this arrangement allows the amount of reverb added to each mixer channel to be controlled using each channel's post-fade send control, and the overall amount of reverb to be set with the bus fader. Obviously each channel will be treated with the same type of reverb – just as each real instrument placed in the same physical recording space will create the same-sounding room reverb – but the amount added to each is fully and independently controllable using the aux sends.

SENDS AND GROUPS

A slight complication arises if you set up post-fade sends from channels that are routed through a group bus. This is because the effect level won't change as you



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Any effects sent from the source channels routed to a bus will not be attenuated when the bus fade is pulled down, thus inadvertently changing the wet/dry balance – if you have enough computer horsepower, you can solve this potential problem by using a dedicated effect with its returns routed to the same bus.

adjust the group fader if the output of the effect process is routed directly to the main stereo bus. If you fade down the group, any send effects associated with contributing channels would remain active, and this isn't usually what you want! (Although, you can use this to create the odd special effect where the dry signal fades to zero leaving just a ghostly reverb.) Fortunately, though, this problem is easy to remedy because all you need to do is to route the output from the effect bus back into to the same group so that when you fade down the group, the effects go down with it. The only restriction is that you have to make sure that the effects processor or plug-in is only dealing with signals from channels contributing to the same group bus.

Where you need the same effect to work on other channels outside of that group you will have to copy the plug-in settings and insert another instance of it in a different effects bus wherever else you need it, but with the horsepower of a modern computer that's rarely a problem.

GROUPING THE VCA WAY

Another way to set up control over specific combinations of channels is to use fader grouping. This is available on some larger analogue and most digital hardware consoles, but is a function that almost all of the mainstream DAWs support. The concept is that instead of routing and physically combining multiple-channel audio signals via a separate mix-bus, the relevant channels are routed directly to the main mix bus, but their faders are linked together to make a 'virtual group' so that they all move together. This means that a number of related channels can be controlled as if they were a single entity, with a single fader, but their audio signals don't have to be combined in an audio sub-group.

Most DAWs work in a way that is roughly equivalent to analogue mixer 'VCA' (or digital mixer 'DCA') groups, where selected channel faders are linked so that they operate together. The process usually involves assigning the required channel faders to a particular fader group – once linked, the group channel faders will all move together whenever any one of them is adjusted, each changing by the same proportional amount to retain the correct relative balance between member tracks. Multiple fader groups can be set up, although most DAWs don't allow the same channel to be a member of more than one group – which is just as well as the result could be quite confusing! With VCA-style fader grouping any post-fade send effects will behave normally, changing in level along with the channel fader, as the 'virtual group' level is now being controlled by collectively changing the individual track faders directly.

THE ART OF MIXING

During the initial stages of mixing you shouldn't need EQ, panning or effects: you just want to get a feel for how all the parts work together, and to listen for conflicting sounds that may be masking other important elements sharing the same frequency range. The lower midrange between 150Hz and 500Hz is particularly prone to this kind of congestion. Once you've balanced the drums and brought in the bass guitar, you can proceed to bring up the vocals and other instruments, accepting that you might need to adjust the bass or kick sounds later, as they can sound very different when everything else is playing.

If the original parts are all recorded well your rough initial-level balance should sound pretty good as it is, although drums often tend to need a little EQ as a close-miked drum is not really a natural-sounding source at all! Kick drums often need beefing up a bit too, but may also need some low-mid cut to avoid boxiness. It's important that the arrangement leaves adequate space for the vocal, either by leaving temporal space between notes and phrases, or by leaving the vocal part of the frequency spectrum relatively uncluttered – or both. This is largely an arrangement issue, although there are some useful techniques we will explore that can help push the vocals to the front if that turns out to be a problem.

POLISHING AND SHAVING

Once you have a workable initial balance you can scrutinise the various parts to see if anything would benefit from adjustment. If supporting parts, such as keyboard pads or acoustic guitars, are clouding the lower midrange, this can be remedied by thinning them out using low-cut EQ set at a slope of 12 or 18dB/octave. You might, in some instances, be able take the filter cut-off frequency as high as 300 or 400Hz, and although this will leave the instrument sounding very thin when heard in isolation, other parts will be providing the necessary low end in the track and in context it will still sound fine – but you'll now have plenty of spectrum space to work with for the rest of the mix. If you want a part to really sit back in the mix so that it doesn't fight with the vocals and solo instruments you can also take off a little of the higher frequencies, starting around 10kHz and working down until you achieve the desired effect. You'll generally want to use a softer, 6dB/octave slope for this.

VOCAL LEVELLING

Vocals usually sit better in a pop or rock mix if they are compressed to some extent, as compressors not only help level out the differences between the loudest and softest sections, but often also add a musically attractive density to the sound that helps push it to the fore. However, if you try to 'level' a vocal using only compression, you may find that you have to apply far too much compression to the louder sections to keep them under control, and that excessive compression affects the tonal quality as well as emphasising the background noise and room tone. Indeed, too much compression can undo the thickening effect of appropriate compression and allow the sound to slide back into the mix, rather than standing proud in front of it.

So, where the vocal levels fluctuate excessively – as it often does with less experienced vocalists – you'll probably find that you get a better-sounding end result if you first use your track level automation to smooth out the worst variations so that the compressor doesn't have to work quite so hard. However, since fader automation affects the signal after it has passed through any plug-ins in the channel, and we want the level automation to smooth the worst level changes before the compressor, one solution is to send the vocal to a bus and insert the compressor there. Alternatively, you can 'bounce' a version of the fader-automation levelled vocal track, and use that one instead in the mix with the compressor now installed back in that (pre-smoothed) channel.

You may also need to automate the overall vocal level throughout the track to help it sit correctly in different parts of the song where the backing level changes – for example, it may need to be a few decibels louder in the choruses if the arrangement gets busier there, or towards the end of the song where the intensity usually builds up.

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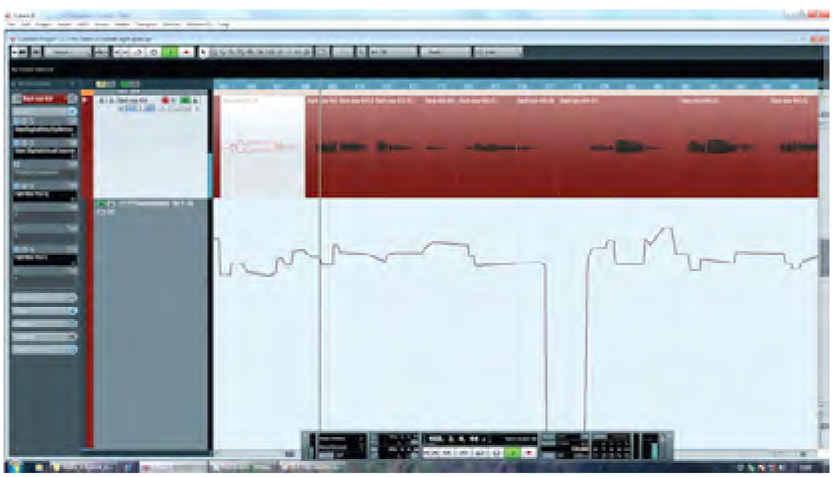
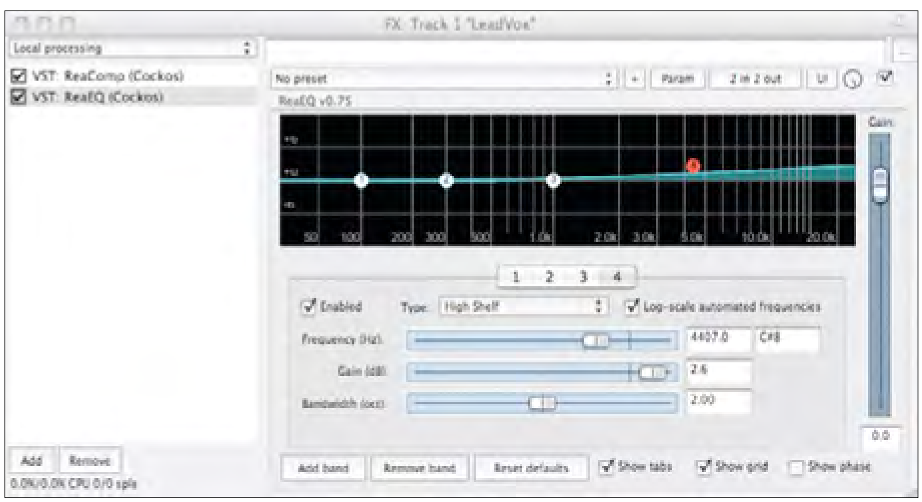
If you want your vocal levels to remain fairly constant, as in most contemporary productions, it is not a good idea to expect the compressor to do all the work – some pre-levelling with fader automation (remembering to insert the compressor after the fader) will always produce a better result.



When it comes to setting up a compressor for vocals, as a general rule you should adjust the threshold control initially to show about 6dB of gain-reduction, and then listen critically to see if the vocal sounds over-compressed. The action of a compressor depends on the attack and release parameters, the ratio and make-up gain and, most importantly of all, upon the relationship between the threshold control and the level and dynamics of the input signal. There's nothing wrong with using the presets supplied with plug-in compressors so long as you adjust the threshold (or input level) control to achieve the desired amount of gain reduction. If you don't, the plugin probably won't work in the intended way! This is such an important point to make that we've included a section to cover the use of plug-in presets.

Where EQ is needed to hone the tonal quality of a vocal, try to keep it subtle and avoid using narrow-bandwidth (high-Q) boosts, as these nearly always sound coloured and unnatural. A common, nasal kind of harshness can often be located around 1kHz, where a little narrow dip can often sweeten things up. A good rule of thumb is to attenuate elements of the sound that you don't like rather than boosting those you feel need to stand out more. If the overall sound is too soft and woolly apply some gentle, shelving low-cut, setting the frequency by ear. Sparkle and breathiness can be enhanced using a little HF shelving boost EQ above 8kHz or so.



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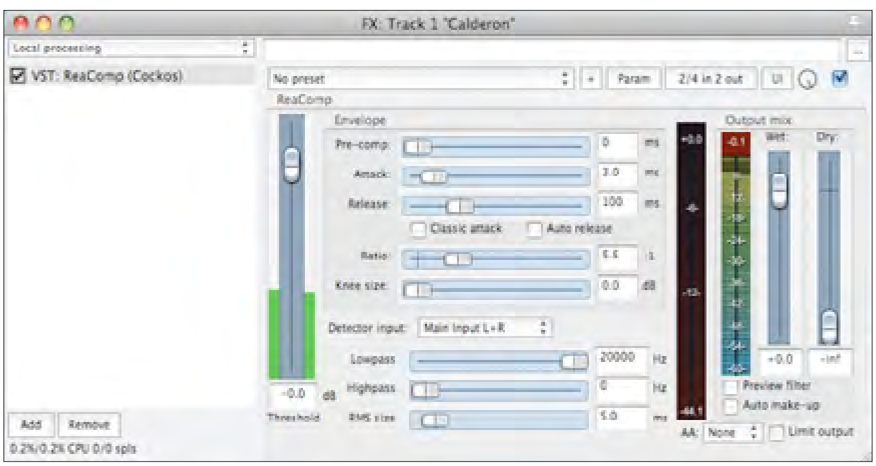
Lifting the whole of the top end with a broad 'air' EQ setting will bring out the detail in vocal, without making it sound harsh.

PLUG-IN PRESETS: GOOD OR BAD?

Presets for effects such as reverb, delay, modulation and pitch changers are often very helpful as you can usually find something that sounds good on a subjective level, and if you tweak the factory settings the results are immediately evident. Processing plug-ins, on the other hand, need to be handled with a little more care: as we've discovered on numerous Studio SOS visits, problems can and do arise when you start relying on presets for EQ, compression, gating and other 'processor' tasks. For example, EQ presets are created with no knowledge of what your original recorded track sounds like, so the parameter settings are based on assumptions that may be way off the mark. Furthermore, plug-ins relating to dynamic processing, such as compression or gating, have to make assumptions about the average and peak levels of the recorded track – and in many cases they're completely unsuitable without proper adjustment. For example, you may call up a vocal compression preset, but if you've left plenty of headroom while recording, as we advised earlier, then the signal level may never get high enough to reach the threshold setting included in that particular preset. We've come across exactly this scenario many times where a compressor has been inserted across a track and yet does absolutely nothing because its threshold is set way too high.



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The Threshold setting determines the level at which the compressor will start applying gain reduction –set it too high and the processor won't actually be doing anything.

The ratio, attack and release times within the preset are fine for the suggested task but you'll still need to adjust the threshold control to achieve the desired amount of gain reduction. For most tasks, something between 4 and 10dB of gain reduction will do the trick, and there's invariably a gain reduction meter of some kind to show you what affect the threshold control is having on the amount of gain reduction. Once you get the gain-reduction meter showing between 4 and 10dB on the loudest parts of the track, you should be able to fine tune for the optimum amount by ear.

Exactly the same is true of gates and expanders – you'll still need to adjust the threshold control to ensure the gate opens in the presence of signal and shuts down during pauses without chopping off any of the wanted sound. It wouldn't be so bad if this information was made obvious when you open a dynamics processor plug-in preset, but in most cases inexperienced users are given no indication that further adjustment is necessary.

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Watch out for software-instrument channel presets that also install a collection of processing plug-ins – not only will this eat up your resources unnecessarily, but also they are extremely unlikely to arrive with the right settings for your mix!

Also be wary of software-instrument channel presets that also install a collection of processing plug-ins, often ending with a reverb. Not only does this reinforce the impression that it is 'normal' to use lots of different reverb plugins within a mix, the EQ added is often designed to make the instrument stand out and sound impressive… and that's not always what's needed.

CASE STUDY ■ EQ PRESETS

We helped out a reader who wrote and performed some really good songs that were well played and well arranged, but he freely admitted to knowing a lot less about recording and mixing than playing and performing. Because of this he'd used a lot of plug-in presets, including complete instrument and vocal channel presets comprising multiple plug-ins, and had also used a large number of reverb plug-ins in separate channels rather than setting up just one or two different ones on post-fade aux sends.

We listened to some of his mixes and although they seemed pretty well balanced, they were also very harsh-sounding and consequently fatiguing to listen to. There was obviously far too much upper mid-boost – the area we often associate with presence in the 2 to 4kHz range – across the mix, and when we looked at his plug-in settings the cause was obvious: most of his EQ plugins used presets that boosted in exactly this area. Presets are often designed to make the tracks to which they were applied sound big, bright and up-front. In every mix some sounds really do need to be up front, whilst others need to sit behind them in order to create some front-to-back perspective, but the preset designers can't know how your song is arranged. So if you call up lots of presets, you may end up with everything trying to sound big and bright, pushing itself to the front of the mix, and resulting in a harsh flat mix as was the case here. Simply bypassing all the EQ plug-ins produced a marked improvement in the general sound of our case-study mix, and we could then show our reader how to apply the appropriate EQ to the appropriate sources for himself.

TIP

Even when you use plug-in presets just as a starting point, you still need to learn enough about how that particular plug-in works to be able to make further adjustments, according to the needs of the song. Dynamic plug-ins will almost always need you to adjust the threshold setting, while EQ is always best handled on a bespoke basis.

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EQ presets are often designed for maximum presence and punch, which is fine when that's what you need, but you can't allow everything in your mix to compete for the same sonic space.

PANNING

Stereo panning is an essential element of contemporary music mixing, as many sources are recorded in mono and then panned to suitable positions in the soundstage to create the impression of a stereo sound stage. In a stereo mix panning can also help to improve the aural separation between sounds, but is best not to rely on this as some people still consume a lot of their music in mono. Some mix engineers actually prefer to get their mixes sounding good in mono first (to make sure that any spectral congestion or masking is dealt with from the start), and only then think about the panning – although this usually means the mix levels need to be tweaked very slightly after panning to compensate for the inherent level changes that are imposed by the panning process (the extent being determined by the 'pan law' in use).

The soundstage of a typical pop mix usually approximates the way you might hear the musicians on stage, with the kick drum, snare, bass and lead vocal at the centre and other sources arranged towards the sides. Individual drums are usually panned to sound the correct way around from the audience's perspective (although some mix engineers prefer a 'drummer's perspective), and for most musical genres it is important not to pan things too much, making the drum kit unnaturally wide. Close-miked drums should always be panned to match their subjective positions portrayed in the stereo overhead mics to avoid generating confusing and conflicting stereo image information.

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The panned soundstage of a typical pop mix often approximates the way you might see the players line-up on stage, with the kick drum, snare, bass and lead vocal at the centre and guitars and keyboards arranged further out towards the sides. Toms and stereo overheads will be panned out of centre to give the kit some perspective, but this can sound very unnatural if it is overdone.

Backing vocals and other instruments can be panned wherever your artistic aspirations dictate and, as a rule, you should try to create a fairly even balance left to right – but it's probably wise not to pan any loud, low-frequency instruments too wide. Any instruments producing predominantly low frequencies, in particular the kick drum and bass guitars or synths, are normally panned to the centre so that their energy is shared equally between the two speakers. Stereo reverb effects returns are usually panned hard left and right to create the widest possible sense of space.

REVERB SPACE

Most reverb plug-ins expect a mono input signal but generate a stereo output. In a real acoustic environment the reverberation arrives at the listener from all directions from the side walls, floor and ceiling equally, regardless of where the original signal source is positioned inside the room. So artificial reverb should be the widest element in the mix, and all the other sound sources should be panned in a less extreme way to replicate being located within the acoustic space, in order to sound natural.

For instruments that have been recorded in stereo you need to judge that source's stereo width in the final mix by ear, as much depends on the way the source was miked in the first place. Different stereo mic techniques, and the relative distance between the stereo mic array and the source, will produce different source image widths, but don't worry if you need to turn the pan-pots inwards to reduce the apparent width (or offset them to one side), if that works better in the context of the complete mix.

TIP

Adding stereo reverb inevitably 'dilutes' the precise positioning of a sound, so where you want to establish a stronger sense of position, try adding a mono reverb to that sound and then pan the reverb to the same position as the dry sound. This is one occasion where dropping a mono reverb plug-in into a track insert point might be the best option.

PANNING TIPS

As a general rule, keep the bass instruments, including kick drums, to the centre of the mix so as to spread the load of these high energy sounds over both speakers.

Don't pan your drum kit too widely. At a typical gig, a physical drum kit probably occupies less than 25% of the width of the stage, so it probably shouldn't appear to occupy much more than this in your final mix.

Lead vocals are traditionally kept in the centre of the mix too, as they are the focus of the performance. You can be more adventurous with panning when you come to the backing vocals.

When you are deciding where to pan an instrument try to conjure up a mental picture of where that instrument should be on an imaginary stage, then close your eyes and adjust the pan control until the sound is where you want it. Go by your ears, not by the numbers around the pan dial.

Don't pan stereo mics or the outputs from stereo electronic instruments too widely or they will appear to take up most of the stage. In particular avoid having a stereo piano where all the bass notes are on one side of the mix and all the treble notes on the other.

As a rule, pan the outputs from your stereo reverb unit hard left and right and ensure the reverb decays away evenly, without tending to drift towards one side or the other. Mono reverbs can be used to help focus the position of a sound. Delay outputs can be panned wherever your artistic fancy dictates.

MIX PERSPECTIVE

If you rely only on panning for positioning sounds within a mix, the result can sound rather flat with all the sounds sitting along an imaginary line drawn between the two speakers. Achieving a sense of front-to-back perspective is also important in giving your mix interest and scale, but there are no dedicated front/back controls in a typical DAW mixer (unless you're doing a surround mix) and you have to create the illusion of depth using other techniques. We have already pointed out the dangers of trying to optimise each sound in isolation before mixing, and in particular the problems that result from trying to make every track sound as detailed and up-front as possible (especially if you rely on plug-in presets). Everything will be fighting for a place at the front of the mix, leading to a congested and often aggressive sound. The answer is to treat the individual sounds to make the music sound more three-dimensional, with key elements at the front and supporting instruments placed further back.

A fundamental rule of acoustics is that the intensity of the reverberation is essentially similar throughout the room, whereas the level of the direct sound from a source diminishes as you move away from it (according to the 'inverse square law'). That's why sounds heard from further away within a large space seem to be more reverberant. So in a real space the closer sound sources tend to be heard with less reverb and with a greater emphasis on strong early reflections from nearby hard surfaces, while sounds that are further away tend to comprise a larger percentage of reverberant sound, with the emphasis on the diffuse reverb tail.

Also, closer sound sources tend to be brighter while more distant sounds are heard with less high-frequency energy due to air absorption, and you can exploit these effects to enhance the perspective – for example, sounds that you want to appear at the front of the mix can be kept drier and brighter than those set further back.

You can go some way towards mimicking these natural characteristics by setting up a couple of aux sends feeding two different reverb settings, one of which is bright and weighted in favour of the early reflections to create a 'forward' sound, and a second that is warmer and more diffuse to create the impression of distance. You can then add these in different proportions to individual tracks to help place them appropriately in the front–back axis. Using an ambience reverb based mainly on strong early reflections helps to reinforce the illusion of closeness, while still adding the necessary 'ear candy' to the vocal sound – and if you listen to the more intimate-sounding records you'll often find that reverb has been used quite sparingly. Conversely, if you want to create a 'stadium rock' effect, where the band is supposed to be a fair distance away from the listener, you can use greater amounts of reverb combined with longer reverb pre-delay times (to make the walls appear further away) to help create this illusion.

EQ TO SEPARATE

Even though the 'ideal' is to try to arrange and record all your source sounds so that the mix sounds almost finished just by balancing the channel faders, you can often bring about a further improvement in a mix by the careful use of EQ. As you add more tracks the mix becomes more crowded and may start to sound congested, especially where the inappropriate choice of sounds or a musical arrangement brings several key parts into conflict. Before reaching for the EQ you should listen critically to the arrangement to see what the various parts are contributing, and consider rearranging them (use different chord inversions or orchestrations, for example) or even removing some parts altogether if they are contributing nothing particularly useful to the overall piece. Other parts can be dropped in level to push them further into the background. If that approach doesn't resolve the problem, equalisation gives you the ability to boost or cut the level of some frequency bands so as to change the frequency spectrum of the sound, but you must be aware that an equaliser can only boost frequencies that are actually present in the original sound – EQ is simply frequency-specific gain.

If you need to brighten a sound that has no natural high end, you could try a harmonic exciter as these actually synthesize new high-frequency components based on what's happening in the midrange. Provided that you use them in moderation they can actually work very well, and we have found them particularly useful for adding 'bite' to dull-sounding snare drums. Similarly, sub-octave plug-ins can be used to add a lower octave to sounds lacking in deep bass, such as weak kick drums.

CASE STUDY

We attended one *Studio SOS* session where the musician was using a default sampler sine wave tone for the melody lead line part but was concerned because he wanted a more 'edgy' tonality but couldn't get it to sound any brighter by using EQ. If you take a pure sine wave tone and try to modify it using EQ you'll find that it changes in level but not in tonality – and the reason is that a sine wave has no harmonics or overtones. That means that there's nothing for an equaliser to adjust other than pure volume! EQ only makes sense on harmonically complex sounds where it can change the balance of the various different frequency components that make up that sound. The only practical way of changing the tonality of a sine-wave bass sound is to process it via a distortion plug-in to add new harmonics which will then respond to EQ.

EQ 'BRACKETING'

EQ can be particularly useful in its role as a mix 'decongestant' when used to narrow down the frequency range occupied by certain instruments. Some engineers call this process 'bracketing', as it uses EQ roll-off at both frequency extremes to 'enclose' the frequency range covered, like brackets around words, and this technique helps reduce the amount by which the spectrum of one sound source overlaps other sounds occupying the same part of the frequency spectrum. For example, many pad sounds are rich in lower midrange frequencies that conflict with the lower end of the male vocal range, while any bright highs may merge with the sound of the guitars or conflict with the vocals. You can often squeeze them into a narrower range, without affecting their role in the track, by using high- and low-pass filters (with 12 or 18dB/octave slopes), or shelving EQ cut at both the high and low ends. Although the EQ'd sound might then seem thin or dull in isolation, you will invariably find that it sits better in the mix. A further benefit of bracketing is that as the mix becomes less congested, you may then be able to further lower the levels of some supporting sounds without them getting lost.

An alternative technique is to use a parametric EQ to carve a 'dip' in the middle of the spectrum of some sounds to make room for other sounds at a similar frequency. One example of this is using a parametric EQ to place a dip in a bass guitar sound to help keep it separate from the kick drum. You have to find the optimum frequency by ear, but in the case of kick and bass, it's usually in the 100 to 250Hz range.

>
In a busy mix, it can help to narrow down the frequency spectrum of many instruments, thereby reducing their capacity to mask other sources.

It is always a good idea to first try to fix a spectrum congestion problem using EQ cut, reducing the level of what you don't want, rather than boosting the bit that you want to hear more, especially if you want to achieve a natural sound. The human hearing mechanism seems to take far less notice of EQ cut than it does of boost, especially when the latter is concentrated in a narrow range. Where you do need to use boost EQ, keeping a wide bandwidth (low Q) sounds more natural than boosting a narrow (high Q) region of the spectrum.

Where the sounds are not natural (such as synthesised sounds or electric guitars), more radical EQ solutions may sound perfectly fine, although sticking to the 'cut first' rule still often produces the best-sounding results. Since these sources have no natural reference, the only rule is that if it sounds right it is right!

VOCALS

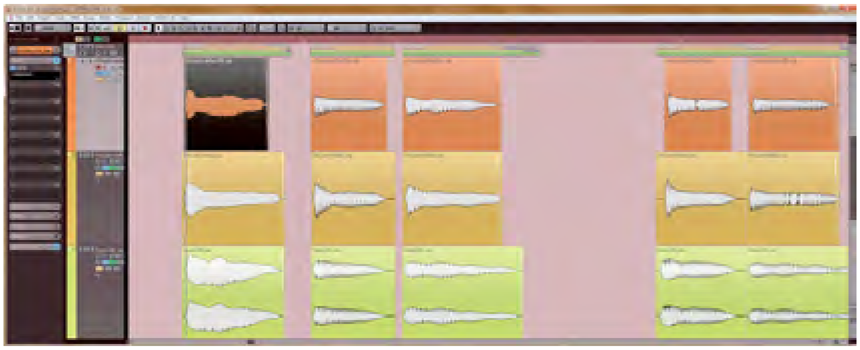
In a typical pop song the vocals are usually the most important element, so once you have controlled the level using either compression, automation, or both, you need to move on to the matters of EQ, reverb and possibly delay – the main tools in shaping a pop vocal. If you've chosen a suitable mic for your singer you should need very little EQ unless there is something problematic about the voice in the first place. However, here are some strategies that may help you focus your EQ efforts. Even if you've recorded your vocal using a good pop shield some air disturbances or mechanical vibrations may still reach the microphone, so placing a steep low-cut filter at the start of your plug-in processing chain can improve things by removing unwanted sub-sonic rubbish that you probably can't hear, but which will gobble up the headroom and make the speaker cones flap about to no useful effect! If you have a plug-in spectrum analyser it will often reveal unwanted activity in the region below 50Hz. A typical low-cut turnover frequency to employ with a vocal is 80 to 100Hz.

Cutting or boosting in the 2kHz to 5kHz range will allow you to fine-tune the amount of vocal 'presence', although adding significant boost here can cause harshness, too. An alternative strategy that avoids the risk of emphasising potentially aggressive presence frequencies is to apply a broad parametric boost at around 12kHz or an HF shelving boost above 8kHz, often known as 'Air EQ'. Boxiness or any tendency to sound nasal can be improved by applying cuts around 250Hz (boxy) and 1kHz (nasal). You should always try to avoid excessive EQ on a vocal though, as it can all too easily make the voice sound unnatural.

Compression can be applied before or after EQ, but the results will be slightly different depending on the processing order. You get more control if the compressor comes before the equaliser as there is no interaction between the two processes, but putting the EQ first makes the compressor respond more strongly to areas in which you've used EQ boost and less strongly in areas where you've applied EQ cut. Effectively, the compressor is trying to 'level' out any frequency-selective amplitude changes you've imposed with the EQ, although in some situations this is what gives the best subjective sound so it's usually worth trying both options and listening to the difference. However, we'd always advocate putting a low-cut filter before the compressor (or in its side-chain), otherwise there's a risk that it will react to breath blasts and subsonic rumbles rather than to the actual vocal level.

Backing vocals can also benefit from low-end thinning to stop them fighting for attention with the lead vocal and you may also need to add less 'air' EQ so that they sit back a little behind the lead vocal. Other than panning to create the desired stereo image there's no particular special treatment required for backing vocals, although where there are several layers of the same vocal part it can really help to tighten up the sound if you use your computer's editing facilities to line up the timing of some of the phrases (unless of course the original timing was spot on), especially where consonants (like 'S' and 'T') are involved. It often helps to make sure that any audible breaths occur at the same place, too.

> Backing vocals that use a lot of overdubbed parts can be difficult for the singer to keep completely tight. You can make a big difference to their impact in the mix with a little trimming and time adjustment of the major consonants.



It can sometimes be helpful to have only one backing vocal part pronounce all of the consonants at the start and end of words during recording. If the others are sung to deliberately de-emphasise these you avoid the untidy effect of three or four 'T's all turning up at slightly different times. You can also fake this effect using level automation to tail off the beginnings and ends of offending words in all but one of the backing-vocal tracks.

VOCAL REVERBS

It's often a good idea to set up a specific reverb just for your vocals. Convolution reverbs are brilliant for conjuring up the illusion of a real space, but you'll often find that a good synthetic reverb or plate emulation gives the most flattering vocal effect. Bright reverb sounds are popular on vocals but can also tend to emphasise any sibilance, so you may need to choose a warmer reverb if you detect any problems, or alternatively insert a de-esser plug-in into the reverb send. By de-essing just the reverb input signal, the process is much less obvious than if you de-ess the dry sound. It's also a good idea to roll off the low end, either from the send or the reverb return, to avoid adding more low-mid congestion and clutter to the mix. Many modern vocal effects use a mix of delay (usually with some high-frequency cut to make the repeats less obvious) combined with a suitable reverb, with pre-delay of between 60 and 120ms.

AUTOMATIC TUNING

Where the vocal performance is reasonably well pitched an automatic pitch corrector such as Autotune or one of its equivalents can add that final professional polish, and so long as you don't set the correction speed too high there will probably be no audible side-effects. The best results are usually be obtained by setting the correction scale to match the notes being sung, rather than using the default chromatic mode, where it will try to correct every note to the nearest semitone, regardless of whether or not it is in the correct scale. If the song contains a key change you can split the vocal onto different tracks and insert a different instance of the pitch-correction plug-in on each track set up with the correct scale notes for each section.

> Pitch correction processes like Autotune and Melodyne are fantastically powerful studio tools that can be used for both subtle 'invisible mending' and spectacular creative abuse!

TIP

A very convincing double-tracking effect can be created by duplicating a vocal track, delaying it by around 50 to 80ms, and processing it with pitch correction.

If you 'pitch-correct' a number of double- or multi-tracked vocal parts, the result can sound somewhat 'phasy' as the pitch of each part is now virtually identical. You can avoid this by backing off the pitch-correction speed to different degrees on some of the tracks, or you can apply pitch correction to some layers but not to others. For really serious 'pitch surgery' an off-line pitch-editing solution will provide more precise results.

CASE STUDY

We visited one reader who was recording his daughter's singing over some well-arranged and recorded backing tracks. However, we both felt that the use of Autotune was too evident on the vocals and suggested reducing the correction speed. To our surprise he said he hadn't used any pitch correction – that was how his daughter naturally sang! We know now that this is by no means an isolated case – it turns out that many young people sing to emulate what they hear on record and that extends to imitating the processing artefacts of pitch correction!

BASS GUITAR

DI'ing a bass guitar may well produce a clean sound with lots of depth, but it can also tend to get lost when the rest of the faders come up. Purpose designed bass-guitar-recording preamps or plug-ins often give better results, as we discussed in Chapter Nine, because they add some of the colouration of a real amplifier and speaker cabinet, giving focus and character to the sound. It is possible to get a good DI'd bass sound by adding compression and EQ, but a dedicated recording preamp or plug-in is often the fastest way to get a sound that works if you don't want to mic a bass speaker cabinet. To some extent, it depends on the type of music you are producing as well, since a DI'd bass may work perfectly well in a sparse arrangement, whereas bass-amp modelling may help the bass to cut through in a busy rock mix.

DI'd bass nearly always benefits from compression to firm it up a bit – an attack time of 20ms or so will help emphasise the transient attack at the start of each note, and the release time can be set anywhere from 50ms to 250ms, depending on the pace of the bass part and on how obvious you want the compression to be. A ratio of about 4:1 and a threshold setting to give a gain reduction of 4 to 10 dB on peaks is a good starting point, but you'll always need to fine tune these parameters by ear, depending on how evenly the bass was played in the first place. On many of the mixes we've encountered in our Mix Rescue column, the bass part has suffered from the instrument not being played assertively enough. This tends to result in a sound that lacks punch no matter what processing you use, and fret rattle is also often a problem. Recording can only capture the original performance, so that performance has to be good if the end result is going to be up to standard. It might be better to use a sampled bass rather than a poorly played real bass.

The punch of a bass guitar sound within a mix comes as a combination of its low end and its mid-range. There's very little signal content above about 4kHz other than finger noise, so to increase or decrease the amount of bass, you need to cut or boost between 70 and 120Hz – but remember that boosting the bass end too much will reduce the amount of headroom you have, forcing you to turn the overall bass level down. In fact, a lot of the apparent punch and tonal character of the bass guitar comes in the 200 to 300Hz harmonics range, and the best way to prove this is to listen to your mix on small speakers which have a weak response below 100Hz or so. If the bass seems to vanish from your mix you probably have too much deep bass and not enough going on in the 250Hz harmonics region. Use a low-cut filter to reduce anything below 30Hz or so, as any energy down there will probably be unwanted very low-frequency 'noise' caused by moving the strings slightly above the pickups, just by touching them.

> A clean, DI'd bass guitar may sometimes not have enough harmonic complexity to sit comfortably in a mix. A little parallel distortion, with its own EQ, can give it a more interesting midrange without it ever sounding noticeably 'dirty'.

CASE STUDY

We were asked to improve a mix in which the bass guitar seemed to be getting a little lost. Rather than put it through a bass guitar amp model, we first tried using a simple overdrive plug-in to add just a hint of thickness to the sound, and then applied some EQ boost at around 200Hz to give it more definition in the lower mid-range. The result was surprisingly effective, which shows you don't always need fancy or expensive plug-ins to get the job done.

ELECTRIC GUITARS

A typical electric guitar part covers mainly the 150Hz to 3kHz region of the spectrum – unless you use a clean DI, in which case it covers a lot more of the audible frequency spectrum. A sharp, low-pass filter cut at 18dB/octave or more can be useful to smooth out a gritty top end without making the instrument sound dull, whilst boosting between 1kHz and 3kHz brings out the natural bite. Be careful not to overdo the latter – things can get very harsh very quickly!

Where there are two electric guitar parts you can use EQ to try to differentiate them, although this is best done at the recording stage with different pickup, chord shape, and amplifier settings. Panning them apart will also help, but a good mix sounds nicely organised in mono as well as in stereo. During our Mix Rescue series we've come across a number of tracks where the electric guitar has been so distorted that it has absolutely no dynamics at all. Worse still, if the parts are riffs or chords, played solidly with no breaks, they tend to trample all over everything else producing mid-range sounds, making it very difficult to get a satisfactory mix. You can use EQ to bracket them, which helps, but the real solution is to record using less distortion, and to organise the song arrangement so that there are some spaces and variety. If you need to create a sense of power with more sustain consider using less distortion combined with a compressor.

For ambience, a simulated spring or plate reverb often works best, although if you don't want an obvious reverb effect, try using a hint of convolution reverb with a room, chamber or studio setting. This eliminates the dryness you get in closemiked guitar sounds (real or simulated), and adds a sense of room space but without washing out the sound with obvious reverberation. For 'stadium-style' rock solos, a delay of around 700ms either mixed in with the reverb or used instead of reverb works well.

Modulation effects such as chorus, phasing and flanging can be applied using conventional pedals during the recorded performance, or added afterwards using plug-ins. While plugins can usually achieve the right sound guitar players often need to be able to hear an effect whilst playing in order to perform most effectively. If you feel the need to preserve the flexibility to change the sound later, the simplest option is to take a second feed from the guitar as a clean DI and record this to a spare track just in case the effected version isn't quite right. You then have the option to process this clean track in software using an amp-modelling plug-in or re-amp it, to replace or augment the originally recorded guitar track.

TIP

Old-fashioned CRT computer screens and laptop computers radiate a lot of electromagnetic interference that can affect magnetic guitar pickups, especially single coil types when used with overdrive and compression. Flat panel displays do not radiate the same type of interference and are far more 'guitar-friendly'.

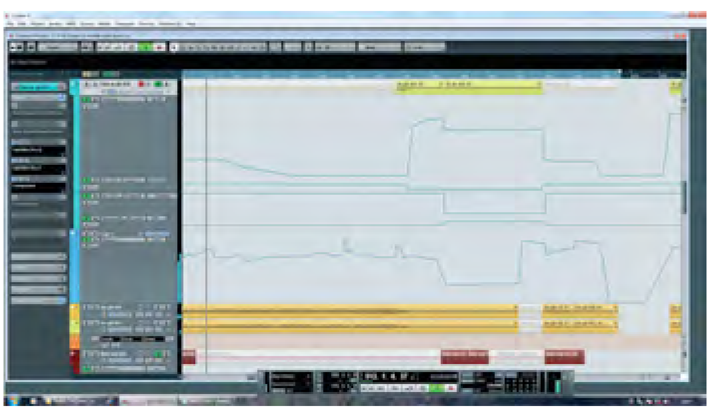
ACOUSTIC GUITARS

Where the guitar is part of an acoustic-band performance or playing a solo piece you'll probably want to achieve a natural-sounding tonal balance in the recording. A gentle top or bottom cut or boost may be all you need to fine-tune the sound in the mix. If there is any honkiness or boxiness in your recorded sound, however, you can locate it accurately by first setting up a fairly narrow-Q parametric EQ boost, and sweep that across the frequency range of the guitar until the offending part of the sound really jumps out at you. Once located you can apply cut at that frequency to reduce the undesirable element within the sound.

If the recording has been made in a fairly dead room a convolution-based ambience reverb can reintroduce a bit of natural-sounding life to the sound without an audible tail. In a pop mix the low end of acoustic guitars may conflict with other lower mid-range sounds, so it's a good idea to apply a low-cut filter or shelving EQ to thin out the bottom end. This keeps the body sound of the guitar away from the vocals and also stops it blurring into the low end of the electric guitars or the upper reaches of the bass. Listen to an Eagles album to hear how an acoustic guitar can really work well in an electric band context.

JUDGING THE BALANCE

Level adjustments are almost always necessary during the mix, as the song evolves and builds, even if it is only riding vocal level changes so the compressor has less work to do or lifting guitars slightly during solos, and so on. You can make these small fader movements manually on simpler hardware systems, but mix automation makes this very easy, of course, and on a DAW you can simplify the mix further by putting different sections onto separate tracks. As a rule, avoid changing the levels of the drums or bass guitar as these provide the backbone of the track against which



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Even the best-performed tracks can still benefit from small pushes and fader rides in the mixing, to highlight certain phrases or allow other material to come through in places.

TIP

Always check your mix on headphones, as these tend to show up noise and distortion problems, as well as any duff edits, more effectively than speakers. They also give you an idea of how the stereo panning will sound on an MP3 player with earphones – one of the main ways music is experienced these days.

level changes in the other parts take place. Changes in intensity in the rhythm section should ideally come from the playing, rather than the mixing.

When you have your basic mix sounding close to how you feel it should be, it is always worth taking a break to listen to a couple of commercial tracks in a similar style. Do bear in mind however that the commercial tracks will also have been mastered, so your mix may sound less tight and punchy at this stage – you can get some idea of how your track might sound when mastered by inserting a compressor and limiter temporarily in the main mix output. Use a low compression ratio of say 1.2:1 and then set the compressor threshold to give you around 4dB of gain reduction so that the track's entire dynamic range is squashed a little bit. Adjust the limiter so that it just catches the peaks giving 1 or 2dB of gain reduction, and if necessary adjust the output level control to match the level of your reference tracks. Remember to bypass these plug-ins when you resume mixing, as they are just for comparative listening.

>

If you are comparing your mix to reference tracks, try inserting a temporary mastering chain, consisting of a compressor and a limiter, to give your mix a similar density and loudness to the fully mastered material you are comparing to.

With that frame of reference fresh in your mind you can then make any adjustments to your mix that you feel necessary. Double-check the mix by walking around and listening to it without looking at the computer screen, and do this at several different listening levels, from quiet background music to fairly loud. Also listen to the mix from the next room, or the corridor with the adjoining door left open, as this seems to highlight any serious balance issues remarkably well. A lot of pros do this too, so it is a tip worth remembering. Finally, bounce a rough mix to an MP3 player or CD-R and check how it sounds on the car stereo, portable stereo system and earphones.

PROBLEM SOLVING

Recording problems can be technical, such as spill, noise and distortion; or musical, such as when two full-on, distorted guitars are both playing busy rhythm parts all the way through a song. And then there are timing and tuning errors, or incorrectly played notes... One of the most prevalent technical problems used to be noise or hiss, although this is far less of an issue now than it was back in the days of analogue tape. Today, hiss is more likely to be due to lack of attention to setting up a proper gain structure, although it can also be generated by older synths, guitar amplifiers, effects pedals and so on. The simplest tool in the fight against hiss is the noise gate, although this has its limitations.

NOISE GATE

A noise gate simply attenuates the signal during the pauses between sounds on the track to which it is applied. The amount of attenuation can range from a full mute to a more modest level reduction, depending on the gate's design and control settings. Gating can be very effective, especially where you have a lot of tracks in your mix, so long as the 'wanted' sound is significantly louder than the noise you need to remove. If, for example, a guitar solo appears only once in the middle of a song, you can either trim both ends of the section by editing (the preferred option) or pass the track through a noise gate. The gate is more useful if there are lots of breaks or pauses in the performance, as these will be muted automatically (but would take a long time to edit out manually).

>
You can use a noise gate to automatically clean up your recorded tracks prior to mixing, provided that there is a clear difference betweens spill and background noise and the level of the material you want to keep. If there isn't, you may be better off using one of the other methods outlined here.

When gating vocal tracks, it is better to set the gate's attenuation (sometimes called range or depth) to between 6 and 12dB, rather than allowing it to completely silence the track. This is because we still need to hear the breath noises, but at a reduced level, otherwise the performance can sound very unnatural. Ensure you set suitable hold and release times, so that the ends of the notes aren't cut off abruptly.

Constant broadband noise (hiss or other constant background sounds) can sometimes be dealt with using dedicated de-noising software plug-ins. Some of these work by taking a noise 'fingerprint' of the unwanted sound in isolation, and subtracting that from the wanted signal, whilst others involve a single-stage, multi-band filtering process. There will generally be a control that sets the amount by which the noise is reduced – trying to take it all out invariably introduces odd 'ringing' or 'chirping' artefacts, and better results are usually obtained by aiming for much more modest reduction to the noise of only a few decibels. If the noise is really bad, then doing two or three gentle passes usually sounds much better than one heavy-processing pass. Note that systems relying on a noise 'fingerprint' to calibrate themselves are only effective where the level and frequency spectrum of the noise is reasonably constant. Some of the more sophisticated de-noisers track the noise profile and adapt accordingly, although they can be fooled if the nature of the noise changes too abruptly.

TIP

If you need to use a noise gate in a track that also has a compressor patched in, put the noise gate before the compressor to ensure it triggers reliably. If you put the gate after the compressor there will be less of a differential between the levels of the loud sounds and the quiet ones, so the gate will tend to mis-trigger.

TIP

Reverb or delay can help hide any abrupt note endings caused by gating. If you try to gate after adding reverb or delay, you'll invariably truncate or shorten the decay tail of the effect in an unnatural way –although that is also the basis of the once-popular Phil Collins gated drums effect.

>
Dedicated digital de-noising processes can be remarkably effective, so long as you don't push them too far, causing side-effects that are more objectionable than the original noise. Several gentle de-noising applications are usually more effective and less noticeable than one heavy-handed application.

DISTORTION

While distortion is often used as an effect, we can sometimes face the problem of unintentional distortion. Distortion can either be fairly gentle and musically beneficial, as in analogue overdrive, or very hard and unpleasant as in the case of digital clipping. If not too severe, analogue (harmonic) distortion can be softened by using a very sharp high-cut filter to attenuate the unwanted upper harmonics, but where possible it is better to re-record the part to avoid having to solve the problem in the mix.

DAW users can automate filter plug-ins to apply cut only when the distortion is present, which avoids compromising the clean parts of the recording. Digital clipping is more problematic as it generates anharmonic distortion artefacts that crop up at frequencies both below and above the source frequencies, so they can't be tackled by high-cut filtering. Currently, other than very expensive specialised restoration software, there is no way to properly reconstruct distorted sounds, although it is likely that this technology will eventually percolate down to the project-studio price range where, it could be argued, it is most needed.

SIMPLY THE WRONG SOUND

Sometimes you may be presented with a technically good recording, but the sounds still don't seem to work in the context of the mix. Sometimes this is an arrangement issue that needs to be resolved through creative editing, or perhaps discarding or replacing some of the parts, but in other cases you can often use the EQ techniques already described to squeeze the sounds into shape. Where the problem can't be corrected in this way you may find that instruments such as electric guitar, electric bass or drawbar organ can be improved by re-amping. The one problem that EQ never seems to entirely resolve is the over-distorted rhythm guitar part.

CASE STUDY

One Mix Rescue song we worked on suffered through having a rock organ sound that sounded far too 'polite' and didn't cut through the mix. We added some fairly heavy upper-mid EQ boost and then fed it through an overdrive plug-in which we adjusted to dirty up the sound without making it too fizzy or raspy. This produced a perfectly usable result, but where processing fails re-amping may succeed. Re-amping simply involves routing the offending track to a physical output, then feeding it into a suitable amplifier and speaker that can be miked and re-recorded onto a new track. In this way, the amplifier and speakers modify the tone and allow additional distortion to be introduced where necessary. The same technique can be used to impart a more organic quality to guitar, bass guitar, bass synths, drawbar organs and other keyboard pad sounds.

Although you can feed a line output from the computer interface directly into a typical instrument amplifier (if you take care to reduce the line level signal down to instrument level), a dedicated re-amping box usually gives better results, as it deals with the signal level reduction in a convenient way, sorts out the balanced-to-unbalanced conversion, and provides a ground lift to avoid ground loops. Virtual re-amping is also possible if you use a DAW that has plug-in modelling guitar preamplifiers. By inserting one of these on a track, it is easy to reshape a sound in a similar way to passing it through a miked external guitar amplifier and, of course, it is far more convenient.

<
EQ presets are often designed for maximum presence and punch, which is fine when that's what you need, but you can't allow everything in your mix to compete for the same sonic space.

THE MIX MEDIUM

Many users record their stereo mix onto new tracks within the DAW, recording the results by 'bouncing' all the tracks to a new stereo audio file, after the mix has been set up and optimised. The use of (16-bit) DAT machines or CD-R recorders as mastering machines is far less prevalent now than it was in the past, as bouncing your mix within the computer is a lot more convenient and allows you to keep your final mix files at a high resolution. If you like to mix on an analogue mixer, either because you like the sound or the more tactile control it offers, the mixer's analogue output can still be recorded back to a new stereo audio track within the computer project. This means there's less requirement for dedicated stereo recorders for this application these days, although there are still plenty of portable stereo digital-audio recorders with 24-bit capability to choose from, if you have a need or preference for mixing onto a hardware platform.

Of course, some people still like to mix to analogue tape, simply for what it does to the sound, and then bounce that mix back into the digital domain as their master. If you are lucky enough to have a good quality two-track analogue tape machine that you can maintain and keep in good working order, then that's certainly a viable option. Alternatively, you could use one of the many tape simulation plug-ins on the market, some of which are excellent, and some rather less so. Beware of any tape-sim that causes audible distortion without being overdriven; that's really not the effect you are after, and not what analogue tape does to the sound at all!

\>
Where an external, hardware stereo recorder is still preferred, a stand-alone solid-state or 'card' recorder capable of 24-bit operation is probably the best solution. The recordings made on such devices can also be backed up to any computer.

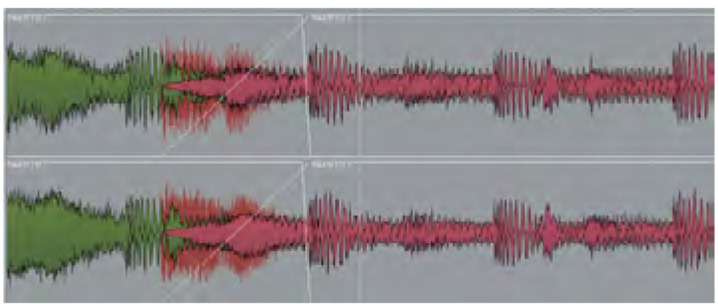
\>
Some of the latest tape-emulation plug-ins, such as this one from Slate Digital, make excellent sound polishing and enhancement tools, regardless of whether you think they are accurate or not – and many users will never have actually heard a high-quality analogue tape machine in action!

COMBINING MIX AUTOMATION WITH ADVANCED EDITING

Not everyone likes to mix with full automation in action across the whole mix, so 'final' mixes are sometimes created by editing together different sections from different mix passes. The desired edit points will usually be on transitions between sections, and it is worth trying to position them shortly before a drum beat so any small discontinuities are likely to be masked by the beat. Cross-fades should be made as short as possible – typically 20ms or so – while still providing a smooth, glitch-free edit, and they shouldn't overlap into drum beats.

Once you've edited a track, listen to it carefully on headphones to ensure that none of the edits are audible, as you can easily miss subtle editing problems when listening on speakers. Listen for and correct any level or tonal changes, as well as more obvious timing irregularities.

A common editing fault we hear is that although the timing of the edit may be correct and the edit made on a suitable beat, the decay of notes played before the edit are missing after the edit point. Once you've created a tight edit, wind back a few bars and then play through your edit, listening to the way the various instruments sound as you pass through the edit point. If there's a problem you may need to choose a different edit point, or perhaps use an asymmetrical cross-fade to achieve the desired effect.



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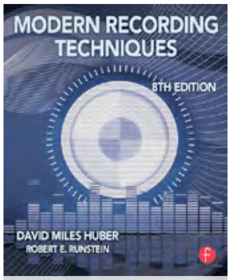
When you have an edit that just won't work, try changing the crossfade parameters.

FOR MORE INFORMATION ON THIS TOPIC, PLEASE CHECK OUT THE FOLLOWING CREATIVELIVE VIDEO COURSE:

Check out Fundamentals of Mixing Rock and EDM HERE.

Check out Gear Gods presents Studio Pass HERE.



The following is excerpted from *Modern Recording Techniques* by David Miles Huber and Robert E. Runstein.

You can purchase a copy HERE.

The mastering process is an art form that uses specialized, high-quality audio gear in conjunction with one or more sets of critical ears to help the artist, producer and/or record label attain a particular sound and feel before the recording is made into a finished manufactured product. Working with such tools, a mastering engineer or experienced user can go about the task of shaping and arranging the various cuts of a project into a final form that can be replicated into a salable product (Figures 19.1 and 19.2).

In past decades, when vinyl records ruled the airwaves and spun on everyone's sound system, the art of transferring high-quality sound from a master tape to a vinyl record was as much of a carefully guarded art form as it was technology. Because this field of expertise was (and still is) well beyond the abilities of most engineers and producers, the field of vinyl mastering was left to a very select few.

The art of transferring sound to a CD, DVD, Bluray or downloadable media is also still very much an art form that's often best left to those who are familiar with the tools and trade of getting the best sound out of a project. However, recent advances in computer and effects processing technology have also made it much easier for producers, engineers and musicians to own high-quality hardware and software tools that are fully capable of creating a professional-sounding final product in the studio or on a desk/laptop computer.

FIGURE 19.1
Emily Lazar, chief mastering engineer, The Lodge, New York City.
(Courtesy of The Lodge, www.thelodge.com)

FIGURE 19.2
Darcy Proper's mastering room at Wisseloord Studios, Hilversum, Netherlands
(Courtesy of Wisseloord Studios, www.wisseloord.nl, acoustics/photo by Jochen Velth, jv-acoustics.de)

THE MASTERING PROCESS

In addition to the concept of capturing the pure artistry of a production onto tape, hard disk or other medium, one of the primary goals during the course of recording a project is the overriding concept that the final product should have a certain "sound." This sound might be "clean," "punchy," "gutsy" or any other sonic adjective that you, the artist, the producer or the label might be striving for. Of course, once all of the cuts have been mixed, you'll hopefully be able to sit back and say, "Yeah, that's it!" If this isn't the case, having an experienced mastering engineer help you to "shape" the project's sonic character through the careful use of level balancing, dynamics and EQ could help save your sonic Technicolor day.

Another factor that can affect a project's sound is the reality that recordings may have been recorded and/or mixed in several studios, living rooms, bedrooms and/or basements over the course of several months or years. This could mean that the cuts would actually sound different from each other. In situations like this, where a unified, smooth sound might be hard to attain, it's even more important that someone who's experienced at the art of mastering be sought out.

In essence, the process of successfully mastering can help a project:

- *Sound "right"*: This is often accomplished through the use of careful EQ matching and dynamics processing. As was previously mentioned, this process not only takes the right set of processing gear, but also requires experienced ears that intuitively know how the project will most likely sound under a wide range of playing conditions.
- Be in the right sequential order: Choosing a project's song order is an art form that's best done by the artist and/or producer to convey the overall "feel" of a project.
- Have the proper timings between songs. The intuitive process of setting the gap times between songs can also make the difference between having awkward pauses and a project that "flows" smoothly from one cut to the next. Just remember, automatically setting the "gaps" to their default 2-second settings will probably not be in the project's best interest.
- *Playback at optimum levels*: Traditionally, the industry as a whole tends to set the average level of a project at the highest possible value. This is often due to the fact that record companies will always want their music to "stand out" above the rest when played on the TV, radio, MP3 player or the Web. This is usually accomplished by applying compression to the track or overall project. Again, this is an artistic

technique that often requires experience. Over-compression can lead to audible artifacts or a sound that can "squash" the life out of your hard-earned sound. In fact, light compression or even no compression at all is also an alternative. Classical music lovers, for example, often spend big bucks to hear a project's full dynamic range.

- *Match levels throughout the project*: In addition to getting the best overall level, it's often important that levels be properly matched when transitioning from one song to the next. The goal is to improve the flow and professionalism of a project by keeping songs from sticking out like a sore thumb.

The equipment that deals with the art and technology of creating a finished "master" is available in many different guises. Often, top-level mastering engineers will use specially designed EQ, dynamics and level matching gear that often won't commonly be found in the recording studio environment. Having said this, with the advent of CD and DVD burning software, dedicated hardware and software processing systems are now on the market that give musicians, producers and engineers a greater degree of control over the final mix and/or finished master than ever before.

TO MASTER OR NOT TO MASTER—WAS THAT THE QUESTION?

As mentioned, the process of mastering often requires specialized technical skills, audiophile equipment, a carefully tuned listening environment and talented ears in order to pull a sonic rabbit out of a problematic hat or even one that could simply use some dressing up.

A few years back, I had the good fortune of sitting around a big table at a top LA restaurant with some of the best mastering engineers in the US. As you might expect, the general consensus was that it's never a good idea for artists to master their own project ... that they're just too close to it to be objective. I'm not sure I agree that this is always the case. However, when approaching the question of whether to master a project yourself or to have a project professionally mastered, it's important that you objectively consider the following questions:

- Are you objective enough to have a critical ear for the sound and general requirements of the project, or are you just too close to it emotionally? (realizing that the "sound" of a particular project might follow you throughout your career)
- Is your equipment and/or listening environment adequate for the task?

- Is the final mix adequate (or more than adequate) for its intended purpose, or would the project benefit most from an outside set of professional ears?
- Does the budget allow for the project to be professionally mastered?

If the services of a professional mastering engineer are sought, the next question to ask is "Who'll do the mastering?" In this instance, it's important that you take a long hard look at the experience level of the person who will be doing the job and make it your job to familiarize yourself with that person's work and personal style. In fact, it's probably best to borrow from the traditional business practice of finding three of the most appropriate mastering house/engineer facilities to bid for the job and follow due diligence in making your decision by considering the following:

- What is their mastering track record?
- Are you familiar with examples of their work? If not, you should definitely ask for a client list and recent examples of their work ... then, have a critical listening session with the producer and/or band members.
- Are they familiar with your music genre as well as the type of "sound" that you're hoping to achieve?
- What are their hourly or project rates? Do they fit your budget?
- Are they willing to do a complementary test mastering session on one of your cuts?

Bottom line: Beware of the inexperienced mastering engineer (especially if that person is you). Once the choice of an experienced professional has been made, it's often wise for you/your band and the producer to personally sit in on the mastering session. Make sure that there are several ears around to listen to the project and listen over several types of systems. And above all, be patient, be critical of the project's sound and listen to the opinions of others. Sometimes you get lucky and the mastering process can be quick and painless; at other times it takes the right gear, keen ears and lots of careful attention to detail.

TO MASTER OR NOT TO MASTER THE PROJECT YOURSELF —THAT'S THE NEXT QUESTION!

Obviously, there are different production levels for deciding whether or not to master. Is it a national major release? Is it your first self-produced project? Is there virtually no budget or did the group blow it all on the release party? These and countless other questions come into play when choosing how a project is to be mastered.

Having said this, mastering a project (whether it's your own or not) is increasingly becoming an available option. If you do your own, you should understand a few basic concepts:

- The basic concept of mastering is to present a song or project in its best possible sonic "light." This requires a basic skillset, level of "comfort- ability" and knowledge of your room's equipment, acoustics and overall downfalls.
- A basic level of detachment is often required during the mastering process. This is a lot harder than it sounds, as sound is a very subjective thing. It's often easy to fool yourself into thinking that your room, setup and sound are absolutely top-notch ... and then when you go to play it on another professional system (or any home system) ... your hopes can be dashed in an instant. One of the best ways to guard against this is to listen to your mix on as many systems as possible and ask others to critique your work. How does it sound to them on their system?
- As with mixing, mastering is a process of balancing all of the variables that go into making a project sound as good as it can. One of the best ways to hear if you're on the right track or not, is to create a "my favorite songs" directory or CD that includes mixes by artists and producers that you love and admire. By listening to your project alongside of your favorite "best of" recordings, you can often get quick insights into how your work sounds compared to the work of others.

If you're willing to put in the time and effort, it's my belief that mastering your own project is definitely an option.

"PRE"PARATION

Just as one of the best ways to make the mixdown process go more smoothly is to fix most of your technical problems and issues BEFORE you start the mixdown process, one of the best ways to ensure that a mastering session has as few problems as possible is to ask the right questions and deal with the technical issues BEFORE the mastering engineer even receives your soundfiles. By far, the best way to avoid problems during this phase is to ask questions ahead of time. The mastering engineer should be willing to sit down with you or your team to discuss your needs and preplanning issues (or at least direct you to a document checklist that can help you through the preparation process). During this introductory "getting to know you and your technical requirements" session, here are just a few questions that you might ask:

- What should the final master sample rate and bit rate be?

- What should the master's maximum level be?
- Should all master compression be turned off? Would you like for us to supply you a copy with the bus compression turned on, so you can hear our intended ideas?
- Would you like separate instrument/vocal stem tracks, so they can be treated separately?
- Are there any special requirements that we should be aware of?

IMPORTANT NOTES TO REMEMBER

- It's ALWAYS wise to consult with the project's mastering engineer about the general specifications of the final product BEFORE beginning the mixdown (or possibly even the recording) process. For example, that person might prefer that the files be recorded and/or mixed at a certain bit rate and bit depth, as well as in a specific format.
- A mastering engineer might prefer that the final master be changed or processed as little as possible with regard to normalization, fade changes, overall dynamic changes (compression) and applied dither. These processing functions are best done in the final mastering process by a qualified engineer.

MASTERING THE DETAILS OF A PROJECT

From the mastering standpoint (as well as those of the artists and producer who are overseeing the project), a wide range of artistic decisions need to be carefully finessed in order to master a recording into its final, approved form. Just a few of the decision-making steps include:

- Choosing the proper song order
- The use of level changes to balance the relative track levels and improve the overall "feel" of the project
- The application of EQ to improve the sound and overall "tone" of the project
- The judicial use of dynamics to balance out the sound and increase the project's overall level.

SEQUENCING: THE NATURAL ORDER OF THINGS

Whether the master is to be assembled using analog tape, or on a DAW/editor, the running order in which the songs of a project will be played often affects the overall flow and tone of a project. The considerations for which song follows which is infinitely varied, and can only be garnered from experience and having an artistic "feel" for how their order and interactions will affect the listening experience. Of course, sequence decisions are probably best made by the artist and/or producer, as they have the best feel for the project. A number of variables that'll directly affect the sequenced order of a project include:

- *Total length*: How many songs will be included on the disc or album? If you've recorded extra songs, could we include the Bonus Tracks on the disc? Is it worth adding a weaker song, just to fill up the CD?
- *Running order*: Which song should start? Which should close? What order feels best and supports the overall mood and intention of the project?
- *Transitions*: Altering the transition times between songs can actually make the difference between an awkward silence that jostles the mood and a transition that keeps up with the pace and feel of the project. The Red Book CD standard calls for 2 seconds of silence as a default setting between tracks. Although this is necessary before the beginning of the first track, it isn't at all the law for spacings that fall between the first and second or later songs. Most editors will allow you to alter the index space timings between tracks from 00 seconds (butt splice) to longer gaps that help maintain the appropriate transitional mood.
- *Cross-fades*: In certain situations, the transition from one song to the next is best served by cross-fading from one track directly to the next. Such a fade could seamlessly intertwine the two pieces, providing the glue that can help convey any number of emotional ties.

DIGITAL SEQUENCE EDITING

With the advent of the DAW, the relatively cumbersome process of sequencing music tracks in the analog domain using magnetic tape has given way to the faster, easier and more flexible process of editing the final masters from hard disk. In this process, all of the songs can be loaded into a workstation, and then adjusted with respect to volume, equalization, dynamics, fades and timing (song start and endpoints) for assembly into a final, edited form.

Although the length of time between the end of one song and the beginning of the next can be constant, this isn't always the wisest choice, as the timing between songs can have a profound effect upon the "feel" of a transition from one song to the next. Decreasing the time between them can make a song seem to blend into the next (if they're similar in mood) or could create a sharp contrast with the preceding song (if the moods are dissimilar). Longer times between songs help the listeners get out of the mood of the previous song and prepare them for hearing something that might be quite different.

Another way to vary the gaps between songs is to do the process manually. This can be done in the DAW by placing the gap times directly within the session itself. In this way, the song can be exported (including blank silence at the end), so that project songs can be entered into any regular CD burning program and burned to a final CD disc (remembering to enter in a "0 sec" timing into the global settings).

It's always a good idea to make at least one master copy of the final mix sound-files, session data and master files as a backup (just in case the record company, producer or artist wants to make changes at a later date). This simple precaution could save you lots of time and frustration.

ANALOG SEQUENCE EDITING

Although the process of assembling a final master in the analog domain occurs far less frequently than in the digital realm, it's still done. During this process, the engineer edits the original mixes out from their respective reels and begins the process of splicing them together into a final sequence on a master reel set. At this time, the level test tones (which were laid down at the beginning of the mixdown session) should be placed at the beginning of side one. Once this is done, the mix master in/out edits should be tightened (to eliminate any noise and silence gaps) by listening to the intro and outro at high volume levels, while the heads are in contact with the tape (this might require that you place the transport into the edit mode). The tape can then be moved back and forth (a process known as "jogging" or "rocking" the tape) to the exact point where the music begins (intro) and after it ends (outro). Once the in (or out) point is positioned over the playback head, the exact position is marked with a grease pencil. If there's no noise directly in front of this spot, it's a good practice to cut the tape half an inch before the grease pencil mark as a safety precaution against editing out part of the first sound. If there is noise ahead of the first sound, the tape should be cut at the mark and a leader inserted at that point. Paper (rather than plastic) leader tape is used because plastic will often cause static electricity pops.

The tail of the song might need to be monitored at even higher volume levels because it's usually a fade-out or an overhang from the last note and is, therefore, much softer than the beginning of the song. The tape is marked and cut just after the last sound dies out to eliminate any low-level noises and tape hiss. Of course, the length of time between the end of the song and the beginning of the next can be constant in a sequenced project, or the timings can vary according to the musical relationship between the songs.

RELATIVE VOLUMES

In addition to addressing the overall volume levels of a project, one of the tasks of the mastering process is to smooth out the relative volume differences between songs over the course of the disc or album. These differences could occur from a number of sources, including general variations in mixdown and program content levels, as well as levels between projects that have been mixed at different studios.

Cues as to smoothing out the relative rms and peak differences can be obtained by:

- Using your ears to fine-tune the volume levels from song to song
- Looking at the general attributes of a soundfile from within a digital audio editor
- Carefully watching the master output meters on a recorder or editor
- Watching the graphic levels of the songs as they line up in a digital audio editor

Contrary to popular belief, the use of a standard DAW normalization tool can't smooth out these level differences, because this process only detects the peak level within a soundfile and raises the overall level to a determined value. Since the average (rms) and peak levels will often vary widely between the songs of a project, this toll isn't always useful, although certain editors provide normalization tools that have more variables that are more useful and in depth.

EQ AND MASTERING CONTROL

As is the case in the mixdown process, equalization is often an extremely important tool for boosting, cutting or tightening up the low end, adding presence to the midrange and tailoring the high end of a song or overall project. Again, EQ can be used as a tool to smooth out differences between cuts or for making changes that affect the overall character of the entire project. Of course, wide ranges of hardware and software plug-in EQ systems are available for applying the final touches both within a studio and project-based setting **(Figures 19.3 and 19.4)**.

FIGURE 19.3
iZotope's Ozone 5 Mastering Plug-In System with maximizer, equalizer, dynamics, imaging, exciter, reverb, and dithering.
(Courtesy of iZotope Inc., www.izotope.com)

FIGURE 19.4
Maselec MTC-1 Mastering Transfer Console.
(Courtesy of Mases Electronics Ltd., www.maselec.com)

DYNAMICS

One of the most commonly used (and overused) tools within the mastering process relates to dynamics processing, or specifically, compression **(Figures 19.5 through 19.7)**.

FIGURE 19.5
Manley Stereo Variable Mu® mastering outboard compressor.
(Courtesy of Manley Laboratories, Inc., www.manleylabs.com)

FIGURE 19.6
The Shadow Hills Mastering Compressor plug-in for the Apollo and the UAD effects processing card.
(Courtesy of Universal Audio, www.uaudio.com ©2013 Universal Audio, Inc. All rights reserved. Used with permission).

FIGURE 19.7
EMI TG12412 mastering outboard EQ plug-in.
(Courtesy of Digidesign, a division of Avid, www.digidesign.com)

Although the general name-of-the-game is to achieve the highest overall average level within a song or complete project, care must be taken so as not to apply so much compression that the life gets dynamically sucked out of the sound (Figure 19.8). As with the first rule in recording – "There are no rules" – the amount of dynamics processing is entirely up to those who are creatively involved in the final mastering process. However, it's important to keep in mind the following guidelines:

- Depending on the program content and genre, the general dynamic trend is toward raising the overall levels to as high a point as possible.
- When pushed to an extreme, compression will often have an intended or unintended side effect of creating a sound that has been "squashed," giving a "wall of sound" character that's thick (a good thing) and/or one that's sonically lifeless (a bad thing).
- When compression is not applied (or little is used), the sound levels will often be lower, thinner (that might be bad) or full of dynamic life (that's good).

From all of this, you'd be correct if you said that the process is entirely subjective and relative! The use of compression can help add a strong presence to a recording, while overuse can actually kill its dynamic life ... so use it wisely!

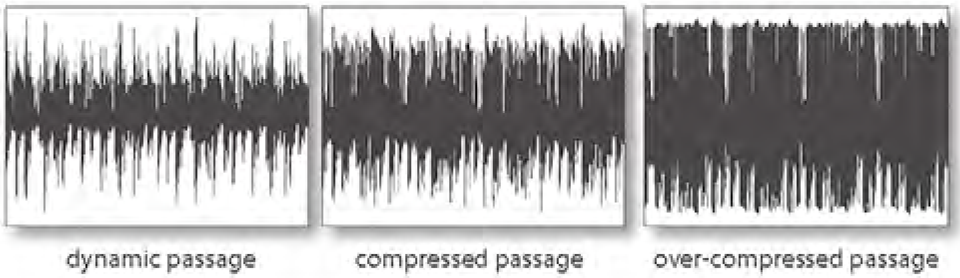


FIGURE 19.8
Figure showing the same passage with varying degrees of compression.

MULTIBAND DYNAMIC PROCESSING

The modern-day mastering process often makes use of multiband dynamic processing (Figure 19.9) in order to break the frequencies of the audio spectrum into bands that can be individually processed. Depending on the system, for example, up to five distinct bands might be available for processing the final signal. Such a hardware or software system could be used to strongly compress the low frequencies of a song using a specific set of parameters, while applying only a small amount of compression to the sibilance at its upper end.

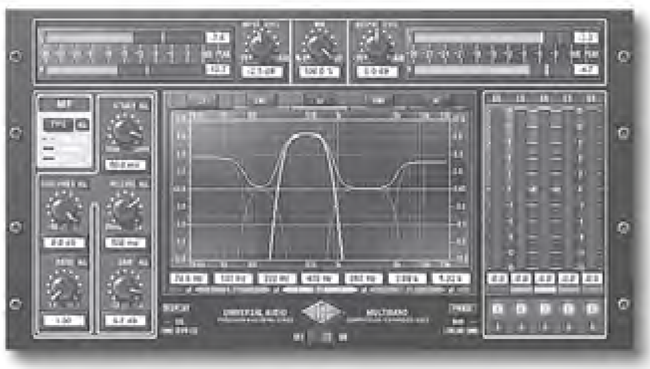


FIGURE 19.9
Precision Multiband Plug-in for the Apollo and the UAD effects processing card.
(Courtesy of Universal Audio, www.uaudio.com ©2013 Universal Audio, Inc. All rights reserved. Used with permission).

SOUNDFILE VOLUME

While dynamics is a hotly debated topic among mastering and recording engineers, most agree that it's never a good idea deliver a soundfile to a mastering house that has been compressed and raised in level. This, of course, effectively limits the dynamics of the song and takes control over the final dynamics and sound of the song away from the mastering engineer. For this reason, most mastering engineers will ask that the project be delivered without any compression or dynamics of any kind ... and that overall session gain be reduced so that it peaks at a lower level (-12dB for example). Again, it's always a good idea to check the facts beforehand.

SOUNDFILE RESOLUTION

The sample rate and bit rate resolution that a project is recorded at is a personal matter (as with many things in sound recording). Given that high-quality converters and a DAW with a high-quality bit-processing structure are used, many believe that recordings made with 44.1-kHz/16-bit file resolutions are sufficient to capture the full nuances of sound (for both the recorded tracks and the final mixdown). Others believe that rates upward to 96 kHz/24 bits or higher are necessary to capture the extended high frequencies and increased resolution of music.

One thing is for sure, if higher bit rates are chosen, it's almost always best to deliver the final master recording to the mastering lab at the original native rate (i.e., if the session was recorded and mixed at 24/96, the final mixdown resolution should delivered to the mastering engineer at that rate and bit resolution). I should mention, though, that some mastering engineers will prefer that the final track be exported at a higher bit-depth than the native session depth. For example, if the session was recorded at 44.1/16, many engineers will ask that the session be exported (bounced) to a soundfile at 44.1, but with a 24-bit depth. The reasoning behind this is that the mastering engineer will be able to process the mix at the higher, more detailed depth resolution and will have better tools by which to dither the soundfile down to the necessary 16/44.1 rate for final transfer to CD or the intended medium.

DITHER

The addition of small amounts of randomly generated noise to an existing bitstream can actually increase the overall bit resolution (and therefore low-level noise and signal clarity) of a recorded signal. Through the careful addition of dither, it's actually possible for signals to be encoded at levels that are less than the data's least significant bit level. You heard that right … by adding a small amount of random noise into the A/D path, the resolution of the conversion process can actually be improved below the least significant bit level and reduce a soundfile's harmonic distortion.

Within mastering, dither is often manually applied to soundfiles that have been recorded at 20- and 24-bit depths. DAW plug-ins can be used to apply dither to a soundfile or master mix, so as to reduce the effects of lost resolution due to the truncation of least significant bits. For example, mastering engineers might carefully experiment with applying dither to a high-resolution file before saving or exporting it as a 16-bit final master. In this way, noise is reduced and the soundfile's overall clarity is increased.

THE DIGITAL AUDIO EDITOR IN THE MASTERING PROCESS

FIGURE 19.10
Sadie PCM4 Editing System.
(Courtesy of Prism Media Products Ltd, www.sadie.com)

By far, the most commonly used system for modern-day mastering is the digital audio editor **(Figure 19.10)**. These two-channel and multichannel workstations make use of the personal computer's existing processing, disk storage and data I/O hardware to perform a wide range of audio editing, processing and mastering production tasks. These programs allow each song to be loaded onto its own track and be independently processed according to its own needs, while possibly adding overall effects processing to the master output section (allowing EQ, dynamics, dither, etc., to be applied to the entire project mix). In such a graphic, on-screen environment, individual songs can be imported, processed and exported into a final form that can then be burned to disc—either directly from within the editing program or by using a CD burning software application.

Of course, a DAW doesn't have to be especially designed for the mastering phase. In fact, most mastering is done using many of the workstations that are commonly available on the market **(Figure 19.11)**.

FIGURE 19.11
Example of a self-mastered project using Nuendo.
(Courtesy of David Miles Huber, www.davidmileshuber.com)

ON A FINAL NOTE

In closing, it's always a wise idea to budget in some time with the final mastered recording before committing it to the final product. If at all possible, take a week and listen to it in your car, on your boom box, home theater system, in another studio—virtually everywhere! As a musician, producer or record label, it will be your calling card for quite some time to come. Once you're satisfied with the sound of the finished product, then you can move from the mastering phase to making the project into a finished, salable product.

DISTRIBUTION

One of greatest misconceptions surrounding the music, visual and other media-related industries is the idea that once you walk out the door of a studio with your final master in hand, the creative process of producing a project is finally over. All that you have left to do is upload a file to your favorite online store or hand the CD over to a duplication facility and … Ta-Dah!—the adoring public will be clamoring for your product, website and merchandise. Obviously, this scenario is almost always far from the truth. Even before you have your blood, sweat and tears physically in hand, it's vitally important to think through and implement your master plan, if your product is to make it into the hands (or iDevice) of the consumer.

If you don't pre-plan, and follow through with these plans once the project is recorded, you can be fairly sure that your project will sit on a shelf. Or, worse, you'll have 1000 CDs sitting in your basement that'll never be heard—a huge shame given all the hard work that went into making it.

Early in this book, I told you about the first rule of recording ... that "there are no rules, only guidelines." This actually isn't true. There is one rule:

Given the huge number of changes that have occurred within the business of music marketing and distribution, it stands to reason that an equally large number of changes have occurred when it comes to planning and considering how you are going to get your new-born project out to the masses. In short, many of the rules have changed ... and it's the wise person who takes the time to put their best foot (and plan) forward to make it all shine. So, having said this ... what are our options in this day and age? Well there's:

- Online (download) distribution
- Hi-res download distribution
- Music Streaming
- CD
- DVD and Bluray
- Vinyl

Just as each recording facility has its own unique personality and particular "sound," the right mastering and physical duplication and even online download service might also have a profound effect on the outcome of a project. If a project is being underwritten and distributed by an independent or major record label, they will generally be fully aware of their production needs and will certainly have an established production and product manufacturing network in place. If, however, you're distributing the project yourself, the duty of choosing the best facility or manufacturing organization that'll fit your budget and quality needs will fall on your or your band's shoulders.

On the subject of online distribution, besides the important fact that your music has the potential to reach a large fanbase, the best part about downloadable media is that there's no expensive physical media that needs to be manufactured and distributed. Gone are the days when you have to give up a closet or area in your basement for storing CDs and LPs. However, the flipside to this is that the artist will need to research and understand their online distribution options, needs and legal responsibilities before signing up to a download service.

DOWNLOADABLE MEDIA

In this day of surfing and streaming media off the Web, it almost goes without saying that the WWW has become the most important and effective marketing tool for the musician and labels alike. It allows for us to cost-effectively upload, distribute, and promote our songs, projects, promotional materials, touring info and liner notes out to mass audiences. As with other media, mastering for the Internet can either be complicated, requiring professional knowledge and experience, or it can be a straightforward process that can be carried out from a laptop computer. It's a matter of meeting the level of professionalism and development that's required by you and your audience's needs.

Once the recording and mix phases of a project have been completed (assuming that you're also doing your business and promotion homework as to your audience, distribution methods, live and Web marketing presence, production budgeting, etc.), the next step toward getting the product out to the people is to transform the completed song or project into a form that can be mass produced, distributed, marketed and SOLD. Over the web, this will often mean direct sales to the public via popular download sites, such as iTunes, CDBaby, Amazon, etc. Although most sites will have their own system for having the artist upload their music, it's often wise to be aware of the final codec, bitrate and overall look/feel of the final product. The definitions and details for the most popular distribution formats and bitrates can best be found in Chapter 10. Considerations for these final media formats might include the concept of mixing for your intended audience, taking care to take final levels, mastering dynamics and EQ, as well as the overall look and attitude of the final product.

HI-RES DOWNLOADS

Over the last decade, the concept of quality vs. data compression size has had some time to shake out a few cobwebs. With improvements in data download transfer rates and increased device memory storage, the problems, constraints and need for smaller, more "compressed" data soundfiles have greatly reduced. Given the recent advances in bandwidth, storage and memory storage, the buying public is now asking for a return to uncompressed (or less-compressed), higher-quality downloads, allowing them to enjoy their favorite music in its originally-intended fidelity. For example, such higher-sample-bitrate files might be encoded and distributed as 320kps MP3, AAC, FLAC or even WAV.

MUSIC STREAMING

Although music downloads are one of the strongest-growing markets in the music industry, another market that's also showing huge growth potential is the online music and media streaming market. Unlike downloads (where the data is physically stored onto the buyer's computer and/or music playback device), media streaming actually stays in the "clouds" ... that's to say, stays on the provider's server and is "streamed" in real-time to the authorized "pay-per-month" buyer or free service listener. With the advent of "smart phones, TVs and media centers, data streams can be enjoyed from any number of media devices ... not just from a computer.

PHYSICAL MEDIA

Although downloadable and streaming media are well on their way to dominating the music and media industries, the ability to buy, own and gift a physical media object still warms the hearts of many a media product purchaser. Often, containing uncompressed, hi-quality music, these physical discs and records are held in regard as something that can be held onto, looked at and packaged up with a big, bright gift bow. Their market-share days may be numbered ... but don't count them out any time soon.

THE CD

Beyond the process of distributing audio over the Internet (using an online service or from your own site), as of this writing, the compact disc (CD) is still a strong and viable medium for distributing music. These 120-mm silvery discs **(Figure 20.1)** contain digitally encoded information (in the form of microscopic pits) that's capable of yielding playing times of up to 74 or 80 minutes at a standard sampling rate of 44.1 kHz.

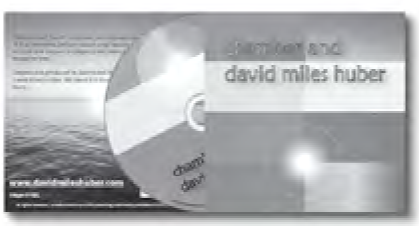


FIGURE 20.1
The compact disc.
(Courtesy of www.davidmileshuber.com)

The pit of a CD is approximately half a micrometer wide, and a standard manufactured disc can hold about 2 billion pits. These pits are encoded onto the disc's surface in a spiraling fashion, similar to that of a record, except that 60 CD spirals can fit into the groove of a single long-playing record. These spirals also differ from a record in that they travel outward from the center of the disc, are impressed into the plastic substrate, and are then covered with a thin coating of aluminum (or occasionally gold) so that the laser light can be reflected back to a receiver. When the disc is placed in a CD player, a low-level infrared laser is alternately reflected and not reflected back to a photosensitive pickup. In this way, the reflected data is modulated so that each pit edge represents a binary 1, and the absence of a pit edge represents a binary 0 **(Figure 20.2)**. Upon playback, the data is then demodulated and converted back into an analog form.

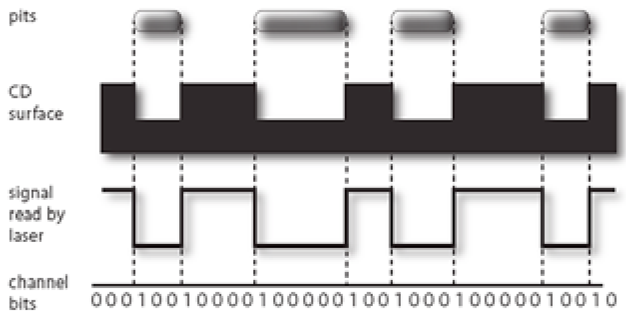


FIGURE 20.2
Transitions between a pit edge (binary 1) and the absence of a pit edge (binary 0).

Songs or other types of audio material can be grouped on a CD into tracks known as "indexes" This is done via a subcode channel lookup table, which makes it possible for the player to identify and quickly locate tracks with frame accuracy.

Subcodes are event pointers that tell the player how many selections are on the disc and where their beginning address points are located. At present, eight subcode channels are available on the CD format, although only two (the P and Q subcodes) are used.

Functionally, the CD encoding system splits the 16 bits of information into two 8-bit words with error correction (that's applied in order to correct for lost or erroneous signals). In fact, without error correction, the CD playback process would be so fragile and prone to dropouts that it's doubtful it would've become a viable medium. The system then goes about translating this data (using a process known as eight-to-fourteen modulation or EFM) into a special code, known as a data frame. Each data frame contains a frame-synchronization pattern (27 bits) that tells the laser pickup beam where it is on the disc. This is then followed by a 17-bit subcode word, 12 words of audio data (17 bits each), 8 parity words (17 bits each), 12 more words of audio, and a final 8 words of parity data.

THE PROCESS

In order to translate the raw PCM of a music or audio project into a format that can be understood by a CD player, a compact disc burning system must be used. These come in two flavors:

- Specialized hardware/software that's used by professional mastering facilities
- Disc burning hardware/software systems that allow a personal computer to easily and cost effectively burn CDs.

Both system types allow audio to be entered into the system, after which the tracks can be assembled into the proper order and the appropriate gap times can be entered between tracks (in the form of index timings). Depending on the system, cuts might also be processed using cross-fades, volume, EQ and other parameters. Once assembled, the project can be "finalized" into a media form that can be directly accepted by a CD manufacturing facility. By far, the most common media that's received by CD pressing plants for making the final master disc are user-burned CD-Recordable (CD-R) discs, although some professional systems will still make use of a special Exabyte-type data tape system.

Note that not all CD-R media are manufactured using high-quality standards. ... In fact, some are so low in quality, that the project's data integrity could be jeopardized.

As a general rule:

- It's always good to use high-quality "master-grade" CD-Rs to burn the final master (you can sometimes see the difference in pit quality with the naked eye).
- It's always best to send two identical copies to the manufacturer (just in case one fails).
- Speaking of failing, you should check that the manufacturer has run a data integrity check on the final master to ensure that there are few to no errors.

Once the manufacturing plant has received the recorded media, the next stage in the process is to cut the original CD master disc. The heart of such a CD cutting system is an optical transport assembly that contains all the optics necessary to write the digital data onto a reusable glass master disc that has been prepared with a photosensitive material.

After the glass master has been exposed using a special recording laser, it's placed in a developing machine that etches away the exposed areas to create a finished master. An alternative process, known as nonphotoresist, etches directly into the photosensitive substrate of the glass master without the need for a development process.

After the glass or CD master disc has been cut, the compact disc manufacturing process can begin (Figure 20.3). Under extreme clean-room conditions, the glass disc is electroplated with a thin layer of electroconductive metal. From this, the negative metal master is used to create a "metal mother," which is used to replicate a number of metal "stampers" (metal plates which contain a negative image of the

CD's data surface). The resulting stampers make it possible for machines to replicate clear plastic discs that contain the positive encoded pits, which are then coated with a thin layer of foil (for increased reflectivity) and encased in clear resin for stability and protection. Once this is done, all that remains is the screen-printing process and final packaging. The rest is in the hands of the record company, the distributors, marketing and you.

FIGURE 20.3
Various phases of the CD manufacturing process:
(a) the lab, where the CD mastering process begins;
(b) once the graphics are approved, the project's packaging can move onto the printing phase;
(c) while the packaging is being printed, the approved master can be burned onto a glass master disc;
(d) next, the master stamper (or stampers) is placed onto the production line for CD pressing;
(e) the freshly stamped discs are cooled and checked for data integrity;
(f) labels are then silk-screen printed onto the CDs; and
(g) finally, the printed CDs are checked before being inserted into their finished packaging.
(Courtesy of Disc Makers, Inc., www.discmakers.com)

Whenever possible, it's always wise and extremely important that you be given art proofs and test pressings BEFORE the final products are mass duplicated.

As with any part of the production process, it's always wise to do a full background check on a production facility and even compare prices and services from at least three manufacturing houses. Give the company a call, ask a few questions and try to get a "feel" for their customer service abilities, their willingness to help with layout questions, etc. You'd be surprised about how much you can learn in a short time. Once you've found a few companies that seem to fit your needs, ask them for a promotional pack (which includes product and art samples, service options and a price sheet). You might also want to ask about former customers and their contact information (so you can email them about their experiences).

Once you've settled on a manufacturer, it's always a good idea to research what their product and graphic arts needs and specs are before delving into this production phase. When in doubt about anything, give them a call and ask; they are there to help you get the best possible product (besides, asking questions helps to get the job done right the first time, with less mess, fuss, time and $$$).

The absolute last thing that you or the artist wants is to have several thousand discs arrive on your doorstep that are ... WRONG! Receiving a test pressing and graphic "proof" is almost always well worth the time and money. It's never wise to assume that a manufacturing or duplication process is perfect and doesn't make mistakes. Remember, Murphy's Law can pop up at any time!

CD BURNING

Software for burning CDs/DVDs/Blu-ray media is available in various forms for both the Mac and PC. These include the simple burning applications that are included with the popular operating systems, popular third-party burning applications and more complex authoring programs that are capable of editing, mastering and assembling individual cuts into a final burned master **(Figure 20.4)**.

FIGURE 20.4
DiscWelder Steel CD/DVD audio authoring tool for the PC.
(Courtesy of Minnetonka Audio Software, Inc., www.minnetonkasoftware.com)

There are numerous ways in which a CD project can be prepared and burned. For starters, it's a fairly simple matter to prepare and master individual songs within a project and then load them into a program for burning. Using this method, any program should be able to burn the audiofiles in a straightforward manner from beginning to end. Keep in mind that the Red Book CD standard specifies a beginning header silence (pause length) that's 2 seconds long. After this initial lead-in, any pause length can be user specified; the default setting for silence between cuts is 2 seconds, however, any musician/producer will tell you that these lengths will vary, as one song might want to flow directly into another, while the next might want a longer pause to help set the proper mood (it's an artistic "feel" kinda thing).

Most CD burning programs will allow you to enter "CD Text" information (such as title, artist name/copyright and track name field code info) that is then written directly into the CD's subcode area. This can be a helpful tool, because important artist, copyright and track identifiers can be directly embedded within the CD itself and will be automatically displayed on most modern CD players. As a result, illegal copies will still contain the proper copyright and artist info, and discs that are loaded into a computer or media player will often display these fields. While you're at it, you should check with the manufacturer to see if they're able to enter your track and project information into iTunes and Gracenote. These databases (which should not be confused with CD Text) allow CD titles, artist info, song titles and graphics to appear on most media-based music players ... an important feature for any artist and label.

As always, careful attention to the details should always be made, making sure that:

- High-quality media is used
- The media is burned using a stable, high-quality drive
- The media is carefully labeled using a recommended marking pen
- Two copies are delivered to the manufacturer, just in case there are data problems on one of the discs

ROLLING YOUR OWN

With the rise of Internet music distribution and the steady breakdown of the traditional record company distribution system, bands and individual artists have begun to produce, market and sell their own music on an ever-increasing scale **(Figure 20.5)**. This concept of the "grower" selling directly to the consumer is as old as the town square produce market. By using the global Internet economy,

FIGURE 5.1
CD Baby is a distribution service that helps sell physical CDs and downloadable music over the Internet.
(Courtesy of CD Baby, www.cdbaby.com).

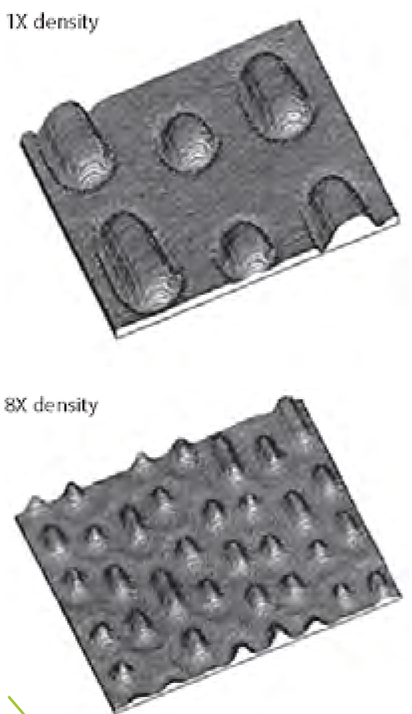


FIGURE 20.6
Detailed relief showing standard CD and DVD pit densities.

independent distribution, fanzines, live concert sales, etc., savvy independent artists are taking matters into their own hands by learning the inner workings of the music business. In short, artists are taking business matters more seriously in order to reap the fruits of their labor and craft … something that has never been and never will be an easy task.

Beyond the huge tasks of marketing, gigging and general business practices, many musicians are also taking on the task of burning, printing, packaging and distributing their own physical products from the home or business workplace.

This homespun strategy allows for small runs to be made in an "on-demand" basis, without tying up financial resources and storage space with CD inventories.

Creating a system for burning CD-Rs for distribution can range from being a simple home computer setup that creates discs on an individual basis to sophisticated replication systems that can print and burn stacks of CD-Rs or DVD-Rs under robot control at the simple touch of a button.

DVD AND BLU-RAY BURNING

Of course, on a basic level, DVD burning technology has matured enough to be available and affordable to the general Mac and PC public. From a technical standpoint, these CD-drive compatible discs differ from the standard CD format in several ways. The most basic of these are:

- An increased data density due to a reduction in pit size **(Figure 20.6)**

- Double-layer capabilities (due to the laser's ability to focus on two layers of a single side)
- Double-side capabilities (which again doubles the available data size)

In addition to the obvious benefits that can be gained from increasing the data density of a standard CD from 650 Mbytes to a maximum of 17 Gbytes, DVD discs allow for much higher data transfer rates, making DVD the ideal medium for the following applications:

- The simultaneous decoding of digital video and surround-sound audio
- Multichannel surround sound
- Data- and access-intensive video games
- High-density data storage

As of this writing, Blue-ray disc (BD) technology has established itself as the video delivery media du jour. (Of course, this might be old hat by the time you read this.) Because Bluray uses a shorter wavelength red laser (650 nm) and smaller pit densities than a DVD, the capacity has risen to 25 Gbytes on a single disc (or 50 on a dual-layer)!

As DVD and BD-R/RW drives have become commonplace, affordable data backup and mastering software has come onto the market that brought the art of video and Hi-Def production to the masses. Even high-level DVD media mastering is now possible in a desktop environment, although creating a finished product for the mass markets is often an art that's best left to professionals who are familiar with the finer points of these complex technologies.

DISC HANDLING AND CARE

Here are a few basic handling tips for optical media (including the recordable versions) from the National Institute of Standards and Technology:

DO:

- Handle the disc by the outer edge or center hole (your fingerprints may be acidic enough to damage the disc).
- Use a felt-tip permanent marker to mark the label side of the disc. The marker should be water or alcohol based. In general, these will be labeled as a non-toxic CD/DVD pen. Stronger solvents may eat though the thin protective layer to the data.

- Keep discs clean. Wipe with a cotton fabric in a straight line from the center of the disc toward the outer edge. If you wipe in a circle, any scratches may follow the disc tracks, rendering them unreadable. Use a disc-cleaning or light detergent to remove stubborn dirt.
- Return discs to their cases immediately after use.
- Store discs upright (book-style) in their cases.
- Open a recordable disc package only when you are ready to record.
- Check the disc surface before recording.

DON'T:

- Touch the surface of a disc.
- Bend the disc (because this may cause the layers to separate).
- Use adhesive labels (because they can unbalance or warp the disc).
- Expose discs to extreme heat or high humidity; for example, don't leave them in direct sunlight or in a car.
- Expose discs to extreme rapid temperature or humidity changes.
- Expose recordable discs to prolonged sunlight or other sources of ultraviolet light.

ESPECIALLY DON'T:

- Scratch the label side of the disc (it's often more sensitive than the transparent side).
- Use a pen, pencil or fine-tipped marker to write on the disc.
- Try to peel off or reposition a label (it could destroy the reflective layer or unbalance the disc).

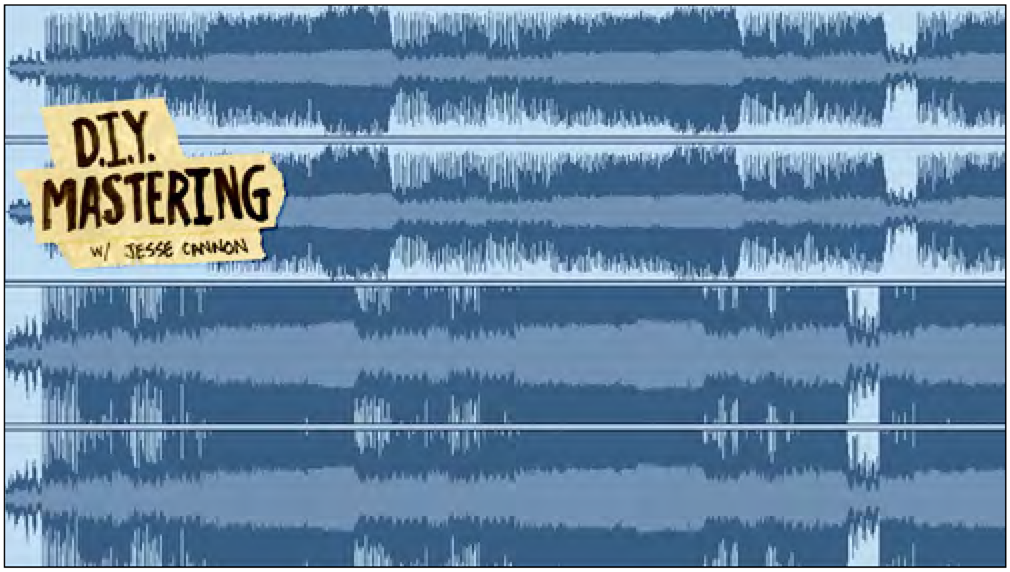
DISC LABELING

Once you've burned your own CD-R/RW or DVD-R/RW, there are a number of options for printing labels onto the newly burned discs (burning the disc first will often reduce data errors that can be introduced by dust, fingerprints or scratches due to handling):

- *Use a felt-tip pen*: This is the easiest and fastest way to label a disc. However, water-based ink pens should be used, because most permanent markers use a solvent that can permeate the disc surface and cause damage to either the reflective or dye layer. When properly done, this is an excellent option for archiving discs.

- *Use a disc printer*: Specially designed ink-jet or laser printers are able to print high-quality, full-color layouts onto the face of a printable (white or silver-faced) disc. This is a cost-effective option for those who burn discs in small batch runs, but still want a professional look and feel.
- Use a disc manufacturing company to professionally print a limited run of CD-Rs that can be burned on-demand. These CDs are often printed using the same high-quality screen printing techniques that are used to print mass-duplicated discs.

FOR MORE INFORMATION ON THIS TOPIC, PLEASE CHECK OUT THE FOLLOWING CREATIVELIVE VIDEO COURSE:



Check out the rest of DIY Mastering HERE.

Check out the rest of Gear Gods presents Studio Pass HERE.