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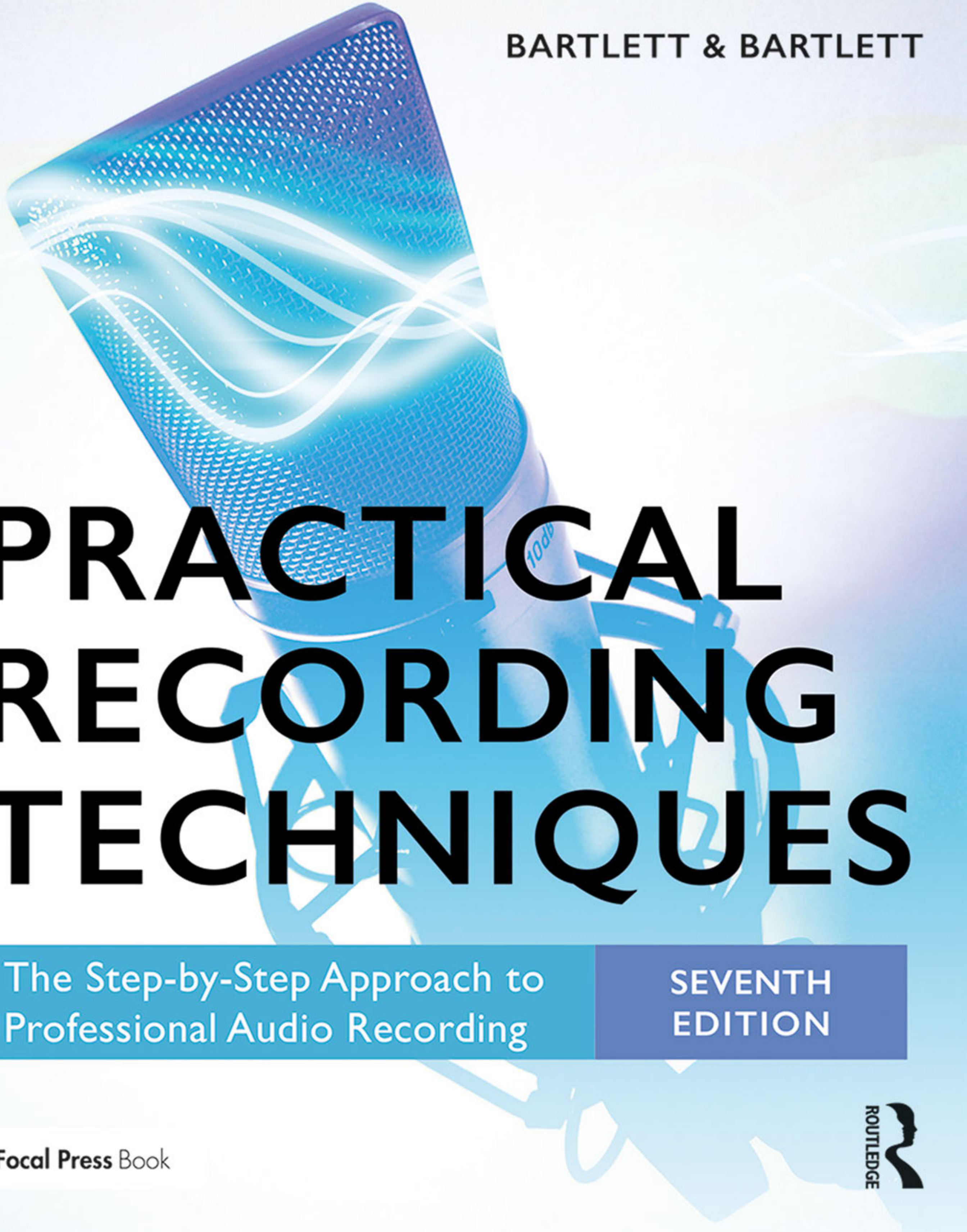

PRACTICAL RECORDING TECHNIQUES

The Step-by-Step Approach to Professional Audio Recording

SEVENTH EDITION

A Focal Press Book

ROUTLEDGE

Practical Recording Techniques

Practical Recording Techniques covers all aspects of recording, perfect for beginning and intermediate recording engineers, producers, musicians, and audio enthusiasts. Filled with tips and shortcuts, this hands-on, practical guide gives advice on equipping a home studio (whether low budget or advanced) and suggestions for setup, acoustics, effects, choosing mics and monitor speakers, and preventing hum. This best-selling guide also instructs how to mike instruments and vocals, judge recordings and improve them, work with MIDI and loops, do mastering, and put your music on the Web. Two chapters cover live recording of classical and popular music.

New in the seventh edition:

- Complete update of all types of recording equipment, plug-ins, and recording software
- Increased focus on current industry and classroom trends like DAW signal flow and operation (during recording and mixdown), while still covering analog fundamentals
- Updated organization to focus and break up topics
- Updated tips on optimizing your computer for multitrack recording—for both Windows and Mac
- New sections on streaming audio, mobile-device recording, live recording with digital consoles, and psychoacoustics
- Listen Online boxes highlight where audio samples on the website relate to chapter discussions
- Updated companion website at www.routledge.com/cw/bartlett with audio examples, articles, and suggested activities, plus expanded and more user-friendly links to the best sites for videos and articles, recording techniques, equipment, and other learning resources
- Instructors can download figures from the book, the audio files, and a test bank

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Seventh Edition

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To family, friends, and music.

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Contents

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Preface

It's every musician's dream: to create and record your music and release it as an album or digital downloads so that others can enjoy it. This book will help you do that.

Recording is a highly skilled craft combining art and science. It requires technical knowledge as well as musical understanding and critical listening ability. By learning these skills, you can capture a musical performance and reproduce it with quality sound for the enjoyment and inspiration of others.

Your recordings will become carefully tailored creations of which you can be proud. They will be a legacy that can bring pleasure to many people for years to come.

This book is intended as a hands-on, practical guide for beginning recording engineers, producers, musicians—anyone who wants to make better music recordings by understanding recording equipment and techniques. I hope to prepare the reader for work in a home studio, a small professional studio, or an on-location recording session.

Practical Recording Techniques offers up-to-date information on the latest music recording technology, such as hard-disk and flash memory recorders, computer recording, tablet recording systems, loop-based recording, keyboard and digital workstations, MIDI, Web audio, realtime streaming, and online collaboration. But it also guides the beginner through the basics, showing how to make quality recordings with the new breed of inexpensive home-studio equipment.

The first chapter overviews the recording process to instill a system concept. Next, the basics of sound, studio acoustics, and signals are explained so that you'll know what you're doing when you adjust the controls on a piece of recording equipment or build a studio. Then advice is given on equipping a home studio for any style of recording, with examples of equipment manufacturers.

Studio setup is covered next, including cables and connections, choosing and using monitor speakers, and preventing hum.

Each piece of recording equipment is explained in detail, as well as the control-room techniques you'll use during actual sessions. Special attention is given to microphones (including detailed microphone techniques), EQ, effects, and signal processors.

Next is a detailed description of hardware mixing consoles, both analog and digital.

A chapter is devoted to the technology of digital audio. That is followed by a chapter on computer recording, which includes a section on audio for video. The text continues with chapters on DAW signal flow and DAW operation.

A special chapter explains how to judge recordings and improve them. The engineer must know not only how to use the equipment, but also how to tell good sound from bad.

After covering audio recording, the book offers a chapter on MIDI sequencing for musicians who want to record in that mode.

After all the equipment and procedures have been explained, a chapter on session procedures tells how to run a recording session. Following that, mastering and CD burning are described as the final stages in the process.

So far the book has focused on studio recording. The next two chapters explain on-location recording techniques for both popular and classical music.

The latest developments in recording are Web audio, realtime streaming, and online collaboration. These topics are covered in detail in the final chapter.

Finally, five appendices explain the decibel, suggest how to optimize your computer for multitrack recording, explain impedance, provide information on phantom power, and review legacy recording devices.

If you want more education in recording technology than this book provides, be sure to check out this book's companion website: www.routledge.com/cw/bartlett. It has links to several books, magazines, videos, audio equipment stores, and literature.

Listen Online In the book's companion website, audio samples demonstrate various topics explained in the book. Throughout the text, references to specific tracks on the website guide the reader to relevant audio demonstrations. These are highlighted in Listen Online Boxes like this one. Listen to Audio Clip 1 for an introduction. The companion website also includes articles with audio, suggested activities, and an extensive list of weblinks.

Based on my work as a professional recording engineer, *Practical Recording Techniques* is full of tips and shortcuts for making great-sounding recordings, whether in a professional studio, project studio, on location, or at home.

STARTING A CAREER IN RECORDING

After you read the material in this book and apply it in your own work, you might consider a career in recording engineering. This book's companion website lists some schools that offer recording classes. Some offer job placement upon graduation.

Currently there is more work for recording engineers in film, computer games, and TV than in music recording. One reason is that many more musicians are recording themselves at home.

Rather than trying to find a job in a music recording studio, you might start your own. The gear costs so little these days. After you acquire some high-quality equipment, offer to record local bands for free while you gain experience. You might specialize in one genre of music that matches your market and your recording skills.

You'll need to promote your studio. Post flyers in music stores. You can advertise your recording service for free on Craigslist. Create a website where you list your studio's musical instruments, recording equipment, and customers. Find potential customers at clubs or festivals, and give them demo CDs and business cards with your website's URL.

If you can't find much recording business, consider taking another job in a related industry—such as audio equipment design or sales—and record on weekends as a second job.

The usual way to join a commercial recording studio is to have an interview. Supply examples of your best work. Be willing to start as a gofer, and show that you can drum up business for the studio, perhaps bringing clients with you. Work your way up to operating recorders, placing mics, and copying/backing up files. Eventually, if you have personal and musical skills as well as technical skills, you can hope to sit at the console as a mixing engineer. Good luck!

MUSIC: WHY WE RECORD

As you learn about recording techniques for music, it's wise to remember that music is a wonderful reason for recording.

Music can be exalting, exciting, soothing, sensuous, and fulfilling. It's marvelous that recordings can preserve it. As a recording engineer or recording musician, it's to your advantage to better understand what music is all about.

Music starts as musical ideas or feelings in the mind and heart of its composer. Musical instruments are used to translate these ideas and feelings into sound waves. Somehow, the emotion contained in the music—the message—is coded in the vibrations of air molecules. Those sounds are converted to electricity and stored magnetically or optically. The composer's message manages to survive the trip through the mixing console and recorders; the signal is transferred to disc or computer files. Finally, the original sound waves are reproduced in the listening room, and miraculously the original emotion is reproduced in the listener as well.

Of course, not everyone reacts to a piece of music the same way, so the listener may not perceive the composer's intent. Still, it's amazing that anything as intangible as a thought or feeling can be conveyed by tiny magnetic patterns on a hard disk or by pits on a compact disc.

The point of music lies in what it's doing now, in the present. In other words, the meaning of "Doo wop she bop" is "Doo wop she bop." The meaning of an Am7 chord followed by an Fmaj7 chord is the experience of Am7 followed by Fmaj7.

INCREASING YOUR INVOLVEMENT IN MUSIC

Sometimes, to get involved in music, you must relax enough to lie back and listen. You have to feel unhurried, to be content to sit between your stereo speakers or wear headphones, and listen with undivided attention. Actively analyze or feel what the musicians are playing.

Music affects people much more when they are already feeling the emotion expressed in the song. For example, hearing a fast Irish reel when you're in a party mood, or hearing a piece by Debussy when you're feeling sensuous, is more moving because your feelings resonate with those in the music. When you're falling in love, any music that is meaningful to you is enhanced.

If you identify strongly with a particular song, that tells you something about yourself and your current mood. And the songs that other people identify with tell you something about them. You can understand individuals better by listening to their favorite music.

DIFFERENT WAYS OF LISTENING

There are so many levels on which to listen to music—so many ways to focus attention. Try this. Play one of your favorite records several times while listening for these different aspects:

- Overall mood and rhythm
- Lyrics
- Vocal technique
- Bass line
- Drum fills
- Sound quality
- Technical proficiency of musicians
- Musical arrangement or structure
- Reaction of one musician to another musician's playing
- Surprises versus predictable patterns

By listening to a piece of music from several perspectives, you'll get much more out of it than if you just hear it as background. There's a lot going on in any song that usually goes unnoticed. Sometimes when you play an old familiar record and listen to the lyrics for the first time, the whole meaning of the song changes for you.

Most people react to music on the basic level of mood and rhythmic motivation. But as a recording enthusiast, you hear much more detail because your focus demands sustained critical listening. The same is true of trained musicians focusing on the musical aspects of a performance.

It's all there for anyone to hear, but you must train yourself to hear selectively and to focus attention on a particular level of the multidimensional musical event. For example, instead of just feeling excited while listening to an impressive lead-guitar solo, listen to what the guitarist is actually playing. You may hear some amazing things.

Here's one secret of really involving yourself in recorded music: imagine yourself playing it! For example, if you're a bass player, listen to the bass line in a particular record, and imagine that you're playing the bass line. You'll hear the part as never before. Or respond to the music visually—see it as you do in the movie *Fantasia*.

Follow the melody line and see its shape. Hear where it reaches up, strains, and then relaxes. Hear how one note leads into the next. How does the musical expression change from moment to moment?

There are times when you can almost touch music: some music has a prickly texture (many transients, emphasized high frequencies); some music is soft and sinuous (sine-wave synthesizer notes, soaring vocal harmonies); and some music is airy and spacious (lots of reverberation).

WHY RECORD?

Recording is a service and a craft. Without it, people would be exposed to much less music. They would be limited to the occasional live concert or to their own live music, played once and forever gone.

With recordings, you can preserve a performance for thousands of listeners. You can hear an enormous variety of musical expressions whenever you want. Unlike a live concert, a record can be played over and over for analysis. Tapes or discs are also a way to achieve a sort of immortality. The Beatles may be gone, but their music lives on.

Records can even reveal your evolving consciousness as you grow and change. A computer audio file or CD stays the same physically, but you hear it differently over the years as your perception changes. Recordings are a constant against which you can measure change in yourself.

Be proud that you are contributing to the recording art—it is done in the service of music.

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CHAPTER 1

A Basic Overview of the Recording Process

Welcome to the brave new world of 21st-century recording! This book will show you an overview of current recording technology, help you choose the equipment that best suits your needs, and guide you in using it to create great recordings. And it will explain the technical jargon in plain English.

As a recording engineer, you are a key player. Your skills help artists realize their visions in **sound**. Your miking techniques capture the vibrancy of the performance, whether it's the shimmering overtones of a string quartet or the sonic assault of an electric blues band. Your "post" work in the studio—adding effects, tweaking **levels**, etc.—will take the raw material of the performance and shape and blend it into a polished musical statement. As you master the technology and become fluent with the audio tools at hand, you will produce exciting recordings that will delight your clients and give you a real sense of pride and achievement.

Be sure to practice what you learn in this book. There's no substitute for hands-on experience. You might offer to record a band's rehearsal for free while you experiment and master the gear. Be patient, let yourself make mistakes, and above all, listen to how the sound changes when you move a mic or tweak a knob.

CAREERS IN AUDIO

This book focuses on music recording, both in the studio and on-location. There are dozens of related **audio** careers, each with its own textbooks: live sound, film sound recording and post production, TV, radio, computer games, CD duplication, online distribution of audio, nature recording, newscasts, electronic news gathering, live music TV and radio shows, streaming concerts, documentaries, instructional videos, forensic audio, audio equipment and software design, studio design, museum audio, tour bus and airplane audio, sound insulation, concert hall design, sound-system design, foley, voiceover, commercials, and more.

TYPES OF RECORDING

Let's get started. Currently there are six main ways to **record** music:

- Live stereo recording
- Live-mix-to-2-track recording
- Multitrack recorder and mixer
- Digital multitracker (recorder-mixer)
- Computer digital audio workstation (DAW)
- MIDI sequencing

This chapter provides a brief overview of these methods, and later chapters explore them in depth.

Live Stereo Recording

This method uses a **stereo** mic technique to record with a stereo microphone or two microphones into a recorder. It is most commonly used to record an orchestra, symphonic band, pipe organ, small ensemble, quartet, or soloist. The microphones pick up the overall sound of the instruments and the concert hall acoustics from several feet away. You might use this minimalist technique to record a folk group, rock group, or acoustic jazz group in a good-sounding room.

Figure 1.1 shows the stages of this method—the links in the **recording chain**. From left to right:

1. The musical instruments or voices make sound waves.
2. The sound waves travel through the air and bounce or reflect off the walls, ceiling, and floor of the concert hall. These **reflections** add a pleasing sense of spaciousness.
3. The sound waves from the instruments and the room reach the microphones, which convert the sound into electrical **signals**.
4. The sound quality is greatly affected by mic technique (microphone choice and placement).

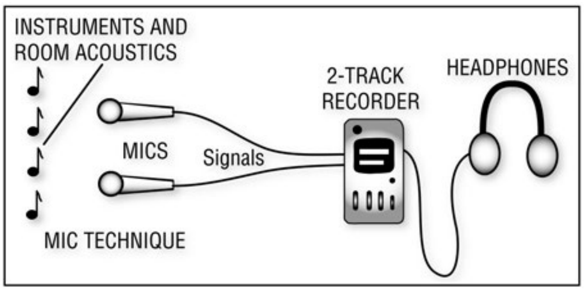

FIGURE 1.1
The recording chain for live stereo recording.

5. The signals from the microphones go to a 2-track recorder such as a handheld digital recorder or laptop. The signal changes to a pattern stored on a medium, such as magnetic patterns on a hard disk. During playback, the patterns on the medium are converted back into a signal. During recording, signals are stored along a **track**—a path or **channel** on the medium containing a recorded signal. A single medium can record one or more tracks. For example, a 2-track hard-disk recording stores two tracks on hard disk, such as the two different audio signals required for stereo recording.
6. To hear the signal you're recording, you need a **monitor** system: headphones or a stereo power amplifier and loudspeakers. You use the monitors to judge how well your mic technique is working.

The **speakers** or headphones convert the signal back into sound. This sound resembles that of the original instruments. Also, the acoustics of the listening room affect the sound reaching the listener.

Live-Mix-to-2-Track Recording

This method is used mainly for live broadcasts or recordings of PA mixes and some orchestral recordings. A PA is a public address or sound reinforcement system. Using a mixer, you set up a mix of several microphones and record the mixer's output signal on a 2-track recorder (flash memory recorder or computer hard drive). Each mic is close to its sound source. Figure 1.2 shows the recording chain for this method.

Multitrack Recorder and Mixer

For this method, you record with several mics into a mixer that is connected to a **multitrack** recorder of some sort: hard disk, solid state drive (SSD), flash memory card, computer software, or USB thumb drive. You record the signal of each microphone on its own track, then mix these recorded signals after the performance is done. You can also record different groups of instruments on each track. This method is often used in live, on-location recording (described in detail in Chapters 21 and 22). Most digital mixers used in live sound can make multitrack recordings easily.

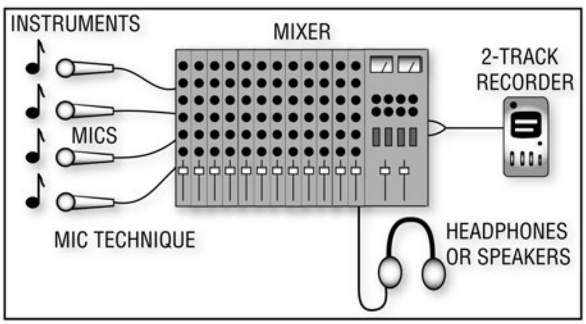

FIGURE 1.2
The recording chain for live-mix-to-2-track recording.

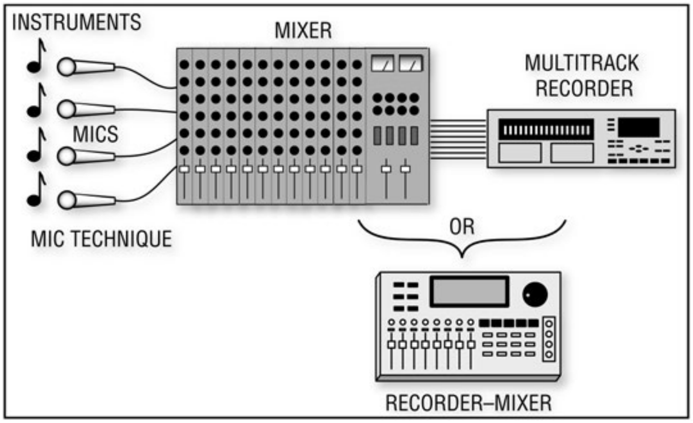

FIGURE 1.3
The recording chain for multitrack recording.

Figure 1.3 shows the stages in this method; they are:

1. Place microphones near the instruments.
2. Plug the mics into a mixing **console** (a big, sophisticated mixer). During multitrack recording, the mixing console amplifies the weak microphone signals up to the level needed by the recorder. The console also sends each microphone signal to the desired track.
3. Record the amplified mic signals on the multitrack recorder.

You can record more instruments later on unused tracks—a process called **overdubbing**. Wearing headphones, the performer listens to the recorded tracks and plays or sings along with them. You record the performance on an unused track.

After the recording is done, you play all the tracks through the mixing console to mix them with a pleasing **balance** (Figure 1.4). Another option is to mix the tracks with computer recording software. Here are the steps:

1. Play back the multitrack recording of the song several times, adjusting the track volumes and tone controls until the mix is just the way you want it. You can add **effects** to enhance the sound quality. Some examples are echo, reverberation, and compression (explained in Chapter 11). Effects are made by signal processors that connect to your mixer or by software applications (plug-ins) that are part of a recording program.
2. Record or export your final stereo mix onto your computer hard drive.

Digital Multitracker (Recorder-Mixer)

This is a multitrack recorder and a mixer combined in one portable chassis (Figure 1.5). It's relatively easy to use. The recording medium is a hard drive or a flash memory card.

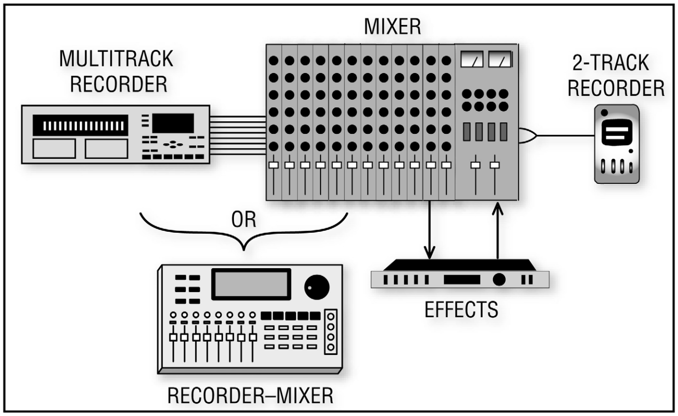

FIGURE 1.4
The recording chain for a multitrack mixdown.

FIGURE 1.5
TASCAM DP-03, an example of a recorder-mixer.
(Courtesy TASCAM.com)

Other names for this device include "**digital multitracker**," "personal digital studio," or "**portable studio**." Most **recorder-mixers** have built-in effects.

Computer Digital Audio Workstation

This low-cost system includes a computer, multitrack recording software, and a sound card or audio interface that gets audio into and out of your computer (Figure 1.6). The computer's hard drive or SSD records the audio.

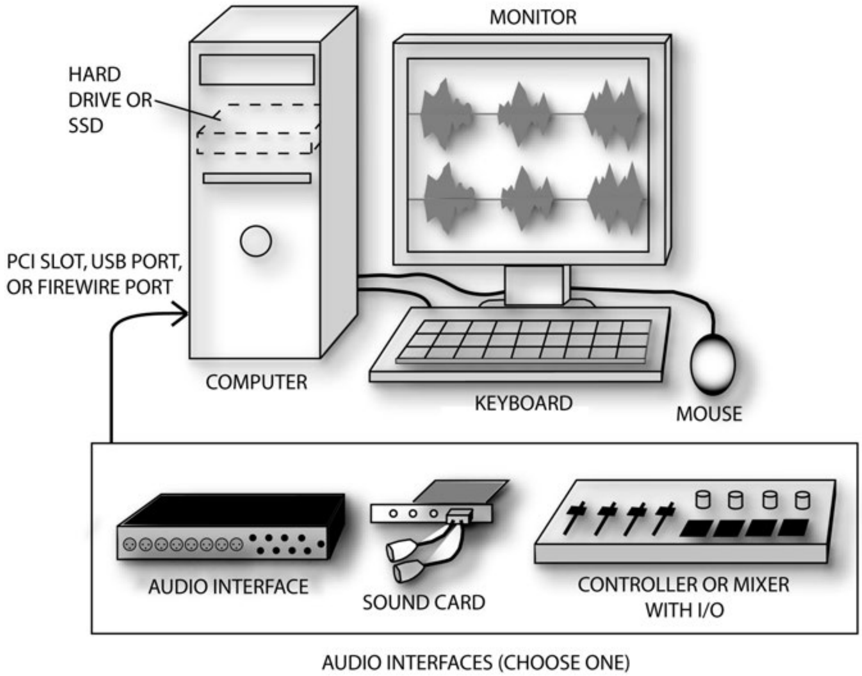

FIGURE 1.6
Computer with a choice of audio interface and recording/editing software.

Using the recording software, you perform these operations:

1. Record one or more tracks of music on the computer's hard drive or SSD.
2. **Punch in**: Record over segments of a musical part that were played incorrectly.
3. Overdub: Record new musical parts while listening to previously recorded tracks.
4. Edit the tracks to fix mistakes, delete unwanted material, or copy/move song sections.
5. Mix the tracks with a mouse or **controller** by adjusting controls that appear on your computer screen.

You might also assemble a song from samples or from loops. **Samples** are recordings of single notes of various instruments. **Loops** are repeating musical patterns. Chapter 18 covers this method in depth.

MIDI Sequencing

With this recording method, a musician performs on a **MIDI controller**, such as a piano-style keyboard or drum pads. The controller puts out a MIDI signal, a series of numbers that indicates which keys were pressed and when they were pressed. The MIDI signal is recorded and stored by a standalone **sequencer** or sequencer program in a computer. When you play back the MIDI sequence, it plays the tone generators in a synthesizer or sound module. The synthesizer can be hardware or software (as a "**soft synth**," also

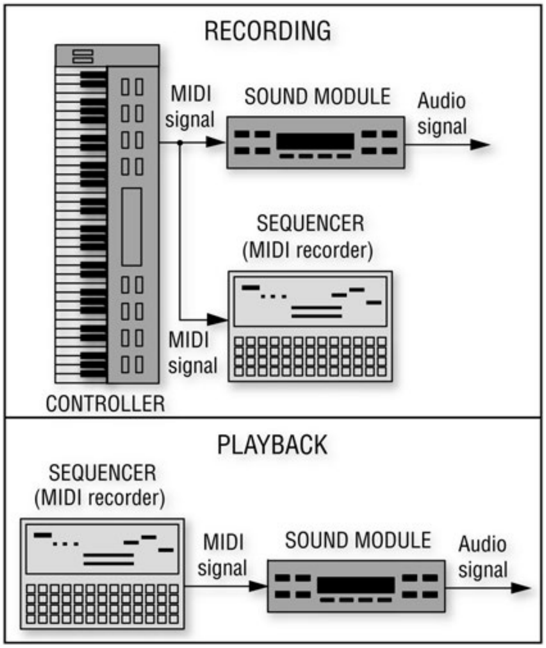

FIGURE 1.7
MIDI sequencing system.

called a "virtual instrument"). A MIDI sequencer also can play samples (**digital recordings** of musical notes played by real instruments). Like a player piano, MIDI sequencing records your performance gestures rather than audio. Figure 1.7 shows the process, and Chapter 18 gives a detailed explanation of MIDI.

MIDI/digital audio software lets you record MIDI sequences and **digital audio** onto a hard disk. First record a few tracks of MIDI sequences, then add audio tracks: lead vocal, sax solo, or whatever. All these elements stay synchronized.

PROS AND CONS OF EACH METHOD

Live stereo recording is simple, cheap, and fast. But it usually sounds too **muddy** with rock music, and you must adjust balances by moving musicians. It can work well with classical music, and sometimes with folk or acoustic jazz music.

Live-mix-to-2-track recording is fairly simple and quick. However, loud instruments might sound distant in the recording because their sound "leaks" into other instruments' mics. You can't correct mistakes in the mix or performance unless you rerecord. Also, the live sound of the band can make it hard to hear the monitored sound clearly.

Multitrack recording has many advantages. You can punch in (fix a musical mistake on a track by recording a new, correct part over the mistake). You can overdub (record one instrument at a time). This reduces leakage and gives a tighter sound. Also, you can postpone mixing decisions until after the performance. Then you can monitor the mix in quiet surroundings. This method is more complex and expensive than live-mix recording.

If you use a separate multitrack recorder and mixer, each component can be used independently. For example, you might do a PA job with just the mixer. Or, if you already have a mixer, all you need to buy is a recorder. This system also works well for multitrack recording of a live concert from the PA mixer's **insert** jacks. You need to connect cables between the mixer and recorder, and between the mixer and outboard effects units.

A recorder-mixer is easy to use because it is a single portable chassis that includes most of your studio equipment: recorder, mixer, effects, and often a CD burner. It doesn't require cables except for the mics, instruments, and monitor speakers. High-end units let you edit the music. They also have automated mixing: memory chips in the mixer remember your mixdown settings and reset the mixer accordingly the next time you play back the recording.

A computer DAW is inexpensive, powerful, and flexible. It lets you do sophisticated editing and automated mixing. Several plug-in (software) effects are included, and you can purchase and install other plug-ins. Recording software can be updated at little cost. As for drawbacks, computers can crash and can be difficult to set up and optimize for audio work.

MIDI sequencing lets you record musical parts by entering notes slowly or one at a time if you wish. After the sequencing is finished, you can edit notes to correct mistakes. You can even change the instrument sounds or the tempo. A huge variety of sounds are available in synthesizers, sound modules, and soft synths. However, you are limited to their sounds unless you use MIDI/digital audio software, which lets you add miked instruments to the mix.

Listen Online Audio samples, hosted on the companion website, are highlighted throughout the book in boxes like this one to demonstrate relevant topics. Audio Clip 1 introduces the clips, and Clip 2 demonstrates:

- Live stereo recording—orchestra
- Live stereo recording—rock group
- Live-mix-to-2-track recording—jazz group
- Multitrack recording—pop group
- MIDI sequencing—synthesizer funk

RECORDING THE MIXES

No matter which recording method you use, eventually you'll **mix** each song and record the mix as a stereo wave file on your hard drive. You can convert the wave file to **MP3**, **AAC**, or **WMA** format for uploading to the Web (explained in Chapter 23).

You might want to assemble an album of your recorded mixes. You remove noises and count-offs between songs, put the songs in the desired order, and put a few seconds of silence between songs. This is done with a computer and editing software (a DAW, Figure 1.6). The last step is to copy the album to a blank CD. There's your final product, ready to duplicate or replicate. As an alternative, you can create a wave file or MP3 file of each song for online distribution.

No matter what type of recording you do, each stage contributes to the sound quality of the finished recording. A bad-sounding recording can be caused by any weak link: an untrained engineer, low-quality microphones, ineffective mic placement or mixer settings, and so on. You can strive for quality recordings by optimizing each stage. This book will help you reach that goal.

PERSONNEL AT A RECORDING SESSION

Most professional recording sessions have a **producer**. This person is the musical director; he or she discusses with the musicians what songs should be recorded. They also critique performances and suggest ways to improve them. Musicians that lack a producer would produce themselves by making the musical decisions. Typically they rely on the **recording engineer** to guide the recording session. The engineer **mikes** the instruments, documents the session and operates the recording gear. The musicians, producer, and engineer collaborate on the mixes. Large **studios** might employ an **intern** who helps with miscellaneous chores as they learn the tricks of the trade while watching sessions. A **mastering engineer** (which might be the recording engineer) takes all the song mixes and puts them in the desired order to make an album. They also adjust the **tonal balance** of each song, the volume of each song, and the spacing between songs to make a uniform-sounding album.

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CHAPTER 2

Sound and Psychoacoustics

When you make a recording, you deal with at least two kinds of invisible energy: sound waves and electrical signals. For example, a microphone converts sound into a **signal**, a varying voltage that carries information. In our case, it's musical information.

This chapter covers some characteristics of sound and how we perceive it. These facts will help you work with room acoustics and will help you know how our perception of sound affects our use of **monitoring**, mic techniques, and effects. With this knowledge, you can make better recordings.

SOUND WAVE CREATION

To produce sound, most musical instruments vibrate against air molecules, which pick up the vibration and pass it along as **sound waves**. When these vibrations strike your ears, you hear sound. Sound waves are variations in air pressure.

To illustrate how sound waves are created, imagine a vibrating speaker cone in a guitar amp. When the cone moves out, it pushes the adjacent air molecules closer together. This forms a **compression**. When the cone moves in, it pulls the molecules farther apart, forming a **rarefaction**. As shown in Figure 2.1, the compressions have a higher pressure than normal atmospheric pressure; the rarefactions have a lower pressure than normal.

These disturbances pass from one molecule to the next in a springlike motion—each molecule vibrates back and forth to pass the wave along. The sound waves travel outward from the sound source at 1130 feet per second (344 meters per second), which is the **speed of sound** in air at room temperature.

At some receiving point, such as an ear or a microphone, the air pressure varies up and down as the disturbances pass by. Figure 2.2 is a graph showing how sound pressure varies with time, like a wave. The high point of the graph is called a **peak** (maximum compression); the low point is called a **trough** (maximum rarefaction). The horizontal center line of the graph is normal atmospheric pressure.

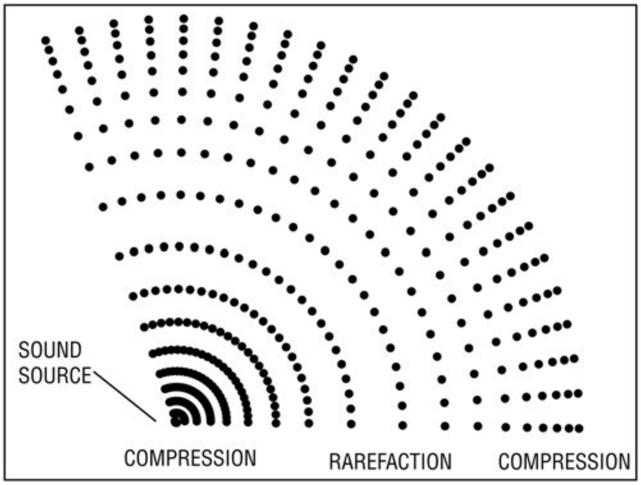

FIGURE 2.1
A sound wave.

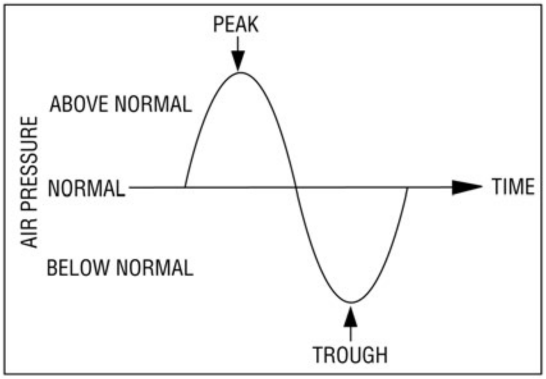

FIGURE 2.2
Sound pressure vs. time of one cycle of a sound wave.

Sound waves tend to spread out as they travel away from the source. The compressions and rarefactions move out as expanding spheres. As the spherical waves expand, the sound pressure spreads over a larger area, so the pressure becomes weaker with distance from the source. That means, the farther you are from a sound source, the quieter its sound is. Specifically, the sound pressure halves—drops 6 **decibels (dB)**—each time the distance from the source doubles. This phenomenon is called the **inverse square law**.

CHARACTERISTICS OF SOUND WAVES

Figure 2.3 shows three waves in succession. One complete vibration from normal to high to low pressure and back to the starting point is called one **cycle**. The time it takes to complete one cycle—from the peak of one wave to the next—is called the **period** of

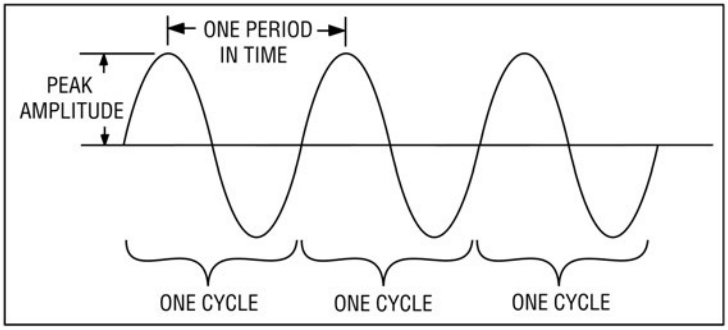

FIGURE 2.3
Three cycles of a wave.

the wave. One cycle is one period long. The formula for period is $p = 1/f$, where $p =$ the period in seconds and $f =$ the frequency in Hz (covered later).

Amplitude

The height of the wave is its **amplitude**. Loud sounds have high amplitudes (big pressure changes); quiet sounds have low amplitudes (small pressure changes).

Listen Online Play Audio Clip 3 on the companion website to hear an example of high and low amplitudes.

Loud, sustained sounds can damage parts of the inner ear, resulting in hearing loss and tinnitus (ringing in the ears). So it's very important to protect your hearing, which is the most valuable tool of the sound engineer. Wear ear plugs at loud concerts and while using loud machinery. If you have to yell over the sound of your studio monitor speakers, they are too loud and can damage your hearing over time.

Frequency

The sound source (in this case, the guitar-amp loudspeaker) vibrates back and forth many times a second. The number of cycles completed in one second is called **frequency**. The faster the speaker vibrates, the higher the frequency of the sound. Frequency is measured in **hertz** (**Hz**), which means cycles per second. One thousand hertz is called "one kilo-hertz," abbreviated kHz.

The higher the frequency, the higher the perceived pitch of the sound. Low-frequency tones have a low pitch (like low E on a bass, which is 41 Hz). High-frequency tones have a high pitch (like four octaves above middle C, or 4186 Hz).

Listen Online Audio Clip 4 on the companion website illustrates that the higher the frequency, the higher the perceived pitch of the sound.

Doubling the frequency raises the pitch one **octave**.

Most children can hear frequencies from 20 Hz to 20 kHz, and most adults with good hearing can hear up to 15 kHz or higher. Each musical instrument produces a range of frequencies, say, 41 Hz to 9 kHz for a string bass, or 196 Hz to 15 kHz for a violin.

An equation for frequency is f = c/w, where f = frequency in Hz, c = the speed of sound (1130 feet per second), and w = wavelength in feet (see below). In meters, it's f = 344/w.

Amplitude and frequency are independent from each other. Any frequency can occur at any amplitude.

Wavelength

When a sound wave travels through the air, the physical distance from one peak (compression) to the next is called a **wavelength** (refer to Figure 2.1). Low-pitched sounds have long wavelengths (several feet); high-pitched sounds have short wavelengths (a few inches or less). Wavelength is the speed of sound divided by frequency. So the wavelength of a 1000 Hz wave is 1.13 feet (0.344 m), 100 Hz is 11.3 feet (3.44 m), and 10 kHz is 1.35 inches (3.45 cm).

An equation for wavelength is w = c/f, where w = wavelength in feet, c = the speed of sound (1130 feet per second), and f = frequency in Hz. In meters it's w = 344/f.

Phase and Phase Shift

The **phase** of any point on the wave is its degree of progression in the cycle—the beginning, the peak, the trough, or anywhere between. Phase is measured in degrees, with 360 degrees being one complete cycle. The beginning of a wave is 0 degrees, the peak is 90 degrees (one-quarter cycle), and the end is 360 degrees. Figure 2.4 shows the phases of various points on the wave.

If there are two identical waves traveling together, but one is delayed with respect to the other, there is a **phase shift** between the two waves. The more delay, the more

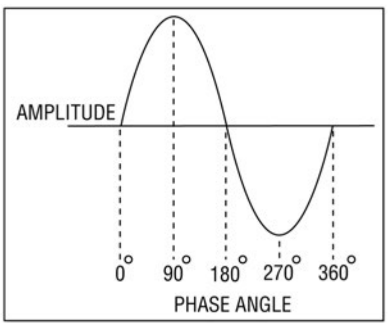

FIGURE 2.4
The phases of various points on a wave.

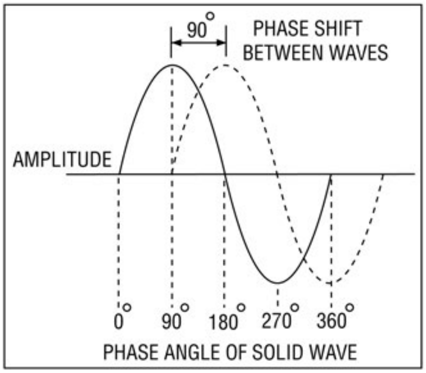

FIGURE 2.5
Two waves that are 90 degrees out of phase.

phase shift. Phase shift is measured in degrees. Figure 2.5 shows two waves separated by 90 degrees (one-quarter of a cycle) of phase shift. The dashed wave lags the solid wave by 90 degrees.

If you combine two identical sound waves, such as a sound and its reflection off a wall, the peaks of the two waves add together at certain points in the room. This doubles the sound pressure or amplitude, creating areas of louder sound at certain frequencies at those points.

Phase Interference

When there is a 180-degree phase shift between two identical waves, the peak of one wave coincides with the trough of another (Figure 2.6). If these two waves are combined, they cancel out. This phenomenon is called **phase cancellation** or **phase interference**.

Suppose you have a signal with a wide range of frequencies, such as the singing voice. If you delay this signal and combine it with the original undelayed signal, some frequencies will be 180 degrees out of phase and will cancel. This creates a hollow, filtered tone quality.

Here's an example of how this can happen. Suppose you're recording a singer/guitarist with one mic near the singer and another mic near the guitar. Both mics pick up the singer. The singer's mic is close to the mouth, and you hear it with no delay in the signal. The guitar mic is farther from the mouth, so its voice signal is delayed. When you mix the two mics, you often hear a colored tone quality caused by phase cancellations between the two mics.

Suppose you're recording a stage play with a mic on a short stand on the floor. The mic picks up the direct sound from the actors, but it also picks up delayed reflections off the floor. Direct and delayed sounds combine at the mic, causing phase cancellations. You hear it as a hollow, filtered sound that changes when the actor walks while talking.

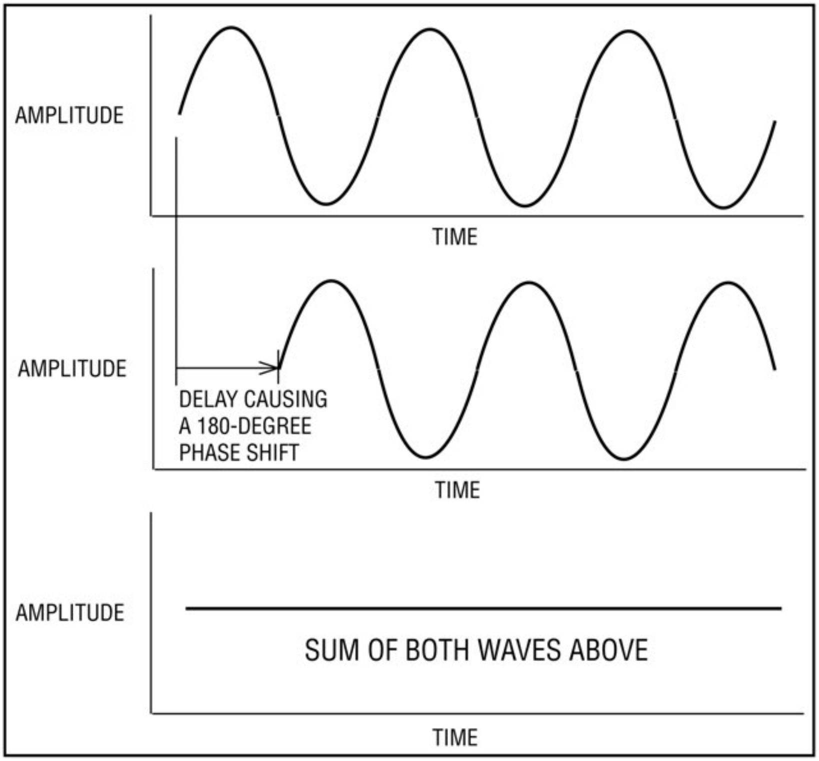

FIGURE 2.6
Phase interference: Adding two waves that are out of phase cancels the sound at that frequency.

The phase interference caused by the combination of a direct and delayed sound has an effect on the frequency response called **comb filtering**. There are peaks and dips in the frequency response graph that resemble the teeth of a comb. Here is the formula for the amplitude versus frequency of a direct sound mixed at equal levels with the same sound delayed:

> dB $= 20 \log | 2\cos(\pi ft) |$ where dB $=$ the amplitude in decibels, π (pi) in radians $=$ 180 degrees, f $=$ frequency, and t $=$ delay time in seconds.

Harmonics

The type of wave shown in Figure 2.2 is called a **sine wave**. It is a pure tone of a single frequency, like a signal from a tone generator. In contrast, most musical tones have a complex waveform, which has more than one frequency component. All sounds are combinations of sine waves of different frequencies and amplitudes. Figure 2.7 shows sine waves of three frequencies combined to form a **complex wave**.

The lowest frequency in a complex wave is called the **fundamental frequency**. It determines the pitch of the sound. Higher frequencies in the complex wave are called **overtones**

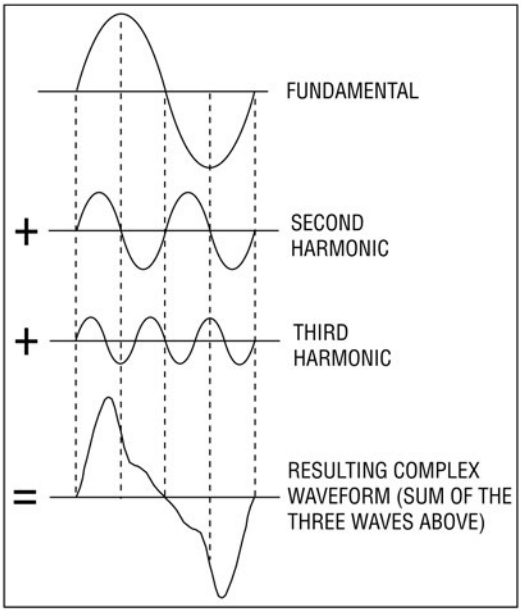

FIGURE 2.7
Adding fundamental and harmonic waveforms to form a complex waveform.

or **upper partials**. If the overtones are integral multiples of the fundamental frequency, they are called **harmonics**. For example, if the fundamental frequency is 200 Hz, the second harmonic is 400 Hz, and the third harmonic is 600 Hz.

Harmonics and their amplitudes partly determine the **tone quality** or **timbre** of a sound. They help you identify the sound as a drum, piano, organ, voice, and so on.

Listen Online Play Audio Clip 5 on the companion website to hear examples of harmonics.

In general, instruments with few or weak harmonics—such as a flute—tend to sound pure and smooth. Instruments with many or strong harmonics—such as a trumpet or distorted guitar—tend to sound bright and edgy.

As explained in Chapter 10, **equalization** can change the tonal balance of a recorded instrument by boosting or cutting its harmonics and fundamental frequencies. Boosting the fundamentals tends to make the sound warm, while cutting the fundamentals makes the sound thin; boosting the harmonics makes the sound bright, defined, or trebly, while cutting the harmonics makes the sound dark or muffled.

Playing an instrument louder usually increases its harmonics. So a piano played loudly sounds brighter than the same piano played softly.

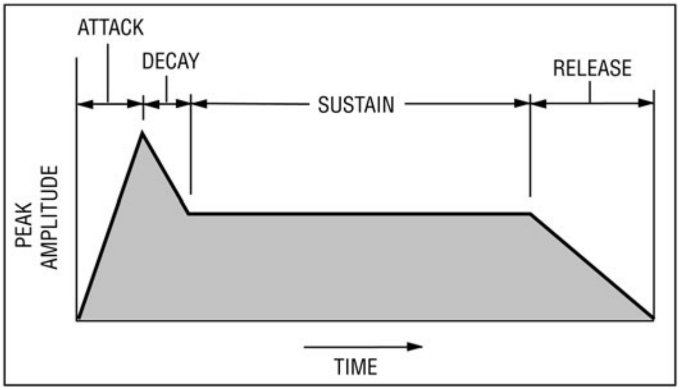

FIGURE 2.8
The four sections of the envelope of a note.

Noise (such as tape hiss) contains a wide band of frequencies and has an irregular, nonrepeating waveform. Some musical sounds with a hissy or noise-like character are a cymbal, snare drum, or a singer's "s" sounds (sibilance).

Envelope

Another characteristic that identifies a sound is its **envelope**. When a note sounds, it doesn't last forever—it rises in volume, lasts a short time, then falls back to silence. This rise and fall in volume of one note is called the note's envelope. The envelope connects the peaks of successive waves that make up a note. Each musical instrument has a different envelope.

Most envelopes have four sections: **attack**, **decay**, **sustain**, **and release** (see Figure 2.8). During the attack, a note rises from silence to its maximum volume. Then it decays from maximum to some midrange level. This middle level is the sustain portion. During release, the note falls from its sustain level back to silence.

Percussive sounds, such as drum hits, are so short that they have only a rapid attack and decay. Other sounds, such as organ or violin notes, last longer. They have slower attacks and longer sustains. Guitar plucks and cymbal crashes have quick attacks and slow releases. They hit hard then fade out slowly.

Listen Online Play Audio Clip 6 on the companion website, which demonstrates the envelope of a note.

You can shorten a guitar string's decay or ringing by lightly touching the string with the side of your hand. You can press a blanket against a kick drum head to dampen the decay and get a tighter sound. To "**damp**" means to add resistance to a vibrating object so that its vibrations die out more quickly after a note is sounded.

Harmonics usually change during a note's envelope. For example, if an instrument has a percussive attack—like a guitar pluck or tom hit—the harmonics are strongest at the attack, then become weaker during the decay.

BEHAVIOR OF SOUND IN ROOMS

Because most music is recorded in rooms, you need to understand how room surfaces affect sound.

Echoes

Musical instruments generate sound waves that travel outward in all directions. Some of the sound travels directly to your ears (or to a microphone) and is called **direct sound**. The rest strikes the walls, ceiling, floor, and furnishings of the recording room. At those surfaces, some of the sound is **absorbed—**it turns into heat due to frictional losses. Some is **transmitted** through the surface, and the rest is **reflected** back into the room.

Because sound waves take time to travel (moving at about 1 foot per millisecond), the reflected sound reaches you after the direct sound. The **reflection** repeats the original sound after a short delay. If the sound is delayed about 50 milliseconds or more, we call it an **echo** (Figure 2.9). In some concert halls, we hear single echoes; in small rooms, we

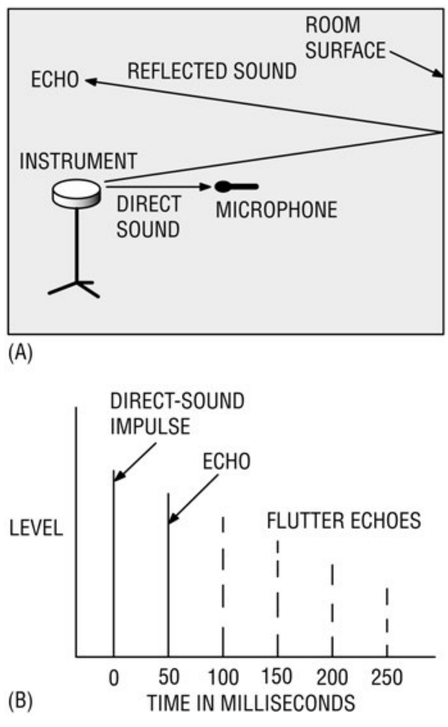

FIGURE 2.9
Echoes. (A) Echo formation. (B) Intensity vs. time of direct sound and its echoes.

often hear a short, rapid succession of echoes called **flutter echoes**. You can detect them by clapping your hands next to a wall. Flutter echoes happen when sound bounces back and forth between two parallel walls.

Reverberation

Sound reflects many times from all the surfaces in the room. These reflections sustain the sound of each note the musician plays. This persistence of sound in a room after the original sound has stopped is called **reverberation (reverb)**. For example, reverberation is the sound you hear just after you shout in an empty gymnasium. The sound of your shout stays in the room and gradually dies away (decays).

Listen Online Play Audio Clip 10 on the companion website to hear echoes and reverberation.

Reverb is hundreds of sound reflections that gradually get quieter. The reflections follow each other so rapidly that they merge into a single continuous sound. Eventually, the room surfaces completely absorb the reflections. The timing of the reflections is random, and the reflections increase in number as they decay. Figure 2.10 shows how reverberation develops in a room.

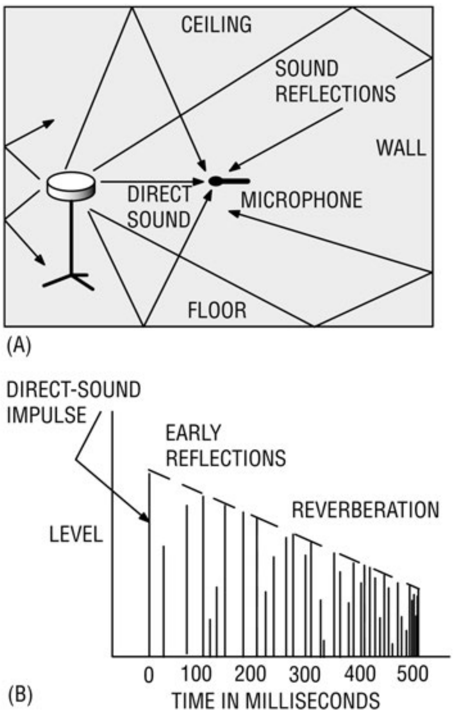

FIGURE 2.10
Reverberation. (A) Multiple sound reflections create reverberation. (B) Intensity vs. time of direct sound, early reflections, and reverberation.

Early reflections are sounds that arrive at the listener after reflecting a few times from the walls, ceiling, and floor. They arrive about 20 to 80 msec after the direct sound, and provide cues about the size of the room and the distance of the sound source.

Reverberation is a continuous fade-out of sound ("HELLO-O-O-o-o"), while an echo is a discrete repetition of a sound ("HELLO hello hello hello").

Reverberation time (RT60) is the time it takes for reverb to decay 60 dB. Too long a reverb time makes a recording sound distant, muddy, and washed out. That's why pop-music recordings are usually made in a fairly "**dead**" or nonreverberant studio, which has an RT60 of about 0.4 second or less. In contrast, classical music is recorded in "**live**" reverberant concert halls (RT60 about 1 to 3 seconds) because we want to hear reverb with classical music—it's part of the sound.

Reverberation comes to you from every direction because it is a pattern of many sound reflections off the walls, ceiling, and floor. Because we can tell where sounds come from, we can distinguish between the direct sound of an instrument coming to us from a single location and the reverberation coming to us from everywhere else. So we can ignore the reverb and concentrate on the sound source. In fact, we normally are not aware of reverberation.

But suppose you put a mic next to your ears, record an instrument in that room, and play back the recording. You'll hear a lot more reverb than you heard live. What's going on? The reverb you recorded and played back is not all around you. Instead, it's all up front between the speakers. So it's more audible; you can't discriminate against the reverb spatially. To reduce the amount of reverb in your recordings, you need to place mics close to instruments, and maybe add some sound-absorbing materials to the room.

Diffusion

When sound waves strike and reflect off a very bumpy or convex surface, they spread out or diffuse. This **diffusion** can be used to weaken sound reflections, say in a **RPG (Reflection Phase Grating)**—a calculated bumpy surface that diffuses sound in a studio **control room**. Sound waves also spread out when they travel through a small opening.

Diffraction

Diffraction is the disturbance of a sound field by an obstacle. For example, when sound waves strike the diaphragm of a microphone, certain frequencies are emphasized: those whose wavelength is similar to the diaphragm diameter. Low frequencies bend or diffract around obstacles easily as if they weren't there, while high frequencies are blocked by obstacles. As an example, a gobo or movable sound barrier in a studio blocks mid-to-high frequencies but does not block low frequencies well.

PSYCHOACOUSTICS

Psychoacoustics is the science of how we perceive sound. It relates sound measurements to sound perception. Let's look at several aspects of psychoacoustics that apply to sound recording.

We **localize** sounds; we can tell the direction a sound is coming from. The brain does this by analyzing the difference in sound appearing at each ear. Sound sources create amplitude differences, time differences, and spectral differences between the ears. The spectrum (amplitude vs. frequency) of sound at each outer ear is called the **HRTF, Head Related Transfer Function**. It varies with the angle of the sound source relative to the head. Stereo, surround, and binaural sound systems create these differences at the ears to help us localize various instruments and voices in a recording.

Masking is an important part of psychoacoustics. One sound can mask or hide another—making it less audible. The closer in frequency the two sounds are, the more masking occurs. For example, a bass guitar can cover up or mask certain frequencies in a kick drum. Cymbals can mask sibilant sounds in a vocal, making them harder to hear. As explained in Chapter 10, equalization and filtering can be used to reduce masking, making each instrument in a mix more distinct.

The **loudness** of a sound depends on its **sound pressure level (SPL).** A chart of SPL levels of various sounds is in Appendix A on the decibel. The quietest sound we can hear is called the **threshold of hearing**. It is 0 dB SPL at 1 kHz. Painfully loud sound is around 120 dB SPL and higher.

As discovered by Bell Labs researchers **Fletcher and Munson**, the perceived loudness of a sound also depends on its frequency. Low-pitched sounds (**bass**) need a much higher SPL than middle-pitched sounds to be perceived equally loud to the ear. And the lower in level an audio program is, the less bass and upper-midrange around 4 kHz we hear. As you'll see in Chapter 6 on monitoring, it's important to monitor at a moderate level—around 85 dB SPL. If you monitor a mix very loudly (say, at 100 dB SPL), it will sound like it has less bass and treble when heard at a more normal level, such as 85 dB SPL.

The **pitch** of a sound depends on its frequency in Hz. The timbre or tone quality of a sound depends on its harmonic content and envelope.

The audibility of distortion depends on how long the distortion lasts. Even-order distortion tends to sound more pleasant or euphonic than odd-order distortion. Generally, 1% to 3% total harmonic distortion is just perceivable.

Our ears tend to fuse two identical sounds into one when the delay between those sounds is less than about 20 milliseconds.

The **Haas effect** or **Precedence effect** states: When we hear two identical sounds coming from sound sources in different locations, we tend to hear a single sound that we localize at the source that produces the earliest-arriving sound. Usually that sound is from a loudspeaker that is close to us. For example, suppose we are sitting in a theater. We hear the actors' live voices coming from the stage, and we also hear their amplified voices through a nearby loudspeaker. Their voices will appear to come from the loudspeaker rather than from the stage. If we delay the signal going to the loudspeaker by an amount that exceeds the travel time of sound from the stage to our ears, we will localize the voices as coming from the stage, and we won't seem to hear the loudspeaker.

Critical frequency bands are about 1/3 octave apart. Changes in frequency response (peaks and dips) that are narrower than a critical band are not very audible. So we use 1/3-octave graphic equalizers to tune the frequency response of loudspeakers.

The **placebo effect** is the perception of a change even though nothing has measurably changed. For example, we might hear an improvement in sound quality when we substitute an expensive amplifier for a cheap one. But if they measure the same, that "improvement" was in our expectations only. A **blind A-B test** can reveal whether any change has an audible effect. You switch between component A and component B playing music at matched levels, either quickly or slowly, without knowing which component is A and which is B. Then you can hear if they actually sound different without fooling yourself.

Another example: If a recording client asks you to turn up an EQ knob slightly and you pretend to do it, the client might claim to hear a change for the better.

Adaptation is accepting as normal a strange situation that we've been exposed to for a long time. Suppose your monitor speakers exaggerate the bass. If you do mixes while listening to them for a while, that bass boost will eventually sound normal and you won't notice it so much. Using those speakers, you will probably create mixes that sound "thin" or weak in the bass when heard by another listener who has more accurate monitors. That's why it's important to have your mixes evaluated by a second pair of ears, and with different monitor speakers.

We hear a sound in **context** with other sounds, and those other sounds affect our perception of whatever sound we are focusing on. For example, a drum hit with a silent background has more impact than the same drum hit as part of a band playing. A bass guitar might sound bright and edgy when heard by itself, but sounds more mellow or muffled when masked by all the other instruments in a mix.

There is much more to psychoacoustics, but that is a subject in itself.

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CHAPTER 3

Studio Acoustics

Recording studios that are judged to sound good are free from echoes, have minimal reverb with a fairly uniform RT60 with frequency, have minimal standing waves (bass resonances), and are well insulated from noise that occurs outside the studio. In this chapter we'll learn how to control the studio acoustics to create such a space.

HOW TO TAME ECHOES AND REVERB

Echoes and reverb can make your recordings sound mushy and distant. There are two ways to prevent these problems: with recording techniques and with acoustic treatment.

Controlling Room Problems with Recording Techniques

Sometimes you can make **clean** recordings in an ordinary room, such as a club, living room, or basement, if you follow these suggestions:

- Mike close. Place each mic 1 to 6 inches from each instrument or voice. Then the mics will pick up more of the instruments and less of the room reflections. You might want to use **mini mics**, which attach directly to instruments.
- Use **directional mics**—cardioid, supercardioid, or hypercardioid—which reject room acoustics.
- Record bass guitar and synth directly with a guitar cord or a direct box. By eliminating the microphone, you remove room acoustics from the recording. To get a good sound when recording an electric guitar directly, record off the effects boxes or use a guitar-amp simulator.

See also the "Leakage" section in Chapter 8 for tips on reducing leakage.

Controlling Room Problems with Acoustic Treatments

When should you add acoustic treatments?

- You clap your hands next to a wall and you hear flutter echoes (a fluttering sound). These are caused by sounds bouncing back and forth between hard parallel walls.

- Your studio is a very live environment, such as a garage or concrete-block basement, so you hear too much room reverberation.
- Your studio is very small.
- Bass-guitar amps and monitor speakers sound boomy.
- You want to mike several feet away without picking up excess room reverb.
- You hear a lot of **leakage** in the mic signals. Examples of leakage are drums picked up by the guitar mic or electric guitar picked up by the cymbal mics.

If those conditions apply, upgrade the acoustics of your studio. Here are some suggestions.

Reverb and echoes are caused by sound reflections off room surfaces. So any surface that is highly sound absorbent helps to reduce those problems. The acoustic treatment should absorb all frequencies well. For example, carpet on the walls absorbs only the high frequencies. So the low frequencies still reflect around the room, creating a boomy or muddy sound quality. We also need to absorb mid and low frequencies, and certain structures can achieve that.

To absorb high frequencies, use porous materials such as convoluted (bumpy) acoustic foam. Foam that is not coated with flame-retardant treatment can be flammable. Natfire.com or flamestop.com offer treatments. Nail or glue the acoustic foam to the walls, or mount it on frames. Thick foam works better than thin. Four-inch foam on the wall absorbs frequencies from about 200 to 800 Hz and up, depending on the angle at which sound strikes the foam. Leave some space between the foam panels (Figure 3.1); this

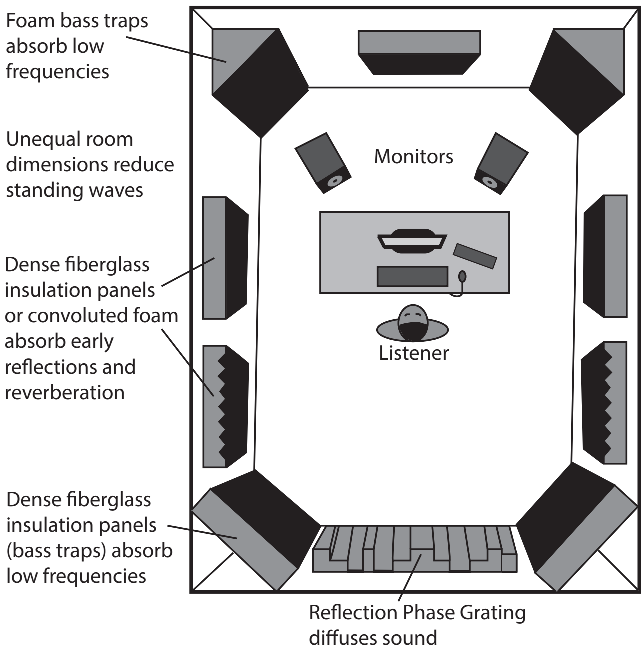

FIGURE 3.1
Acoustic treatments.

helps to diffuse or spread out the sound in the room. Don't overdo the foam padding. A stuffed, dead room is uncomfortable to play in. Keep some reflections because they add "air" and liveliness to the sound.

Other high-frequency absorbers are sleeping bags, moving blankets, carpeting, curtains, and dense fiberglass insulation covered with muslin or burlap. If possible, space these materials several inches from the wall. The spacing helps them absorb mid-bass frequencies. A wide-range absorber is 4-inch-thick or 2-inch-thick pressed fiberglass insulation board covered in muslin or burlap. Owens-Corning type 705 or Johns Manville 817 work well.

Add absorbers a few at a time, until your recordings sound as dead as you wish—typically covering 50–60% of the surfaces. Slide a mirror at eye height along the side walls, and place absorbent panels where you see mirror images of the monitors from the mix position. Put an absorber on the wall behind the monitors, and also hang an absorbent panel centered halfway between the mix position and monitors, suspended near the ceiling on hooks and wire.

Another absorber is a stand-mounted acoustic panel placed near a microphone. Examples are the sE Reflexion Filter, the Auralex MudGuard, and the RealTraps Portable Vocal Booth.

To absorb low frequencies, you can make **bass traps**. Here are three types:

1. Resonant tube trap: Take a 35- to 55-gallon rubber trashcan, stuff it with fiberglass insulation (wear a dust mask and gloves), and cover the open end with muslin or canvas. This tube trap absorbs frequencies near 1130/(2H), where H is the height of the trashcan in feet. For example, a 3-foot-high trashcan absorbs 188 Hz. Placement is not critical.
2. Frictional tube trap: Make a 2-foot-diameter canvas bag, 8 feet tall, and fill it with fiberglass insulation. Hang one bag a few feet out from each room corner. The distance should be 1130/(4f), where f is the frequency in Hz that you want to absorb. For example, to absorb 80 Hz, hang the bag 3.5 feet out from a room corner. (Thanks to David Moulton for this idea.)
3. Insulation panel: Get 8 pieces of rigid fiberglass insulation, 2 × 4 feet and 4 inches thick, from an insulation supplier or a studio acoustical supplier. Owens-Corning type 705 or Johns Manville 817 work well. Cover each piece with muslin or burlap to contain the fibers. Place a piece diagonally across each room corner with the 2-foot edge touching the floor (Figure 3.1). Stack two panels to make them 8 feet high. (Thanks to Ethan Winer for this idea.)

There are other ways to absorb bass. Wood paneling works well. It also helps to open closet doors and place couches and books a few inches from the walls. In a basement studio, nail acoustic tile to the ceiling joists, with fiberglass insulation in the air space between the tiles and ceiling.

You may not need any bass traps if you don't put any bass into the room. For example, don't turn up the bass guitar amp—just record the bass direct and have the musicians wear headphones to hear the bass.

Controlling Standing Waves

Let's look at another acoustic problem: **standing waves**. If you play an amplified bass guitar through a speaker in a room and do a bass run up the scale, you may hear some notes that stand out in the room. The room is resonating at those frequencies. These resonance frequencies, or **room modes**, are strongest below 300 Hz. They occur in patterns in the room called "standing waves." Room modes can give a tubby or boomy **coloration** to musical instruments and monitor speakers.

Standing waves are created when two waves of the same frequency move between room surfaces in opposite directions, and interfere repeatedly. The standing wave patterns occur in stationary **nodes** of low pressure and **antinodes** of high pressure.

The lowest frequency of a standing wave is $f = c/(2d)$, where f = frequency in Hz, c = the speed of sound (1130 feet per second), and d is a room dimension in feet. Other standing wave frequencies occur at 2f, 3f, 4f, 5f, and so on. Outside the U.S., $c = 344$ meters and d is in meters.

Room resonances are worst in a cubical room. They are less of a problem if the room's length, width, and height are not multiples of each other. Table 3.1 shows several room dimensions in feet that reduce boomy-sounding standing waves.

For example, if the room width is 9.1 feet and the ceiling is 8 feet high, the length should be 11.1 feet for best reduction of boominess.

Table 3.1 Room Dimensions in Feet to Reduce Standing Waves

Height	Width	Length
8	9.1	11.1
8	9.4	11.8
8	10.1	11.3
8	10.2	12.3
8	11.6	16.8
8	11.8	13.6
8	12.8	18.6
8	13.0	21.0
10	11.4	13.9
10	11.7	14.7
10	12.6	14.1
10	12.8	15.4
10	14.5	21.0
10	14.7	17.0
10	16.0	23.3
10	16.2	26.2

Try to record in a large room because the room resonance frequencies are likely to be below the musical range. Use bass traps to absorb room resonances. Contrary to popular opinion, nonparallel walls don't prevent standing waves.

MAKING A QUIETER STUDIO

You don't want to record any noises coming from outside the studio. These noises are reduced not by acoustic treatments, but by **sound insulation** (specific wall constructions and sealed openings). A similar term is **soundproofing**. The following tips will help prevent outside noises from getting into your recordings:

- Consider having the studio in a basement, because the earth blocks noises from the outside. The furnace or air conditioning might be a problem, however.
- Turn off appliances, air conditioning, and telephones while recording.
- Schedule your recordings when road and air traffic is light, or pause for ambulances and airplanes to pass.
- Close windows. Consider covering them with thick plywood.
- Close doors and seal with towels.
- Remove small objects that can rattle or buzz.
- Weather-strip doors all around, including underneath.
- Replace hollow doors with solid doors.
- Block openings in the room with thick plywood and caulking.
- Put several layers of plywood and carpet on the floor above the studio, and put insulation in the air space between the studio ceiling and the floor above.
- Place microphones close to instruments and use directional microphones. This won't reduce noise in the studio, but it will reduce noise picked up by the microphones.

When building a new studio, you might want to make the walls out of plastered concrete block because massive walls reduce sound transmission, or make the walls out of gypsum board and staggered studs. Nail gypsum board to 2 × 4 staggered studs on 2 × 6 footers as shown in Figure 3.2. Staggering the studs prevents sound transmission through the studs. Fill the air space between walls with insulation.

The ideal home-recording room for pop music is a large, well-sealed room with optimum dimensions. This room is in a quiet neighborhood. It should have some soft surfaces

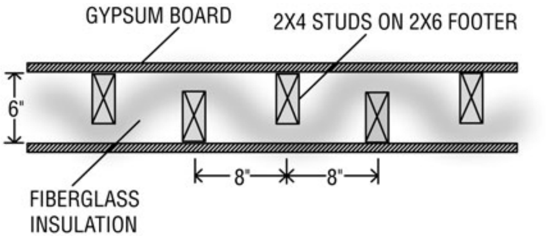

FIGURE 3.2
Staggered-stud construction reduces noise transmission.

(carpet, acoustic-tile ceiling, drapes, couches), and some hard vibrating surfaces (wood paneling or gypsum board walls on studs).

Your home studio may not need acoustic treatment or sound insulation. Do some trial recordings to find out. But if your room could stand some improvement, the previous suggestions should point you in the right direction.

For better results and a more professional appearance, consider buying some acoustic treatments from the following companies: www.atsacoustics.com, http://realtraps.com, www.acousticsciences.com, https://acousticalsolutions.com, http://primacoustic.com, www.auralex.com, http://acousticsfirst.com, http://wallmate.net, www.gikacoustics.com, and http://rpginc.com. Atsacoustics.com has a free online room acoustics analysis, auralex.com and atsacoustics.com offer low-cost absorbers, and primacoustic.com offers acoustic "London Kits" for rooms of various sizes.

For more information on acoustics, I recommend Ethan Winer's forum at Musicplayer.com: http://forums.musicplayer.com/ubbthreads.php/forums/24/1/Ethan_Winer_The_Audio_Expert.

CHAPTER 4

Signal Characteristics of Audio Devices

A **signal** is a varying parameter that represents information. For example, an audio signal is usually a varying voltage in the frequency range of 20 Hz to 20 kHz that represents speech, music, **sound effects**, hum, or noise.

When a microphone converts sound to electricity, this electricity is called the signal. It has the same frequencies and the same amplitude changes as the incoming sound wave.

When this signal passes through an audio device, the device may alter the signal. It might change the level of some frequencies, or add unwanted sounds that are not in the original signal. Let's look at some of these effects.

FREQUENCY RESPONSE

Suppose you have an audio device—a mic, mixer, effects unit, recorder, or speaker. You send a musical signal through the device. Usually the music contains some high and some low frequencies.

The device might respond differently to different frequencies. It might amplify the low notes and weaken the high notes. You can graph how the device responds to different frequencies by plotting its output level versus frequency. This graph is called a **frequency response** (Figure 4.1). The level in the graph is measured in **dB**, while frequency is measured in Hz. Generally, 1 dB is the smallest change in loudness that we can hear.

Suppose the level is the same at all frequencies. Then the graph forms a horizontal straight line and is called a "**flat frequency response**" (Figure 4.2). All the frequencies are reproduced at an equal level. In other words, the device passes all the frequencies without changing their relative levels. You get out the same amount of bass and treble that went in. A flat response does not affect the tonal balance of the incoming sound.

Many audio devices do not have a flat response across the audio band from 20 to 20,000 Hz. They have a limited range of frequencies that can be reproduced at an equal level (within a tolerance, such as ±3 dB). In Figure 4.1, the frequency response shown by the solid line

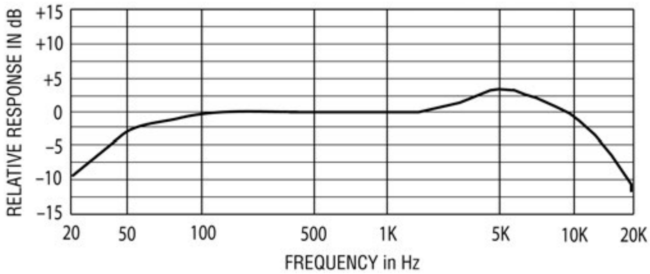

FIGURE 4.1
An example of a frequency response.

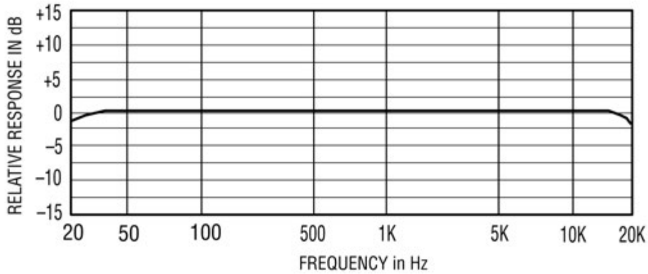

FIGURE 4.2
Flat frequency response.

is 50 to 12,000 Hz ±3 dB. That means the audio device passes all frequencies from 50 to 12,000 Hz at a nearly equal level—within 3 dB. It reproduces low sounds and high sounds equally well. The response is down 3 dB at 50 and 12,000 Hz, and is up 3 dB at 5000 Hz.

Usually, the more extended or "wide" the frequency range is, the more natural and real the recording sounds. A wide, flat response gives accurate reproduction. A frequency response of 200 to 8000 Hz (±3 dB) is narrow (poor fidelity); 80 to 12,000 Hz is wider (better fidelity), and 20 to 20,000 Hz is widest (best fidelity).

Listen Online Play Audio Clip 7 on the companion website to hear frequency response—the range of frequencies that a piece of equipment can reproduce within an equal level.

Also, the flatter the frequency response, the greater the fidelity or accuracy. A response deviation of ±3 dB is good, ±2 dB is better, and ±1 dB is excellent. There are exceptions to this statement, which we'll look at in Chapter 10 on equalizers.

When you turn a bass or treble knob on your guitar amp, mixer EQ, or stereo, you're changing the frequency response. If you turn up the bass, the low frequencies rise in level. If you turn up the treble, the high frequencies are emphasized. The ear interprets these effects as changes in tone quality—warmer, brighter, thinner, duller, and so on.

Figure 4.1 shows a nonflat frequency response. Toward the right side of this line, the response at high frequencies "**rolls off**" or declines. This shows that the upper harmonics are weak. The result is a dull sound. Toward the left side, the response at low frequencies rolls off. This means the fundamentals are weakened and the result is a thin sound.

The frequency response of an audio device might be made nonflat on purpose. For example, you might cut low frequencies with an equalizer to reduce breath pops from a microphone. Also, a microphone may sound best with a nonflat response, such as boosted high frequencies that add **presence** and sizzle.

POLARITY

Polarity refers to the positive or negative direction of an electrical, acoustical, or magnetic force. Two identical signals in opposite polarity are 180 degrees out of phase with each other at all frequencies. The waveform of one signal is inverted compared to the other: peaks become troughs, and vice versa. Polarity is not the same as phase or phase shift, which refers to a delay between two identical signals.

NOISE

Noise is another characteristic of audio signals. Every audio component produces a little noise—a rushing sound like wind in trees. Noise in a recording is undesirable unless it's part of the music.

You can make noise less audible by keeping the signal level in a device relatively high. If the level is low, you have to turn up the listening volume in order to hear the signal well. Turning up the volume of the signal also turns up the volume of the noise, so you hear noise along with the signal. But if the signal level is high, you don't have to turn up the listening level as much. Then the noise remains in the background.

DISTORTION

If you turn up the signal level too high, the signal distorts and you hear a gritty, grainy sound or clicks. This type of **distortion** is sometimes called "**clipping**" because the peaks of the signal are clipped off so they are flattened. To hear distortion, simply record a signal at a very high recording level (with the meters going well into the red area) and play it back. Digital recorders also produce "quantization" distortion at very low signal levels.

Listen Online Play Audio Clip 9 on the companion website to hear examples of distortion.

OPTIMUM SIGNAL LEVEL

You want the signal level high enough to cover up the noise but low enough to avoid distortion. Every audio component works best at a certain optimum signal level, and this is usually indicated on a level meter built into the device. This is discussed in more detail in Appendix A on the decibel.

Figure 4.3 shows the range of signal levels in an audio device. At the bottom is the **noise floor** of the device—the level of noise it produces with no signal. At the top is the **distortion level**—the point at which the signal distorts and sounds grungy. In between is a range in which the signal sounds clean. The idea is to maintain the signal around normal operation level on the average. Generally you want the signal level to be as high as possible without clipping or distorting.

With digital recorders (such as computer recording software), **0 dBFS** (0 decibels full scale) on the meter is the maximum undistorted signal. In that case, an average level of –20 dBFS and a peak level of –6 dBFS is the normal operation level. In a digital audio file that is used to master a CD, an average level of –14 dBFS is a good goal to aim for. There's more on digital signal levels in Chapter 13, Digital Audio.

SIGNAL-TO-NOISE RATIO

The level difference in decibels between the signal level and the noise floor is called the "**signal-to-noise ratio**" or **S/N** (see Figure 4.3). The higher the S/N, the cleaner the sound. An S/N of 60 dB is fair, 70 dB is good, and 80 dB or greater is excellent.

Listen Online Play Audio Clip 8 on the companion website to hear examples of noise and signal-to-noise ratios.

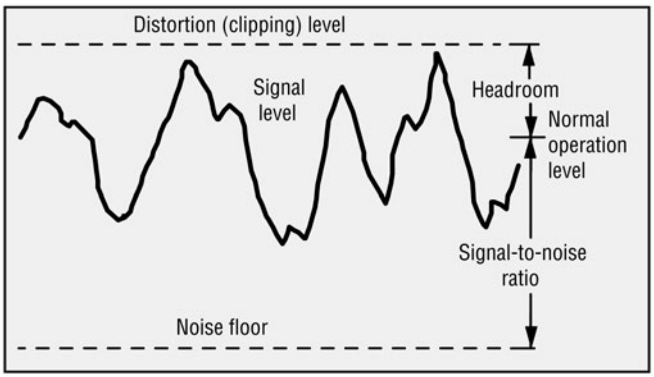

FIGURE 4.3
The range of signal levels in an audio device.

To illustrate S/N, imagine a person yelling a message over the sound of a train. The message being yelled is the signal; the noise is the train. The louder the message, or the quieter the train, the greater the S/N. And the greater the S/N, the clearer the message.

HEADROOM

The level difference in dB between the normal signal level and the distortion level is called "**headroom**" (see Figure 4.3). The greater the headroom, the greater the signal level the device can pass without running into distortion. If an audio device has a lot of headroom, it can pass high-level peaks without clipping them.

You want to set your mixer controls so that the signal has some headroom, is well above the noise floor, and is below distortion. Tips on doing that are in Chapter 13, Digital Audio, under the heading "Digital Recording Level." Also see Chapter 16, DAW Operation, under the headings "Set Recording Levels" and "Set Levels." These procedures are called "**gain staging**."

With these settings, the signal levels in the mixer should be just about right, with no audible noise or distortion. And the mixer should have enough headroom so that loud peaks won't distort.

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CHAPTER 5

Equipping Your Studio

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.

In this chapter we'll examine:

- Equipment
- Cables and connectors
- How to hook up your studio gear
- Preventing hum and radio frequency interference

What is the bare-bones equipment you need to crank out decent recordings? How much does it cost? Thanks to the new breed of affordable equipment, you can put together a complete home recording studio for as little as $1200. 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 devices or effects plug-ins.

Recording Device

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