Roland®

Basics of Mixing

A Beginner's Guide to the Art of Mixing

Roland

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A Beginner's Guide to the Art of Mixing

Written and illustrated by Robby Berman

1—Introduction

Why Does This Book Exist?

As a musician, producer or sound person, you can't get too far in this world without bumping into a central tool of the studio and stage: the mixer. To someone who's unfamiliar with how such a thing works, a mixer can seem impossibly complex. In fact, it's not. Any mixer is actually fairly simple to use, and is packed with helpful tools and creative possibilities.

The difference between bluffing your way around and really mastering a mixer is an understanding of the mixer's basic underlying principles. This book is full of simple explanations of how a typical mixer works. It's a companion to your mixer's documentation, which describes your mixer's own unique way of doing things. Using the two books together, you'll have your mixer mastered in no time.

About this Book

Special Symbols

Throughout "Basics of Mixing," you'll encounter a few special symbols in the right-hand margins. These are intended to draw your attention to something important. You'll see:

*   bricks whenever you're presented a basic concept that helps build the foundation of your understanding.

*   a light bulb whenever there's a good idea, a tip.

*   a push pin that signifies something you should note.

*   an exclamation point that means you're being told something you really need to know to avoid hurting your mix, your equipment or your ears.
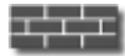
Discussions

You'll encounter some sections in the book that are offset by lines above and below, like this one. These are self-contained discussions about something that relates to the main text.

You'll find a glossary of common mixer terms in the back of "Basics of Mixing." You can use the glossary to refresh your memory as you mix.

How to Read this Book

"Basics of Mixing" should be read in order, since it builds your understanding of a mixer step-by-step. If you skip around, you may not have the basic knowledge required to comprehend what you're reading. Since the book is fairly short, reading it from top to bottom shouldn't be too hard to do.

Success in mixing depends on your thinking and doing things in a consistent, logical order, even if it takes a little extra time to do so. This will help you avoid confusion, and will save you time in the end.

Some Comforting Thoughts Before We Start

Here are two comforting thoughts for you if you're at all concerned that you'll have trouble understanding your mixer:

1. No matter how big a mixer is, or how complex it looks, it's basically made up of just a few things repeated over and over. This means that there's a lot less to learn than you might think. Consider this: if you have a 24-channel mixer (we'll explain what channels are later on), all you have to do is learn how one channel works. Your understanding of the one channel will apply to every one that the board has. Learn only one channel and you'll immediately know how to operate most of the mixer's controls.

If you understand one channel…   …you understand them all.
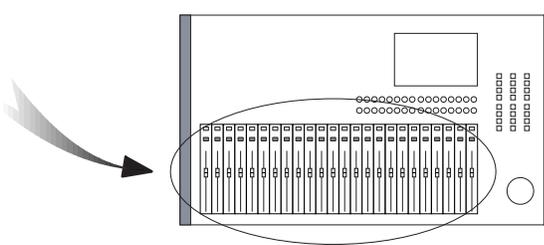
2. There's the normal way of using a mixer, and then there's your way. As you come to understand how a mixer works, you'll see it's an amazingly flexible tool that makes recording and performing easier and more fun. No matter how you like to work—or the demands of the music you're working on, or the people you're working with—your mixer's likely to be able to keep up with every creative whim. Digital mixers, especially, allow you to connect, or "patch," anything to anything else, and to do as many things to your sounds as you can dream up.

2—Overview: What Can a Mixer Do?

A Mixer is a Tool for Shaping Audio

A mixer is a tool for shaping sound, or, more specifically, audio. By "audio," we mean a stream of sound, such as a musician's live or recorded performance, a rapper's rap, or music played by a turntable or CD. A mixer does three basic things:

- It receives audio from some external source.
- It allows you to change the audio by adjusting its volume and a variety of other characteristics, and lets you run it through internal or external effects processors.
- It sends the audio to some other device.
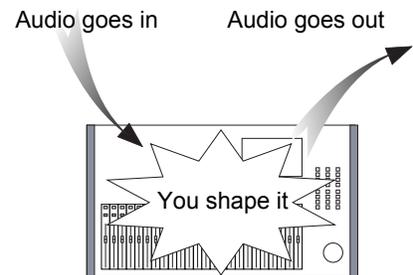
Remember our first comforting thought: a mixer consists the same thing over and over. This means that it can actually receive and change many audio streams before sending them out as separate audio streams, or mixed together.

When you combine two or more audio streams together, you're creating a "mix."

A Most Basic Definition: Mix

Let's make sure we're really clear about what the word "mix" means. When working with sound, it can be used as either a noun or a verb:

- As a noun, a mix is a mono or stereo stream of audio that's actually a combination of two or more other audio streams. This combination is created by directing these streams down pathways called "busses," which we'll describe fully in Chapter 3. When recording or doing live sound, a mix is the ultimate destination for all of the audio with which you're working—in recording, the mix goes to your mixdown recorder; in live sound, it goes to your stage speaker system.
- As a verb, to mix is the act of managing and shaping audio and—finally—the creating of a mix (the noun).

Audio Comes In to a Mixer...

Audio comes in to a mixer through its input—as in "putting in"—jacks. On many mixers, these jacks are located on the mixer's rear panel. They may also be located on a separate stand-alone box—a mixing processor—that's connected to the mixing control surface.

A typical mixer can accept audio from a variety of sources. You can connect:

- a microphone, or "mic."
- the outputs of a synthesizer, sampler or beat box.
- anything with a pickup, such as an electric guitar or bass.

- a CD player, turntable, reel-to-reel tape or cassette deck.
- the output of an external effects processor to which you've sent audio from the mixer.
- the outputs of a multi-track recorder so you can listen to its tracks as you record, and during playback.
- a mixdown deck so that you can hear your final mix as you record it, and so you can play it back as well.

In Chapter 5, we'll discuss the types of connectors you'll be using.

An Important Word: Signal

Audio travels through cables in the form of electrical voltages. Traditional—that is, analog—mics and instruments send out voltages that imitate the peaks and valleys of the audio. In digital gear, the voltages represent the 1s and 0s that make up digital data. In either case, audio traveling through cables and within a mixer is generally referred to as "signal."

When Audio is Inside a Mixer...

The following descriptions will introduce you to some of the things you can do to a signal within a mixer. We'll discuss these activities in more detail later on.

Some mixers can memorize the changes you make throughout a song, and play them back—this is referred to as "automation," as we'll discuss in Chapter 4.

You can change a signal's:

- VOLUME—or "level," so that it's at the optimal volume for recording on a track, or fits well within a mix. During a song or project, you can make minor level adjustments to bring out too-quiet phrases in a performance; you can also hide mistakes by turning them down.
- DYNAMICS—Even with its basic loudness set to a desirable level, a signal's volume constantly changes during a performance. These changes are referred to as "dynamics," and they're generally a good thing, since they make the performance more expressive. Digital mixers often contain dynamics processors—such as compressor/limiters, gates and expanders—that allow you to alter the dynamic level changes in a performance to make them more manageable, discard unwanted noise or add some dynamic variation to dull performances.
- EQ—short for "equalization," lets you change the tonal quality of a signal by adjusting the volume of the various frequencies it contains. For example, when you turn up the treble on your home stereo, you're increasing the volume of the high-frequency content in whatever it is you're listening to. When you increase the bass, you're turning up its low frequencies. Both of these are examples of using EQ. In a mixer, you can select and change the loudness of any part of a signal, not just its highest- and lowest-frequency components.
- PANNING—A stereo mix is divided into two signals: one that plays through a left-hand speaker, and one that plays through a right-hand speaker. When you're creating a stereo mix, you can place a signal at any stereo position from the far left to the far right by setting it to play more from one speaker than the other. When audio plays equally from both speakers, it's heard in the center. Positioning a signal in the stereo field is called "panning."

Panning is not really an issue when you're creating a mono mix—such as when you're feeding a mono PA—since the same sound comes out of all of your speakers.

You can also add effects to a signal. Effects are digital processes that take an original, or "dry," signal and turn it into something else. This new thing can then be mixed in with the dry signal, or replace it altogether. We'll describe when to use which technique in the next chapter.

Some digital mixers provide a set of built-in, internal effects processors that you can use. You can also connect an external processor to a mixer and route signals to and from that device. This can be handy when an external device offers an effect not available in the mixer itself.

Audio Leaves a Mixer...

Audio leaves a mixer through its output jacks. As with input jacks, a mixer's output jacks are often located on its rear panel or on an attached mixing processor.

A mixer's outputs can send audio to:

• a track or tracks of a multi-track recorder.

• external effect-processing boxes.

• a headphone or monitor amplifier for performing musicians.

• a mixdown recorder, such as a DAT (Digital Audio Tape) or minidisc machine.

• an amplifier driving a stage speaker system.

• a private listening system such as control room or studio speakers.

In Chapter 5, we'll get into the types of output connectors you'll be using.

Some beginners think of mixer outputs as being useful only for sending tracks to a multi-track recorder or when creating a finished mix. In fact, they can be used at any time for sending audio to musicians' headphones in the studio or to their monitors onstage, and for directing signal to any external processors or speaker systems.

Some people like to dirty-up a clean sound by sending it from the mixer into a speaker or guitar/keyboard amp, mic'ing the speaker or amp, and then using the mic'd version in the mix or on a track, mixed in with the original signal or alone.

An Important Pair of Words: Send and Return

In a mixer, signals are frequently sent from one place to another and then back to their starting point or some other place. This round-trip routing is such a common technique that special terms have been coined to describe either end of signal's journey: "send" and "return." These words are used as both verbs and nouns.

• As a verb, "send" describes the action of directing a signal from one place to another at the beginning of its journey. As a noun, it's used to describe the mechanism on a mixer—a button, knob or parameter—used for this purpose. The place from which it leaves—a connection point within a mixer or an output jack—may also be referred to as a send.

- As a verb, "return" describes the action of bringing the signal back from wherever you sent it. As a noun, it describes the mechanism for controlling the way the returned signal is handled, such as a level or panning control. The place to which it comes back—a connection point or channel within a mixer, or an input jack—may also be referred to as a return.

Here are some examples of how "send" and "return" are used:

- As a verb: "I'll send this vocal from Channel 1 to a reverb and return it to Channel 2."

- As a noun describing a mechanism: "I'm going to adjust the send from Channel 1 so that the vocal isn't too loud when it goes into the reverb. Then, I'm going to lower the volume of the reverb return and pan it to the center of the mix."

- As a noun describing locations: "I'm going to connect the send to my favorite foot pedal delay, and bring its output back to one of my auxiliary returns."

The Benefits of a Digital Mixer

Superior Sound

Digital mixers can capture and preserve the sound of signals more accurately than analog mixers. This is due to the fundamental way in which the two types of mixers handle audio. As we said earlier, audio from a conventional, non-digital mic or instrument arrives at a mixer's input in the form of an electrical voltage that imitates the shape of the mic or instrument's sound waves:

- In an analog mixer, as the signal passes through various components within the mixer, its resemblance to the original sound degrades. This is because each component creates its own imitation of the signal and passes that imitation on to the next component. This produces a result similar to—though far more subtle than—a round of the game Telephone, in which a phrase totally changes as it's passed among a group of people.

- In a digital mixer, the original voltage is converted into a series of numbers, and it's these numbers that get passed from component to component. Since, for example, "1234" will still remain "1234" no matter how many times it's handed from one component to another, there's no degradation of the signal as it flows through the mixer. The only changes made to the sound—and the numbers that represent it—are those that you deliberately make.

Greater Control

Since digital audio signals are comprised of binary data—the language of computers—they can be subjected to any process that can be dreamed up by software programmers. As a result, digital mixers—which are actually computers—can provide a host of features that an analog mixer just can't match, including:

- unheard-of of sound-altering possibilities.

- customizable input, output, and internal routing options.

- the memorization of routings and settings.

- the recording and playing back of changes you make to the mixer's settings.

3—Crucial Concepts

What's a Channel?
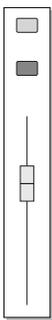
The main action in a mixer occurs in its channel strips. Channel strips, or simply "channels" for short, provide a set of tools that allow you to shape the sound of a single audio signal. This can be any kind of audio signal: a microphone, instrument, track from a multi-track recorder, a turntable. Most of the sound shaping that occurs in a mixer is accomplished using the tools provided in its channel strips.

A mixer can have any number of channels. Mixers typically have 4, 8, 12, 16, 24 or 48 channels, which would allow you to work with 4, 8, 16, 24 or 48 audio signals, respectively. Most of the time, a channel controls a single mono signal, though some mixers offer special stereo channels that treat two mono signals as a single stereo entity.

It can be helpful to picture a channel's signal flowing through the strip from top to bottom, with the top representing the channel's input jack and the bottom representing the channel's final output. This actually mimics the flow of audio, or "signal flow," from one channel tool to the next. Chapter 6 will describe in detail, and in order (an important topic), all of the things a signal encounters as it makes its way down to the bottom of a channel, adding more detail to the picture on the right.
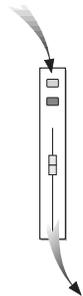
In addition to shaping a signal as it makes its way through a channel strip, you can:

- direct it to an internal effect or to an external effect processor and then back into the channel for further shaping.
- send a copy of it to an internal or external effect processor. The processor's output can then be routed to a desired destination of its own, such as a track or into a mix.
- send a copy of it to a headphone or stage amplifier so that performers can hear what's going on.

At the bottom of the channel strip, you'll find switches, and sometimes level controls, that allow you to send a copy of the channel's signal—having passed through all of the channel's tools—to the desired destination. You might send it, for example, to a track on a multi-track recorder, or into a mix. (On some mixers, these controls may be located elsewhere in the strip.)

Input Patching

Each channel on a traditional mixer is directly connected to an input jack so that the mic/instrument/whatever plugged into a jack is controlled by its like-numbered channel. An instrument connected to the first input jack would be controlled by Channel 1, the signal at the second jack would be controlled by Channel 2, and so on. This scheme is nice and simple, but it can mean that you have to re-plug cables if you want to change the channel controlling a particular signal for some reason.

In a modern digital mixer with patching capabilities, any input jack can be routed to, and controlled by, any channel strip or set of auxiliary controls (we'll get to auxiliary controls later in this chapter). The signal from a single jack can even be routed to more than one channel in situations where you'd like to wind up with two different-sounding versions.

What's a Bus?

There are basically two methods for sending a signal from one part of a mixer to another or to an output jack connected to an external device. You can:

- route, or assign, it to a "bus" that's connected to the desired destination.
- send it directly to an output jack.

Understanding busses is absolutely critical in understanding a mixer. While some mixers label their busses for particular purposes—record busses, mix busses, auxiliary busses and so on—all busses are really the same. Fortunately, a bus is a pretty simple thing.

A bus is a pathway down which one or more signals can travel to reach a destination.
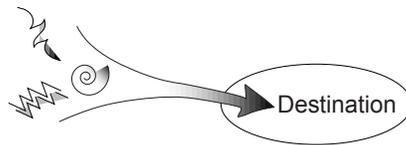
Some traditional mixers refer to a bus as a "subgroup."

Busses are one of the most helpful tools you'll find in a mixer. For example, you can add an internal or external effect to a signal by assigning it to a bus that's connected to the effect's input. And there are a number of circumstances in which you might want to send more than one signal to a single common destination:

- You might want to combine the sound from two or more microphones or instruments on a single track in a multi-track recorder.
- You might want to send several signals to the same reverb so that they share a common ambience.
- You might want to send a group of signals to a mono headphone or monitor amplifier.

Working with a pair of busses at the same time is even more useful:

- You can route a pair of busses to two tracks on a multi-track recorder and assign the desired channels to both of the busses. The resulting tracks contain a stereo image that incorporates all of the signals. This is a great way, for example, to reduce an array of mics used on a drum kit to a single stereo image—a pair of tracks—on a multi-track recorder. You can also position signals at specific places within the stereo image you've created.
- You can use this same technique to reduce a group of recorded tracks to a single stereo image by copying—or "bouncing"—them together into a pair of new tracks.
- If you send your pair of busses to a stereo headphone or monitor amplifier, you can provide your musicians a stereo mix to listen to.

- When you create a mix, what you're actually doing is assigning all of your channels to a pair of busses and routing them to your mixdown recorder or stage speaker amplifier.

Once you understand that all busses are the same, you'll find all kinds of uses for them. In any mixer that allows you to change its internal patching, you can use any bus—regardless of its name and supposed purpose—when you want to send multiple signals to the same destination. (You may be able to route a bus's signal to an output jack of your choice instead of being forced to use a jack that's permanently connected to the bus.)

Pre and Post

If you recall, we mentioned on Page 7 that a signal typically flows through a channel strip in a very particular order (which we'll describe in detail Chapter 6). When you want to send a channel's signal to a bus, a mixer will usually provide a set of points within the strip's signal flow from which its signal can be copied and sent to the bus.

The point you select can make a big difference in the sound of what goes into a bus, since it determines whether the signal will have already been affected by some or all of the strip's sound-shaping components before going there. Though mixers offer different locations within the channel's signal flow from which you can send a signal to a bus, the available choices are always referred to as "pre-some-particular-channel-component" when you're copying the signal before it gets to a particular component, or "post- some-particular-channel-component" when you're copying the signal after it's passed through a particular component. A digital mixer may allow you, for example, to send a channel's signal to a bus pre-delay, post-EQ or post-fader. We'll discuss this further in Chapter 6.

Standard Busses

A traditional mixer divides its busses into two types: standard and Aux busses. Standard, general-purpose busses are most frequently used for the sending of one or more signals to a track on a multi-track recorder—some mixers label them, in fact, as "recording" busses. Typically, each bus will be connected to its own physical output jack that can be connected to a track input on a multi-track recorder. As with any bus, of course, a standard bus can be used to send one or more signals to any destination connected to its output jack.

How a Signal is Sent to a Standard Bus

Each channel strip will provide a set of on/off switches—often buttons located at the bottom of the strip and numbered in pairs such as "1-2," "3-4," etc.—that allow you to assign the channel's signal to a pair of busses. When any of these switches is engaged, a copy of the signal is sent to both of its busses. The level of the signal to be sent is determined by the channel's final level control (we'll discuss channel strip pan and level controls in detail later).

The strip's pan control sets whether the signal is sent to both busses equally, or more to one than the other. When a signal is panned to the left, more of it goes to the odd-numbered bus; when it's panned to the right, more of it goes to the even-numbered bus.

It's worth noting that when a channel is assigned to a bus, a copy of its signal is fed into the bus without interrupting the flow of the signal through the channel strip.

"Aux" Stands for "Auxiliary"

Traditional mixers also provide a collection of Aux busses and returns.

In common usage, an Aux bus—your mixer might label it as an "Aux send"—is used as a way to send a group of signals to an effect. An Aux return will control the volume and panning of the effect's output, and will allow you to send it to a track (or tracks) on a multi-track recorder or into a mix. Some even label the Aux returns "Effect returns." This may make Aux busses simpler to understand, but it misses the wonderful flexibility that Aux busses and returns really provide when you're in need of extra busses and channels.

Here's the truth:

- Your mixer's Aux, or Auxiliary busses, as their name implies, are merely an additional set of busses. You can use them for anything you might use any bus for, such as sending a group of signals to a track, an effect, an external device of some sort, or to a headphone or monitor amp. A mixer will typically provide an output jack for each Aux bus so that you can send its signal to an external device if you wish.

- Aux returns are likewise just an additional set of spare channels. With the proper adapters, you can connect anything you'd connect to a channel strip to an Aux return by plugging it into an Aux return input jack. Though Aux returns typically offer fewer tools than your mixer's channel strips—perhaps only volume and panning controls and some mechanism for routing their signal to a track or into a mix—they can be mighty handy when you need to plug in just a few more instruments or microphones, and all of your full channel strips are already in use. By the way, an Aux return may be a single, mono channel or a pair of channels linked together as a single stereo entity.

- Though they share "Aux" labels and numbers, Aux busses and returns are not really connected to each other. They're separate auxiliary tools that you can use as your needs dictate. Just because you send some signals to an external device using Aux Bus 1, for example, doesn't mean that you have to return the device's outputs to the mixer using Aux Return 1. You could return it to a channel, for instance, or to another Aux return altogether.

How a Signal is Sent to an Aux Bus

More control is typically provided for sending a channel's signal to an Aux bus than to a standard bus. Somewhere high up on the channel strip, you may find some sort of mechanism for enabling or disabling the sending of the channel's signal to each of the mixer's Aux busses. You'll also find a level control for each Aux bus that allows you to set how much of the channel's signal will be sent to the bus.

The Roland FlexBus™

Roland's digital mixers provide a third type of bus that recognizes and makes the most of the wide-open way in which a bus can—and should—be used: the "FlexBus." Any FlexBus can be used for any purpose, and the means by which a signal is assigned to a FlexBus reflects this. On a Roland mixer with FlexBusses, each channel strip provides:

• An on/off switch for assigning the channel's signal to each FlexBus or not.

• Level controls that allow you to set the amount of the channel's signal to be sent to each FlexBus.

With this expanded amount of control available for every FlexBus, there's no need in such a mixer for Aux busses at all. You won't need Aux returns, either, since a Roland mixer with FlexBusses provides so many channel strips. By routing each effect output to a channel strip, you'll gain far more control over its signal than you'd have with a conventional Aux return.

What are Effects and How Can They Be Used?

As mentioned earlier, effects are digital processes that take a dry signal and turn it into something else. Effects can make a tremendous difference to the sound of your tracks and mixes. Some of the most common effects are:

• REVERB—Reverb is an imitation of the sound of an acoustic space. This can be a real-world space, such as a concert hall or a particular room, or something completely unlike anything the physical world has to offer. When added to a signal, reverb makes the signal sound like it actually occurred in the space the reverb implies.

• DYNAMICS-altering effects—The compressor/limiters, gates and expanders first mentioned on Page 4 are effects that allow you to control and enhance the volume changes within a performance.

• DELAY—A delay is a copy of the dry signal that's played back late. Delays are measured in time, that is, the amount of time by which the copy's playback is postponed. Delays can be used in a great number of ways. Short delays can sound like artificial-sounding ambiences, or can make a performance sound like two nearly identical performances (this is called "artificial doubling"). Longer delays are more audible and act as a special effects. You can delay the delay itself—this is called "feedback"—to create multiple copies of the dry signal; this familiar sound was used often on early rock 'n roll vocals.

• CHORUS, PHASING and FLANGING—These three slightly different effects all make a dry signal shimmer and swirl by creating close-together, multiple delays whose delay times vary.

• DISTORTION—Distortion makes the dry signal sound like it's being run through a guitar amplifier at too high a volume.

• MODELING—Modeling applies the key characteristics of one sound to another. Using Roland's exclusive COSM™ modeling technology, for example, you can make your signals sound as if they were captured using a variety of classic mics. You can also use COSM to transform one guitar into another, or make it sound as if it's going through a vintage guitar amp.

If your mixer's digital, it may contain its own internal effect processors: computer chips that create effects. Consult your mixer's documentation to learn more about the effects it offers.

In either event, there are two ways to apply effects to a signal:

- In-line, or insert—This method allows you to replace the dry signal with its effected version, an approach that makes sense in any situation in which you no longer want any of the original, dry signal to remain. This would be the case, for example, when you're running the signal through a compressor, a distortion effect, or Roland microphone modeling—all you'll want to wind up with is the final processed signal. This would be the case with any other dynamics-based effect such as a gate or expander as well.

  To process a signal with an in-line effect, you can use one of its channel's "inserts." An insert is a point within a channel strip's signal flow that's connected to a special internal connection—and/or a physical jack—that provides both output and input capabilities (special cables are available for this purpose). When you route a signal to an effect using an insert, you interrupt the flow of signal through the channel, re-direct it to an internal or external effect, and then return the effected signal back into the channel to complete its journey to the bottom of the strip.
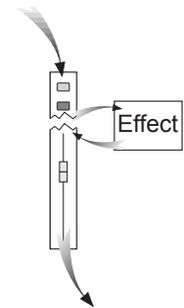
- Using an Aux bus or Roland FlexBus—When you want to add an effect such as a reverb or delay to a signal, and wind up hearing both the original signal and the added effect, you can do so using a bus. Each channel strip will allow you to send a copy of the channel's signal to an Aux bus or FlexBus without interrupting the flow of audio through the channel strip. Typically, you'll be able to select the place within the channel strip's signal flow from which the copy will be sent. You can then direct the selected bus to the desired effect and patch the effect's output (or outputs, if it's a stereo effect) to the desired Aux return or channel strip. You can route the effect's output to the same track to which you're sending the channel strip's final output, or into your mix. An effect you add to a signal this way may be referred to as a "loop effect."
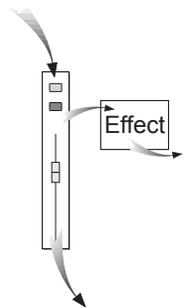
  Often, singers like to hear reverb on their voice as they sing. You can do this by using a bus to send a copy of the vocal to an effect processor, returning the output of the processor to a channel, and assigning the channel to the bus you're sending to the singer's headphone amp.

One More Type of Bus: The Cue Bus

There's one more type of bus you may encounter: a Cue bus. Again, there's nothing unique about this kind of bus except its normal use. A Cue bus, which is typically a pair of mono busses grouped together as a single stereo entity, feeds signal to the musicians with whom you're working, either through headphones—as when you're recording—or through stage monitors in performance. A Cue bus's output must be connected to the appropriate headphone or speaker amplifier.

4—Mixer Geography

What's Where and Why

Mixers are designed to be as convenient as they are flexible. Physically, you'll find that a mixer's inputs and outputs, channel controls and so on are arranged in distinct sections. Within each section you'll find all of the tools you need for a particular activity. The placement of these components generally reflects the flow of signal into, through and out of your mixer, going from left to right. The following illustration shows what we mean, starting with rear-panel input jacks, passing through the individual channel strips, on into controls that affect the mixer as a whole, and ending with rear panel output jacks.

Into the rear-panel inputs...

...out of the rear-panel outputs.

...through the mixer-wide controls...

...through the channel strips...
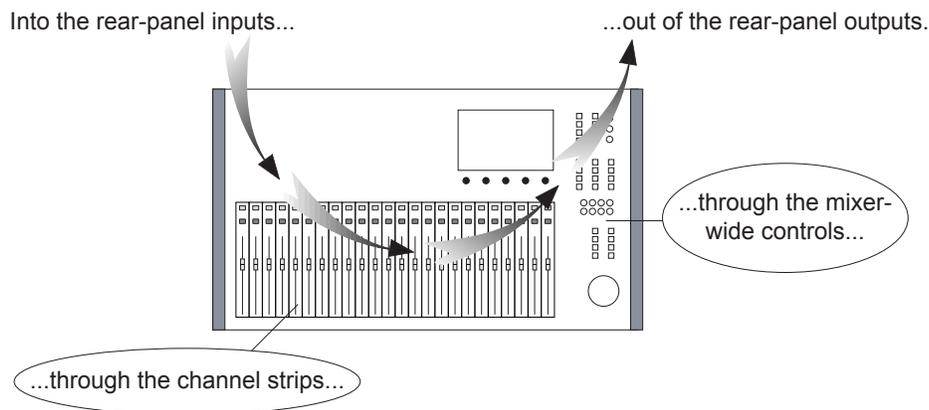
Your mixer's sections may be arranged somewhat differently than this one's. For example, if you've got a component-based mixer with a separate mixing processor, the inputs and outputs will be on the processor, not the mixing control surface itself.

The Rear Panel of Your Mixer

Typically, a mixer's rear panel is where you'll find its input and output jacks. On many mixers, you can plug microphones, instruments, multi-track recorders, mixdown recorders, external devices connected to Aux returns and so on directly into the input jacks. Likewise, you can connect its output jacks directly to a multi-track recorder, mixdown device, speaker amplifier, external device fed by an Aux bus and so on. On some digital mixers, the rear-panel jacks provide a way to link the mixer to an external mixing processor that contains the standard input and output jacks. In either case, the concept is the same: all audio coming into the board comes into a rear-panel connector, and all audio coming out of the board comes out of a rear-panel connector.

Chapter 5 will describe the types of connectors you'll encounter.

Some mixers also provide small rear-panel knobs and/or switches that may:

- aid in setting the proper level for a signal coming into the mixer.
- activate phantom power for condenser mics that require an electrical current to operate.

The Front Panel of a Typical Mixer

When you've connected everything you need to connect to a mixer, the main action moves to its front-panel. Here's an introduction to the sections of a typical mixer.

Monitor Controls

A monitor is a speaker you listen to as you work. Monitor controls let you manage what you hear in this speaker, or speakers when you're listening in stereo. You can also listen to your mixer using headphones. In general, listening is referred to as "monitoring."

Onstage, each of the speakers performers listen to is also called a "monitor."

Some mixers provide separate monitor controls for the person operating the mixer in a control room, and for musicians listening to another set of monitors out in a studio. A mixer may also provide separate digital and analog monitor outputs and controls so that you can use either digital or traditional analog speakers.

There are basically two aspects to a typical mixer's monitor controls. These controls may be buttons on the mixer's front panel, or may appear as parameters in a digital mixer's display:

- The monitor selector lets you choose what you'd like to listen to. Most mixers allow you to listen to their main stereo outputs, any Aux bus or your mixdown recorder (listening to the tracks of a multi-track recorder is a special case we'll discuss on Page 16). Your mixer may also provide a switch that allows you to listen to your mixer in mono. Being able to select what you're listening to is important since it lets you, for example:
  - listen to your mixer's main stereo outputs to hear what you're doing as you work.
  - listen to what you're sending to an effect or external device on an Aux bus. This can help you figure out exactly which signals are being sent and at what volume when you're using your Aux busses.
  - lets you listen to the inputs of your mixdown recorder as you mix so that you know what it's receiving from the mixer and what it will record. You can also listen to your mix being played back when you're done.

The monitoring volume control lets you set your listening volume. Some mixers also provide a "dim" switch that instantly cuts the listening volume, and/or a "mute" switch that silences your speakers completely. Your mixer may have separate volume controls for control-room, studio and headphone listening.

It's important to realize that the monitor controls affect only your mixer's monitor outputs—they have no effect on its other outputs, such as those connected to busses. This means that you can change what you're listening to, and at what volume, without fearing that you're affecting what you're sending out of the mixer's other output jacks.

Metering

Your mixer's meters allow you literally to see what you're doing. On a modern mixer, a meter is typically a vertical row of LEDs or set of bars on your display that shows the loudness of a signal: the taller the column of lit LEDs or bars, the louder your signal is. (On older mixers, a meter was a needle moving left-to-right over a printed background scale.) Your mixer's display can provide you with meters for all sorts of things. It can show you a pair of meters when you're working with a stereo image. It may also be able to show you the levels of any busses you're using and show you the level of your Aux returns.

You may be able to purchase an optional attachment that provides an array of larger meters that let you to view many signals' levels at once. Such a device is called a "meter bridge."

You can adjust a signal's appearance in a meter by adjusting the appropriate level control.

Clipping: How Loud is Too Loud?

There's an art to understanding what you're seeing when looking at a meter, and, in digital devices, it's an important one to master since the best sound is achieved when a signal is as absolutely loud as it can be without being too loud.

"Too loud" means that the signal has reached a volume at which distortion, or "clipping," is heard. Unlike distortion in a guitar amp, or even in an old-fashioned analog mixer, clipping is never a good thing and should always be avoided.

Clipping occurs when the level of a signal exceeds the capabilities of some component in the mixer. If the input of a channel is pushed beyond its limit, clipping will occur (we'll talk about this in Chapter 6). If a channel's output, a bus's output or mixer's output capacity is exceeded, clipping will also occur. Fortunately, you can hear clipping—it's usually a nasty click or thumping sound—so you can tell when it's occurring.

Levels are measured in decibels, or "dB." In digital devices, the precise level at which clipping occurs is assigned a numerical value of "0 dB." Levels below 0 are assigned negative numbers.

Each digital device—mixer, recorder or effect processor, for example—warns you about clipping in its own way. Some mixers display a clipping indicator when your signal's level reaches 0 dB. Some will do so when a signal comes near 0 dB—for example, when a signal reaches -6 dB—as a warning that you're approaching trouble but not quite there yet. Consult your mixer's manual to find out what its clipping indicator looks like and what it really means.

Meters on mixers can show a signal's level in either of two ways:

- Averaging, as its name implies, averages out the changes in a signal's level, giving you the best overall sense of the signal's volume. However, averaging responds to level changes too slowly to show sudden extreme jumps in volume called "transients." This can pose a problem with sounds that contain transients since these instantaneous bursts of volume can cause clipping even though you can't even see them on a meter.
- Peak metering, on the other hand, shows the transients in the signal, allowing you to get an accurate view of any sudden volume spikes that may cause clipping. The downside to peak metering is that it can frighten you into setting a signal's level too low once you see all the

transients it contains. Not all transients cause clipping—only those that are loud enough and last long enough do so—therefore, setting a signal so that none of its transients are over 0 is probably setting it at too low a level. Snare drums, for example, are notorious for their many fast —and harmless—transients. If you record a snare drum at a level that leaves you with no transients above 0 dB, you may be recording it at an unnecessarily low volume.

When recording or mixing live sound, it's a good idea to meter and set your levels using the averaging method, and to switch to peak metering only if you hear clipping, and occasionally just to make sure that you're not missing something.

In the Red

An LED-based meter may have a group of green LEDs at the bottom, amber LEDs in the middle and red LEDs at the top. As a meter's signal gets louder, it travels up the ladder, lighting more and more of its LEDs. If it reaches the point where it becomes too loud, the meter's red LEDs light. On older mixers, the needles in VU meters moved over a red background when a signal was too loud. In either case, when a signal is too loud, it's viewed as being "in the red." Even if your mixer's meters are not in color, the principle is the same, and you can still think of being too loud as being "in the red" when your clipping indicator lights.

Mode-Switching Controls

Channel strips can often function in different operating modes. While the strips may indeed be channel strips in one mode, they may function as tape returns in another mode, or as fader group masters in another—we'll discuss tape returns and fader groups on the next page. In such cases, your mixer will provide a set of controls for switching between these modes.

Channel Strips

We've already learned a bit about what your mixer's channel strips do, back on Page 7. Chapter 6 will get into the details.

Multi-Track Tape Returns

It's likely that your mixer provides a set of input jacks to which you can connect the track outputs from your multi-track recorder—some mixers refer to these as "Tape" or "MULTI" inputs. You'll need to make these connections to hear your tracks through your mixer. Your digital mixer may provide special jacks for connecting the mixer to digital recorders and other digital devices.

These multi-track tape returns are usually connected directly to your monitors, allowing you to listen to your multi-track recorder's tracks. You'll obviously want to do this when playing back any tracks you've recorded—the number of tracks you'll be able to listen to depends on your mixer.

It's a smart idea to listen to the tracks of your multi-track recorder even as you record. Most recorders will allow you to listen to the signal each track is receiving before it's actually been recorded. Listening to the recorder—as opposed to listening to the mixer's outputs—will let you be absolutely sure that you know what it's getting from your mixer,

and therefore what it will actually record. This will help prevent monitoring mistakes—where what you're listening to isn't actually what you think you're listening to—that can result in your tracks not sounding as you intended.

You'll be able to monitor tracks in one of two ways, depending on your mixer.

- Some mixers offer "in-line" track monitoring, where each strip actually controls two signals: an input signal and a signal coming from a track on a multi-track recorder. The strip will usually provide separate controls for each purpose: a full set of channel tools for the input signal and a reduced set of controls for listening to the track. You'll typically be able to switch, or "flip," these controls, allowing you to assign the more comprehensive channel tools to the input signal or the track, as your needs dictate.
- Some mixers split the controls for monitoring tracks out into their own separate section of the mixer's front panel—these are, in fact, called "split mixers."

Fader Group Controls

Some mixers allow you to combine the control of two or more channel strips into " fader groups." This type of control can be handy when the strips are handling related signals assigned to different busses, or going to separate tracks. For example, fader groups can be useful when dealing with larger numbers of channels, such as those that encompass the many microphones used on a drum kit. By assigning the channels to a fader group, you can raise or lower all of their levels at once using a single level control, making the process simpler, and also assuring that the levels are changed by exactly the same amount.

Three channel strips...

...controlled by single group strip.

Your mixer may provide separate group-control strips. In a mixer with multiple operating modes, you may be able to switch the function of your channel strips so that they can be used for the controlling of groups.

Some digital mixers also provide channel linking, a mechanism that allows you to manage the volume of a pair of channel strips using a single level control. This can be handy when you're using two channel strips for the controlling of a single stereo image—perhaps from a stereo keyboard or group of singers, for example.

Aux Controls

Your mixer may provide separate Aux controls that allow you to manage your Aux busses (remember, these are sometimes referred to as "Aux sends") and returns. If so:

- your mixer will typically supply an Aux bus master level control that lets you raise or lower the volume of the corresponding Aux bus. While each channel strip allows you to set the amount of its signal to be sent to an Aux bus (as we'll see in Chapter 6), you'll want a way to set the overall volume of the bus so that it's at its optimal level as it's routed to an internal effect, headphone amp or other external device. That's what the Aux bus master level control does.

- your mixer will also typically provide controls for managing each Aux return. You can select exactly where you want the signal coming into the Aux return to be sent—you may want it to go directly into the overall mix, to a recorder track or to a pair of tracks. You'll also be able to set the level of the return, and set its pan position if it's going into a stereo mix or pair of tracks. If the return is actually a pair of returns presented as a single stereo entity, you'll be able to adjust the stereo balance between its two sides.

A Roland console with FlexBusses and a sufficient number of input channels will not need, or have, separate Aux sends and returns.

Bus Control

Your mixer will probably provide a means by which you can control the overall loudness of each of its standard busses or FlexBusses.

Master Strip

Every mixer has a master strip that contains the mixer's main volume control. This control raises or lowers the level of the mixer's main mono or stereo mix, and thus the volume of what comes out of its mix outputs. You can use the master strip's level control to set the overall loudness of a mix, and to create fade-outs at the ends of songs (or fade-ins at their fronts, if you so desire).

When recording a mix while monitoring your mixer's main stereo outputs, it's best to adjust your listening level using your monitor level controls, not the master strip's level control. Adjusting the monitor level has no effect on what comes out of the mixer's main outputs, but adjusting the master strip's volume control does, and may result in sending your mix to your mixdown recorder at inappropriate levels.

Display

On the front panel of a digital mixer, you'll find an LCD display that's central to every mixer activity. As you do different things, the contents of the display will change. For example, in some operating modes, the display may show you the mode's available settings—your manual probably calls these "parameters." In other cases, it will provide information that helps you accomplish whatever it is you're trying to achieve, such as showing you your signals' levels when you're trying to set their volumes. Your manual will tell you how to switch between operating modes.

It's likely that you'll find a set of buttons or knobs beneath your digital mixer's display. These are sometimes called "soft buttons" or "soft knobs" because their function is not a hard, fixed thing—it changes. What each soft button or knob does depends on what you see above it on the display at any given time.

Automation Controls

Many digital mixers can automatically play back changes you've made while mixing, using a feature called "automation." Your mixer may be automatable in either of two ways:

• A mixer may contain its own recorder that keeps track of what you do, and can play back your actions. Your mixer may allow you to refine an automation recording by doing things over until you get them right, or by allowing you to edit directly what's been recorded.

• A mixer may also transmit MIDI data representing your mixer settings. This can be recorded and played back by an external MIDI sequencer, a recorder that captures MIDI data.

If your mixer supports automation, you may find a set of automation controls on its front panel.

What a Mixer Can Remember

One of the great advantages of a digital mixer is that it has a memory that can contain all sorts of useful things. Your Roland mixer may be able to save, store and recall any or all of the following:

• an EZ Routing™—A Roland EZ Routing is a template that contains a favorite routing setup, and, in some mixers, other settings as well. You can use an EZ Routing over and over again in as many songs or projects as you like. When you recall an EZ Routing on some Roland mixers, the mixer's display will present a series of questions to assist you in the construction of the desired setup.

• a scene—A scene captures all of your mixer's current settings. You can save a scene for each song or project you work on. When you wish to resume work on the song or project, you can recall its scene, and immediately re-establish its settings. A separate physical button on your mixer is assigned to each scene, making each one fast and easy to recall.

• a snapshot—A snapshot is a picture of your mixer's settings, or a particular group of settings, at a specific moment during a song or project. Your mixer can remember the moment at which the snapshot was taken and recall the snapshot at the proper time.

And, as we said earlier, your mixer may also be able to continuously record and play back all of your changes as a song is mixed, performing them as you sit, arms folded, watching all of your moves occur.

Utility Controls

Mixers offer a wide array of special functions that help make mixing easier and more accurate. Your mixer may provide any of the following conveniences and tools:

• Talkback microphone—Some mixers contain a built-in microphone that allows the mixer operator to talk to musicians in a studio or onstage, either through their headphones or a speaker. It will also allow you to "slate" each recording attempt by assigning it a take number that you'll announce into the mic as recording begins; this will help you keep track of which take is which. A talkback system will typically include the mic, a button for activating and de-activating the mic, and a volume control for the mic.

- Speaker calibration tools—Your mixer may provide white- and pink-noise generators that you can use for optimizing your monitors' performance, or your stage speakers when doing live sound. These generators produce a static-like noise containing a broad range of frequencies that can be played through your speakers into a microphone. The mic's signal can be analyzed to learn how accurately your speakers are performing in the room in which you're working. You can then apply EQ to the appropriate outputs to make any necessary corrections to the speaker's frequency reproduction.
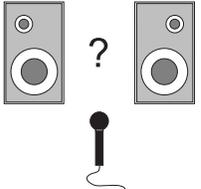
- One of the most powerful utilities in a Roland digital mixer actually resides in its effects area, but since it relates to altering the sound of your monitors, we'll mention it here. When it's connected to the appropriate Roland digital speakers, your Roland digital mixer may allow you to use Roland's exclusive COSM speaker modeling to change the sound of your speakers so that they emulate the characteristics of a variety of popular studio reference monitors. You can thus try out your mixes on different speakers systems by simply switching between different COSM speaker models.

- Oscillator—An oscillator is a device that generates a pure audio tone called a "sine wave" at a specified pitch, or frequency, and at a steady level that you can set. This tone can be handy for all sorts of things: it can help you set multi-track tape input levels so that they equal the output levels shown in your mixer's meters, making life much simpler. It can help you pan signals within a stereo field with greater accuracy. You can also record oscillator tones on a tape for reference purposes when you'll be sending a tape to another studio or mastering facility.

What Else?

Your mixer may provide:

- an area above or below each channel strip on which you can write what you're using the strip for. This will free you from having to memorize what's where. Be sure to use only erasable ink!

- Digital mixers may allow you to name each channel, so that its name appears in your display when you adjust its settings.

- MTC (for "MIDI Time Code") or MMC ("MIDI Machine Control") transport controls that allow you to control any connected recorders or MIDI sequencers that support MTC or MMC. These controls may let you control your recorder's transport directly from the mixer; you may even be able to arm recorder tracks for recording.

- Macro controls. When you repeatedly do the same thing over and over again on a digital mixer, you can memorize your actions as a "macro," a shortcut that can play back the entire series of actions. A macro can be a great time saver; it frees you from repeating a whole group of steps by performing one single action—selecting the macro.

5—Connections

This chapter describes the various physical connections you'll be making as you send signals into and out of your mixer. As we discussed in *Chapter 2*, the mixer will provide input jacks for bringing audio (and perhaps MIDI) into the mixer, and output jacks for sending it out of the mixer. Your mixer's documentation will explain the labeling of its jacks.

A connector is the device, or plug, on either end of a cable that can be inserted into a jack (a bare cable without connectors should not be plugged into a mixer).

Making Connections

It's very important that you plug a connector only into its appropriate jack. All sorts of problems can occur if you insert a connector where it doesn't belong. Fortunately, it's easy to tell what goes where by simply looking at the connector and the jack. When a connector is inserted into the correct jack, you'll feel it fit snugly as it goes in. In some cases, you'll feel a little click when the connector makes its connection. If you have to force a connector into a jack, double-check to make sure it's the right kind of jack.

When removing a connector from a jack, avoid pulling on its cable so that you don't tear it from its connector. Grab the connector itself and pull it out of the jack.

Adapters

When you've got one type of connector and another type of jack, it may be that you can still connect the two by purchasing a special adapter. Check with your local electronics store to see if they have what you need. An adapter can make a connection physically possible, and can remedy any incompatibilities between the connector and jack.

AC Power

Mixers typically utilize grounded AC power. The power cable on your mixer will depend on the country in which you live. It's a good idea to connect your digital mixer to a voltage-regulated, surge-protected, grounded socket—just as with any other digital device, sudden surges or dips in voltage can damage the mixer or cause behavior problems.

Connector Types You Need to Know

There's a range of connector types you'll encounter as you start to plug things into your mixer's input and output jacks. Your mixer may have some or all of the possible connector types, depending on its size and intended use. The nature of an audio signal passing through a cable isn't actually determined by the cable's connector type—manufacturers of audio equipment use the various types of audio connectors in different ways, depending on what makes sense for their users.

The connectors shown in this book are typical examples of each type of connector, along with its corresponding jack. The ones you have may differ slightly in appearance.

1/4-Inch Phone Connectors

Side	Front	Jack

1/4-inch phone connectors are found on cables that come out of electric instruments, such as guitars and basses, many synthesizers, samplers and drum machines, and speakers. You may also encounter phone connectors on cables connected to mics. Other studio gear may also provide phone connectors.

This connector has a single band, or "ring," around its shaft. Two of the wires within its cable are connected to the plug's tip and ring.

TRS Phone Connectors

Side	Front	Jack

An TRS phone connector utilizes all three wires in its cable, attached to its tip, ring and additional ring, called the "sleeve." TRS stands for "tip, ring, sleeve." Cables with TRS connectors may be capable of handling stereo audio, or may be balanced mono cables (we'll explain balanced cables a bit later). Insert cables also utilize TRS connectors.

Phone Connectors and Inserts

Many mixers allow you to use a "Y" cable for sending a channel signal to an external effect and for returning it to an insert jack, a special combination input/output jack. A "Y" cable is named for its (upside-down) shape: it's a single cable on one end, and two cables at its other end. An insert cable has an TRS phone plug on one end—the end that plugs into the mixer—and 1/4-inch phone connectors at its two other ends, one of which goes into the external device's input, and one of which is connected to its output.

To external device

From external device

Connected to insert point
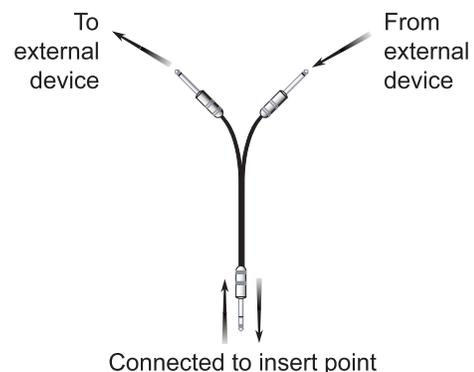
XLR Connectors

Many analog audio devices use cables that have XLR connectors, sometimes called "Cannon connectors." Studio-quality, low impedance microphones almost always use XLR connections, as do many other mic-level devices (we'll get to mic-level devices later). AES/EBU-format digital connections are also made using XLR cables.

XLR connectors come in two flavors, male and female. If an XLR connector has prongs inside it, it's male; if it has holes, it's female. A male XLR connector can only be plugged into a female jack, and vice versa.

Male XLR connector

Side	Front	Jack
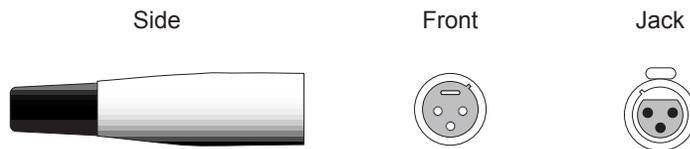
Female XLR connector

Side	Front	Jack
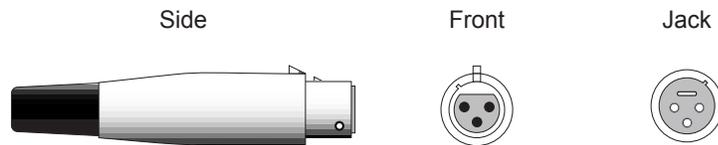
If your board can supply phantom power to mics, that power will be transmitted to the mic through its XLR cable.

Some female XLR jacks provide a release tab on top, as shown in the male XLR illustration above. When you want to remove an XLR connector from such a jack, press the tab and pull the connector to release it. If there's no tab, just pull the connector.

Phono/RCA Connectors

Side	Front	Jack
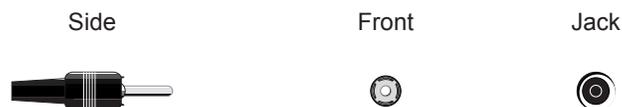
CD players and turntables typically use phono, or RCA, connectors for their outputs. Digital S/PDIF-format connections are also made using phono connectors.

Optical Connectors

Side	Front	Jack

Optical connectors transmit digital audio data through cables as light, using fiber-optic technology, an extremely reliable form of data transmission. Each optical cable can carry stereo S/PDIF-format digital audio, or up to eight streams of audio using Alesis' Lightpipe data format.

MIDI Connectors

Side	Front	Jack
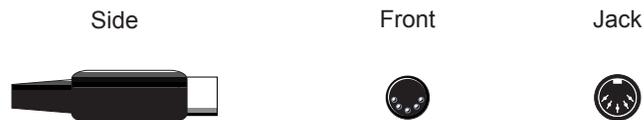
Some digital mixers can transmit and receive MIDI data. This allows you to record your mixer moves into an external MIDI sequencer as MIDI data and play them back. Your mixer may also transmit MTC or MMC data. MTC and MMC provide the ability to control the transport functions of an MTC- or MMC-compliant device, such as a multi-track recorder, from your mixer.

If your mixer supports MIDI, you'll have up to three jacks into which you can plug MIDI cables:

- MIDI In—This jack sends MIDI data from an external device into the mixer.
- MIDI Out—This jack sends out MIDI data generated by the mixer.
- MIDI Thru—This jack passes any MIDI data received by the MIDI In jack back out, unchanged. This can be handy when connecting several MIDI devices in a daisy-chain.

Mini Connectors

Side	Front	Jack

Mini jacks are used primarily by portable tape and CD players. Very few mixers actually provide mini jacks. If you need to connect your mixer to a mini-plug device, you may need an adapter to convert the mini connector to one you can use. The mini plug shown above is a mono mini plug.

Many headphones utilize a stereo mini plug, which has two black bands around its shaft instead of one. A stereo mini-to-stereo phone adapter will allow you to plug the headphones into your mixer's headphone jack.

Since mini connectors are so tiny, you may be tempted to insert one into a 1/4" phone jack in a pinch—don't do it. It won't work. Get an adapter.

Impedance

Remember how we said that audio travels through cables as voltage? Electrical connections are characterized by the degree to which they force voltage—in the case of connectors—or accept it, in the case of jacks. The amount by which they do this is called "impedance." It's important that the things you connect together be of the same type of impedance. Otherwise, their interconnections will not be correctly made, and the resulting audio will not sound as good as it should.

Audio equipment may be low-impedance or high-impedance. In microphones, pro mics are low-impedance, while semi-pro mics are typically high-impedance.

If you need to connect a high-impedance device into a low-impedance one, you can purchase an impedance-matching adapter from a music or electronics store. This will correct the mismatch by converting one of the device's impedance characteristics to match the other's.

Fighting Hum and RF

It's not uncommon to find that audio is arriving in your mixer accompanied by an unwanted buzzing tone—called "hum"—or other electrical noise, such as RF (for "radio frequency"). If you hear local radio stations, CB traffic or static through your cables, you're hearing RF.

These types of sounds are picked up by your cables as they cross your studio or stage on their way to your mixer. While it's not always possible to create a noise-free environment for your cables, there are some guidelines you can observe to help keep unwanted noise to a minimum:

• Always use properly wired, shielded cables. Shielded cables contain electrical or magnetic insulation that makes them less likely to pick up noise. You can usually trust the cables with connectors already attached that you'll find at a professional music or electronics store. Make sure to buy cables with unbalanced or balanced characteristics that match those of your mixer's input jacks (we'll explain this on the next page).

• Keep your cables as short as possible. The longer the cable, the more likely it is that it'll pick up unwanted noise.

• Keep your cables as far away as possible from electrical equipment and outlets on their way to and from your mixer.

• Take care that all of your equipment is properly grounded. Consult your manuals to learn their grounding requirements.

Balanced and Unbalanced Connections

Your mixer may provide "balanced" jacks, which should be used with balanced connectors. Balanced connections reduce noise by taking clever advantage of phase cancellation.

What's Phase Cancellation?

Audio signals are made up of repeated, or cyclical, changes in air pressure—or electrically speaking, voltage—called sound waves. A sound wave travels to its highest amount of air pressure, then to its lowest, back to its highest, and so on, many times per second.

Air pressure

Time

If you have two identical, or very similar, signals, it's important that they rise and fall at the same time—when they do so, they're "in phase." If they're out of sync, so that one falls as the other rises—as shown in this picture—they'll cancel each other out, resulting in silence. This is called being "out of phase." You can fix this by delaying one of the signals so that it lines up with the other—this is referred to as "flipping its phase."

One signal...

The other signal...

If your speakers are wired by attaching a hot and ground wire to posts on each speaker, make sure that you attach the same wires to the same posts on both speakers. Otherwise, your speakers will be out of polarity—a condition that mimics being 180 degrees out of phase—and you'll experience a serious volume drop when you listen in mono.
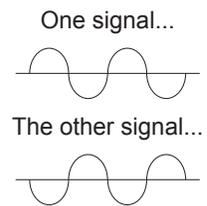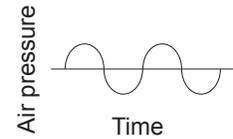
Audio cables typically contain three wires that can be attached to specified places on connectors that correspond to particular places in jacks. Normal unbalanced connectors use the "high" wire to carry the audio signal, while the ground wire provides grounding—the third, "low" wire is not used. Balanced connectors use all three wires, with the audio signal carried over the high, and a copy of the signal carried on the low. The signal on the low is flipped 180 degrees out of phase.

When the audio arrives at a balanced jack, the phasing of the low signal is flipped so that it's back in phase with the high signal. This accomplishes two things:

- First of all, the signal will be twice as loud so that it will be nice and strong.
- Second—and here's the clever part—any noise picked up by the high and low wires will be flipped out of phase, causing it to cancel and disappear.

Therefore, if your mixer provides balanced connectors, take advantage of them.

To ensure the integrity of your balanced connections, make sure that the pin assignments in your balanced cables conform to the wiring of your balanced jacks.

Matching Levels

Every piece of equipment that generates audio in a studio or on stage does so within a specified industry-standard volume range. To get the best fidelity out of the things you connect to your mixer, you'll want to make sure that you've connected them to mixer jacks that are compatible with these basic levels, or to make any necessary level adjustments to compensate for mismatches. Likewise, you'll need to make sure that you connect your mixer's outputs to the appropriate jacks on your multi-track and mixdown recorders, speaker amplifiers and/or other external devices, and to make all of the required level adjustments there as well.

+4 and -10

Audio equipment commonly produces audio at either +4 dBU or -10 dBV level. If your inputs are +4, and you connect a -10 signal, the signal will be too quiet. Likewise, if you plug a +4 signal into a -10 jack, the signal is likely to be so loud that it may distort. Your mixer may provide both -10 and +4 input jacks. Otherwise, if you need to mix and match +4 and -10 gear, you can use a separate amplifier to adjust the signal coming out of the device producing audio so that it matches the input of the device receiving it. You can also use a channel input trim—which we'll discuss in the next chapter—to reduce a too-loud signal.

Line and Mic Levels

Within the +4 and -10 level ranges are two additional sub-categories of volume:

- Mic-level devices produce a very low-level signal. As the name implies, the most common mic-level devices are microphones; electric guitars and basses also produce mic-level signals. Mic-level devices are usually connected to a mixer using an XLR plug. You can convert the 1/4-inch output of an electric guitar or bass to an XLR by plugging it into a device called a "direct box," which has the added benefit of boosting the signal's level.
- Line-level devices, such as mixers, multi-track and mixdown recorders and external effect processors, produce a considerably louder level than mic-level devices.

The level differences between these two types of signal mean that you should avoid connecting mic-level devices directly to line-level inputs and vice versa. You, however, may be able to use your channel input trims to compensate for any mic and line-level mismatches you can't help—we'll discuss input trims in the next chapter.

Making Digital Connections

Sample Rates

When a digital device converts audio into binary numerical data, it does so by creating a digital representation of the audio at a particular moment in time—this is called a "sample." By creating samples many thousands of times per second, and playing them back just as quickly, the illusion is created of a single stream of continuous audio, though, in fact, what you're hearing is really a series of separate samples.

Since audio changes constantly, a digital device most faithfully captures these changes by creating as many samples per second as possible. For example, if a signal were to change three times in a second, but only be sampled twice, one of the changes would be missed. If it was sampled three times, a more accurate representation of the sound would be achieved.

Three volume changes...   ...sampled twice would look like this: ...but sampled three times would look like this:
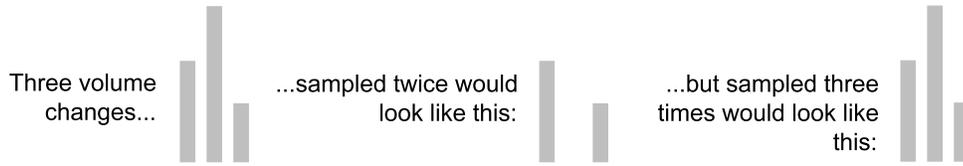
In fact, audio changes many, many times per second.

The number of times per second that a digital device samples audio is referred to as its "sample rate." Most digital devices sample audio 44,100 times per second (referred to as a 44.1 k sampling rate) or 48,000 times per second (a 48 k sampling rate). Your digital mixer and other digital devices may be able to operate at either sampling rate. When working with digital connections, the device sending the audio and the device receiving it must be set to the same sampling rate to ensure that the digital data is transmitted and received correctly. Some devices provide sample-rate converters that can convert incoming data to the required sampling rate.

Play back digital audio at the same sampling rate at which it was recorded—otherwise, the audio will not play at its original pitch.

Here's an interesting note. A digital signal's sample rate limits the frequencies it can contain according to the Nyquist theorem, which states that a digital signal can reproduce frequencies as high as roughly half of its sample rate. In other words, a signal whose sample rate is 44.1 k may contain frequencies as high as 22 kHz—any frequencies in the original audio higher than 22 kHz will not be captured.

Audio CDs use a 44.1k sample rate. If you record audio at any other sample rate, you'll have to use sample-rate conversion software to convert the signal to 44.1 k prior to burning it onto a CD.

Bit Depth

When a digital device records changes in signal level, it divides these changes into a specified number of steps. The more steps a device allocates for the representation of these changes, the finer its resolution, which is referred to as "bit depth." Digital recording is often done at a depth of 16 bits, which breaks up the dynamic range between the loudest and quietest portions of a signal into 65,536 steps. The finer resolution of 24 bits divides the signals' loudest-to-softest dynamic range into 16,777,216 steps. The main advantage of using a higher resolution is that additional steps are available for a more accurate rendering of the quieter parts of the audio.

When making digital connections, you'll get the best results when the interconnected devices share a common bit depth.

Audio CDs have a resolution of 16 bits. If you've recorded audio using any other bit depth, you'll have to convert it to 16 bits before burning it onto a CD. Various computer-based audio programs can do this.

6—Anatomy of a Channel Strip

On a digital mixer, each channel strip may function differently in different modes, as we've already mentioned. However, the main action occurs when it's operating as a channel strip—this is where the most significant chunk of a mixer's sound-shaping power resides. As such, a channel strip is worthy of a closer look. That's what this chapter's about.

The features offered in a channel strip vary from mixer to mixer. In this chapter, we'll describe the major tools you're likely to find in a channel strip—your mixer may offer some or all of them, and perhaps some other tools as well.

Channel Strip Signal Flow

This chapter discusses channel-strip tools in the order that a signal will encounter them. In most mixers, the signal's flow through a channel strip's components is fixed, and can't be changed (though it may vary slightly from mixer to mixer). Signal flows through the strip, passing through the first component to the next one, and so on. After each component alters the signal, it passes it on to the next component in its newly altered form.

Bus controls—with which you send a copy of the channel's signal to the desired bus or busses—are a special case. These controls can sometimes copy a signal from several different places in the signal flow. Even so, we've shown them near the end of the signal flow in the picture to the right because they select the ultimate destination for a channel's signal.

The subsequent pan controls affect how much of the signal is sent to the left or right side of the mixer's main mix busses, or to either side of a pair of linked standard busses in a traditional mixer without FlexBusses.

| Input Trim |
| Phase Switch |
| Delay |
| EQ |
| Dynamics Processing |
| Mute & Solo |
| Level Control |
| Bus Control |
| Pan Controls |

We'll discuss one other type of channel strip component, its inserts, separately. A mixer may allow you to use inserts to interrupt a signal at any of several points along its journey through the strip. Therefore, inserts have no one set position in the signal flow, especially since the possible insert points vary from mixer to mixer.
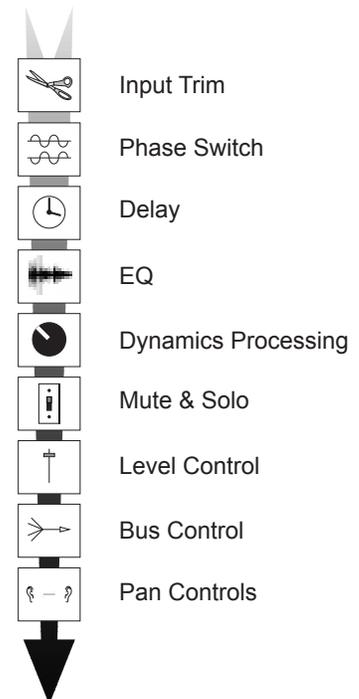
Input Trim
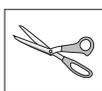
Once you've connected a signal to the appropriate input jack, you can further adjust its level going into the channel strip using the strip's input trim control. A trim derives its name from the fact that it lowers—or trims the top off—a signal's level. A trim typically only makes a signal quieter, not louder. Some mixers refer to a trim as an "attentuator," the technical term for a mechanism that reduces level.

Your mixer may allow you to adjust a signal's level continuously in 10 dB increments or by finer amounts. Some input trims utilize "pads" instead of continuos controls—pads can reduce a signal's level by pre-set, fixed amounts.

Your mixer may provide a clipping indicator for each channel strip that lights when the signal coming into the strip is too loud. If this occurs, use the trim to lower the input signal.

Phase Switch
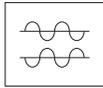
Each channel strip may provide a phase switch to help you correct any phasing problems you might encounter when mixing together two channels' similar signals. Your mixer may label its phase switches with this symbol: ⊘

A phase switch will allow you to flip the phase of either channel—you'll only need to flip the phase of one channel—to get the two channels back in phase with each other.

Phasing problems between two channels can arise when two mics are used on the same instrument—as when using two mics above a drum kit, for example—but placed improperly. If the signals are 180 degrees out of phase, you won't hear either signal; if they're out of phase to a lesser degree, you'll hear a "swirling" sound, and the signals will be missing some of their volume and content. A phase switch is most helpful when the signals are 180 degrees out of phase.

You can easily tell when two audio signals are out of phase by panning them to the same stereo location and listening. First turn one up, and then the other. If the overall volume goes down as you increase the volume of the second signal, they're out of phase.
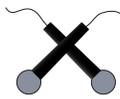
To help avoid phasing problems when using a pair of mics on the same instrument, position the mics in an "x" pattern—so that they're at 90-degree angles to each other—over the instrument.

You can also get phasing problems with improperly wired cables, or poorly wired older electric instruments, such as electric pianos.

Delay

Some channel strips contain a built-in delay with which you can shift the channel's signal later in time, so that it's heard a bit later than it actually occurs. A channel delay can be used for a few purposes:

•   You can use very short delay times to fix the phasing between two channels when they're not a full 180 degrees out of phase, and therefore not a good candidate for using a phase switch.

•   You can use the channel's delay as an effect.

•   You can sometimes line up signals that are synchronized but which start at slightly different points by delaying those that start first. (If the lag between signals isn't constant, a delay won't be as useful.)

EQ

EQ is at the very heart of mixing since it allows you to change the balance of the frequencies that give a signal its basic tonal characteristics. Fortunately, the mechanics of EQ are pretty simple—the art is in learning to understand, hear and manipulate frequencies. Though based on numbers and physics, EQ doesn't require any mathematical or scientific skills—anyone who can listen carefully can become a master of EQ.

What are Frequencies, Anyway?

As we said earlier, a sound is made up of sound waves, changes in air pressure that repeat many times per second. Some travel smoothly from their greatest air pressure to their least, as in the picture of a sine wave on Page 26. For some sound waves, the trip is more abrupt, for others more irregular. In all of these cases, however, each of these little journeys is called a "cycle." The pitch of a sound wave is determined by how many times this cycle repeats each second.

You can demonstrate this principle using your voice. Sing a very low musical note, and listen to how it vibrates, almost buzzing. Lower the pitch of the note little by little and notice how the buzz turns into discernable clicks as you get lower and lower. You can probably lower the pitch of your voice to a point where you hear nothing but a series of clicks. Take a breath and sing again, starting down low and slowly raising your pitch to somewhere in the higher part of your singing range. Notice how the clicks turn to a buzz and then disappear into smoothness as you go higher? When you sing down low, you can hear the clicks that make up your note because there's only a few of them being sung per second, and they can therefore be individually heard. As you move up in pitch, more and more of them occur per second, until the clicks are coming so fast that they actually blur together into a smooth tone. This mirrors the way that a sound wave's cycles work: the fewer cycles per second, the lower the pitch of the wave; the more cycles per second, the higher the pitch.

A sound wave's frequency is a number that reflects how frequently a sound wave's cycle occurs per second. Frequency numbering is based on Hertz (Hz)—each Hertz is one cycle per second. To save space on your mixer's display, a thousand Hertz is referred to as 1 kHz. Depending on the EQ capabilities of your mixer, you'll see frequency numbers running from about 20 Hz all the way up to 20 kHz or higher.

Nearly every sound you hear is actually made up of a pile of sound waves occurring at the same time, with each sound wave humming along at its own frequency. When heard together, the individual pitches of most of a sound's waves are not noticeable—instead, they contribute to its tonal quality, or timbre in musical terms. Higher-pitched sound waves add brightness, or treble, to the sound, while lower waves contribute bass. The frequencies of these sound waves and their relative volumes are what separates one sound from another.

When working with EQ, people use the word "frequency" as a shorthand to identify the sound waves in a sound according to their frequency. For example, if someone wants to ask you to raise the level of the higher-pitched sound waves in a sound—its treble—they'll ask you to increase, or "boost," the volume of its high frequencies, or even simply its "highs." When you reduce frequencies, you "cut" them.

Here's an interesting notion central to EQ: the sound of most instruments is actually made up of several sounds put together. For example:

• A snare drum sound is comprised of the sound of the stick hitting the drum head, the resonation of the head in the snare's shell and the rattle of the snares beneath the drum.

• A guitar sound consists of the pick hitting the string, and the resonating of the string itself.

• A piano sound is made up of the sound of the hammers striking its strings, the resonating of each string, and the sympathetic resonation of all of the piano's strings together.

Each of these elements is made up of its own range of frequencies, which may or may not be shared by other elements in the sound. By pinpointing the frequencies that constitute any of these elements, you can boost or cut the element's volume within the sound, changing the sound's overall nature.

For example, let's say you're working with a guitar with old strings. By listening, you'll realize that part of what makes the strings sound old is that the sound of the pick against the strings is muted. With EQ, you can boost the pick sound's frequencies, making the strings sound brighter and newer.

Why Use EQ?

EQ can do a number of things. It can help you:

• improve the timbre of a less-than-great-sounding instrument or voice.

• compensate for the limitations of a particular mic.

• emphasize particular components of a signal to fit better within a musical arrangement.

• create special timbral effects, ranging from the subtle—adding extra warmth or sparkle—to the obvious, such as the re-creation of the sound of a telephone.

While it can be tempting to use EQ to fix any problem you have with a signal, it should only be used when it's really needed, and you should only use as much as you're sure you need. Too much EQ can make a signal sound worse: too harsh, bright or bass-y, depending on what you do with it. It can also make signals sound artificial, a bad thing unless you're specifically going for that effect.

If you're interested in a natural sound, and can move a mic to make a signal sound better, do so. EQ is a poor substitute for good mic placement.

The Mechanics of Using EQ

The channel strips of most mixers contain a set of EQ modules, with each dedicated to the control of a specific range, or "band," of frequencies. These modules all function in a similar manner. First of all, most mixers also provide a switch with which you can turn a channel's EQ modules on or off, individually and/or together. Most mixers provide at least two basic EQ controls:

• Frequency—the control with which you select the frequency you want to boost or cut.

• Gain, or level—the control with which you boost or cut the selected frequency.

On some mixers, you can only choose from among a set of frequencies selected by the mixer's manufacturer. The manufacturer will have selected those frequencies it judges to be most commonly useful. Typically in such EQ, when you choose a frequency, you're actually targeting a band of frequencies above and below the frequency you've selected. That way—given that you're using one of the manufacturer's frequency selections—when you apply a boost or cut, you're more likely to get to the frequencies you really want.

Parametric EQ, on the other hand, is far more powerful, since it lets you select the frequency you'll boost or cut.

In addition, parametric EQ often allows you to set the width of the band of frequencies above and below the selected frequency using a "bandwidth," or "Q," control. Any boost or cut you apply will affect all of the frequencies in this range. You can set exactly how high above and below the selected frequency range extends. Wider bandwidths allow you to select a frequency with less attention to precision, and can also make your boost or cut sound smoother. Narrower bandwidths let you really zero in on just the frequency you want to change.
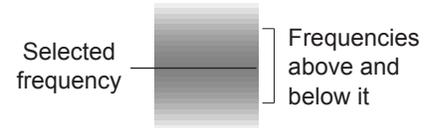
Notching and Shelving

EQ modules can sometimes be configured to affect the selected frequencies in any of three ways, each of which may be represented by a symbol that shows how it works.

Notching boosts or cuts the selected frequency and a range of frequencies above and below it.

Low shelving affects the selected frequency and all of the frequencies beneath it.

High shelving affects the selected frequency and all of the frequencies above it.

The process of using a channel strip EQ module, therefore, has a few general steps:

1. Mentally identify the part of the signal you want to boost or cut.
2. Select the desired frequency.
3. If you're using a parametric EQ, set the desired bandwidth, or choose the form of shelving you want to use.
4. Boost or cut the selected frequency or frequencies.

Some mixers provide a special form of EQ called a "high-pass filter." As its name suggests, a high-pass filter (HPF) allows a signal's higher frequencies to be heard—or "pass"—while silencing its lower frequencies. An HPF can be handy for reducing low-end (as in "low-frequency") rumble or 60 Hz electrical hum from a signal. If a channel strip provides an HPF, you'll be able to set its cutoff frequency—all frequencies above the cutoff frequency will remain audible, while those below it will be silenced.

How Do I Know Which Frequencies I Want?

The frequencies that comprise any particular sound depend to some degree on the mic technique you're using and the quality of the mic, not to mention the quality of the instrument itself (as with our old guitar strings).

Here's a general idea of the typical frequency ranges of some common signals.

Bass Drum	Snare Drum	Cymbal
12 kHz	15 kHz	17 kHz
3 kHz — Skin/flap	9 kHz — Sizzle	11 kHz — Sizzle
1 kHz	Snares	
	1 kHz	4 kHz
80 Hz	300 Hz	1 kHz — Stick tip
	Fatness	
30 Hz — Thump	50 Hz	300 Hz

Electric Bass	Electric Guitar	Acoustic Guitar
10 kHz	10 kHz	13 kHz
6 kHz — Slap	6 kHz	10 kHz — Sparkle
		Pick
1 kHz — Pick	1 kHz — Pick	1 kHz
100 Hz	500 Hz	400 Hz
20 Hz	60 Hz	90 Hz

Electric Piano	Acoustic Piano	Voice
10 kHz	12 kHz	14 kHz — Breathiness
6 kHz — Tines	7 kHz — Hammers	11 kHz
		7 kHz — Sibilance
1 kHz	1 kHz	1 kHz
500 Hz	300 Hz	
	Warmth	200 Hz
30 Hz	70 Hz	100 Hz

If your channel strips have at least one parametric EQ module, here's the best way to identify—and boost or cut—a particular element of a sound.

1. Turn off all of your other signals so that you can hear only the signal whose frequencies you want to investigate.

2. Send signal into the channel strip—play the instrument, run the tape, etc.

3. You'll want to use only one EQ module—any module whose frequency range extends from low frequencies to high can be used. Make sure that none of the strip's other modules are boosting or cutting any frequencies.

4. Decide which element within the sound you want to pinpoint.

5. Turn the module's gain or level setting to a high value. On some mixers, you can turn the gain all the way up; on others, this may be painfully loud. Set it to the most extreme setting you can comfortably tolerate.

6. Turn the module's frequency selector all the way down.

7. Listen carefully as you slowly turn up the frequency selector. As it sweeps through the signal's frequencies, you'll hear each frequency at an exaggerated volume to help you identify it. When it catches the element you're looking for, stop.

8. Reset the module's gain to 0.

9. Slowly raise or lower the gain value until you hear the change you're looking for.

Don't Overdo It!

It's very easy to boost or cut frequencies by too great an amount—the result can be a sound that's too harsh or too dull. To avoid this, use the following method: When boosting or cutting a frequency, raise or lower your EQ gain until you hear the change you want to make, and then back off a step or two. Once you get a little psychological distance, you'll find that you've made the change you wanted without overdoing it.

Some Random EQ Tips

Here's an assortment of EQ tips:

*   If a signal sounds lifeless or muffled, try adding some high frequencies above those you hear in the signal. This can add sparkle, or "air" around an instrument or voice. The reason this works is that instruments and voices contain overtones, high frequency sound waves that are just barely audible, or are felt more than heard. Boosting these "way-highs" can add a surprising amount of realism. This technique is especially effective on vocals and cymbals.

*   On the other end of the spectrum, some instruments such as acoustic guitar have some very low frequencies—around 30-50 Hz—that add little or nothing to their sound, but which can muddy up a mix. It's a good idea to cut, or "roll off," some of this unneeded stuff.

*   It can be too easy to over-concentrate on the highs and lows in a signal, forgetting its important middle. The frequencies around 1 kHz affect how "up-front" a signal sounds. With a boost in this area, the signal can sound as if it's stepped out in front of your speakers, a useful thing when working on a lead vocal, for example. Likewise, you can push a signal

away from the listener by cutting it around 1 kHz . This can be handy when, say, you want one vocal to be more prominent than another vocal.

- When EQ-ing microphones while mixing live sound, watch out when you boost frequencies from 5 kHz to 10 kHz—these can easily lead to squealing feedback. Boost them gradually!

- When attempting to boost or cut an element of a sound, listen to see if some other element shares its frequency range, since it, too, will be affected by what you do. For example, if a hi-hat and snare drum share the same mic, and you raise the volume of the snare drum's snares (the rattles on the underside of the drum) with EQ, you'll also be raising the level of the sound of the drumstick hitting the hi-hat, since that's made up of the same frequencies as the snares. A better solution would be to move your mic closer to the snares.

- Be careful when boosting the very low frequencies—below 75 Hz—of a bass, bass drum or electric piano. While these frequencies do add warmth, they can also add what's not-so-fondly referred to as "mush"—a mix where there's no clarity in your low-frequency instruments. Try boosting basses around 150 Hz, and bass drums around 100 Hz first.

- When you've got several signals in the same octave, you can add clarity to your mix by EQ-ing them out of each other's way. Figure out what elements in each signal identify its sound, and feel free to reduce the volume of frequencies that don't directly affect those elements and that overlap the frequencies of the competing signals.

- When mixing recordings, get to know your speakers so that you can be sure of what you're doing when applying EQ. Each speaker emphasizes frequencies in its own way, depending on its construction and the characteristics of the room in which you're listening. This can be disastrously misleading. For example, if your speakers lack bass, you may add too much low-frequency EQ to your mix to compensate, only to find that your mix sounds muffled on other systems. If your mixer lets you tailor the frequency response of your speakers to your room, take advantage of this feature. If you've got more than one set of monitors, switch between them from time to time for reference. Depending on your Roland digital mixer, if you're using Roland digital speakers, you may be able to test your mix on a variety of classic studio-reference and consumer monitors using Roland's COSM speaker modeling. It's also a good idea to make copies of your work and listen to it on other systems.

Dynamics Processing

Channel strips in digital mixers often provide dynamics processing that helps you control or enhance natural fluctuations in a signal's level. There are three types of dynamic-processing tools you may find in a mixer: compressor/limiters, gates and expanders.

Compressors and Limiters

The device most commonly used for controlling dynamics is a compressor or limiter (we'll explain the difference later). A compressor/limiter reduces the difference between the loudest part of a performance and the softest. There are basically two reasons you'd want to do this:

- First, the dynamic range of some performances can be unmanageably wide: if the loud parts are too much louder than the soft parts, you'd have to record them at an unacceptably low

level so that distortion doesn't occur at the loudest parts; if you're mixing, the quiet parts may disappear in the overall mix.

- The second reason you might want to use a compressor is that it can make a performance sound tighter and somehow more "record-like" by making its dynamics more constant.

The way that a compressor/limiter works is that it lowers—that is, "compresses"—the volume of any signal that exceeds a specified level called a

Threshold

Will be compressed

Won't be compressed

"threshold." You can set a threshold to a high value so that only the loudest parts of the signal will be compressed. You can also set it to a lower level—in this case, most of the signal will exceed it and will be subjected to compression/limiting.
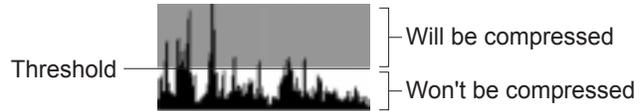
Compression Ratios

You control the amount by which a compressor/limiter reduces a signal's levels by selecting a compression ratio. A compression ratio is a mathematical formula—numerator:1—that means "For every [selected number of dBs] that the signal goes over the threshold, the compressor will reduce the level of the signal so that it goes only 1 dB above the threshold."

For example, with a 4:1 compression ratio, each time a signal tries to go 4 dB above the specified threshold, the compressor will reduce its volume so that it gets only 1 dB over the threshold—it'll bring the signal's volume down by 3 dB. If the ratio is, for example, 10:1, and the signal exceeds the threshold by 10 dB, the compressor/limiter will be bringing the signal's volume down by a much greater amount.

Therefore, the ratio controls how extreme the effect of the compressor/limiter will be. A 4:1 compression ratio gets you a quarter of the level of any peaks that exceed the threshold, a 10:1 ratio will give you a tenth of their levels, and so on.

The logic of compression ratios can be stated simply: to reduce the level of a signal gently—by a small amount—use a lower ratio. To squash it, use a higher ratio.

If you use too high a ratio, you'll hear the compressor doing its job a bit too obviously; this is called "pumping," and is usually not desirable.

Here are a couple of compressor/limiter ideas:

- To fix a signal that's basically good, but that has a few too-loud peaks, use a high threshold that catches only those peaks, and a compression ratio of about 4:1.
- To compress a vocalist, try a low threshold setting with a compression ratio of 2:1. Slowly try out higher compression ratios—don't go higher than 10:1— to tighten up the vocal until you hear the compressor pumping in between phrases, and then back off a bit.
- To make a bass sound smoother, or to tighten up a snare, try a low-to-middle threshold setting and a compression ratio of 4:1.
- To add sustain to a lead guitar, try a low threshold and a ratio of 8:1.

What's the Difference Between a Compressor and a Limiter?

A compressor and limiter are basically the same thing, except that a compressor uses a ratio of less than 10:1 and a limiter uses 10:1 or higher. With its higher ratio, a limiter's threshold is like a brick wall that the signal will just not be able to exceed. You can think of a compressor's threshold as being more like a wall of sponge, in that it's not so absolutely ungiving.

Obviously, a compressor/limiter takes some time to measure the signal and reduce its level. Some mixers allow you to set a compressor/limiter's attack speed, which determines how long an above-the-threshold signal will remain at its original volume before the compressor/limiter seizes control of it. When you want to catch and compress quick loud peaks in level, try using shorter attack time values. If you're only interested in audio that remains above the threshold for a longer period—while letting quick, brief peaks escape—set the attack time to a longer value.

You may also be able to adjust a compressor/limiter's release time, the time it takes for the compressor/limiter to stop the compression it applies to a signal that exceeds its threshold. When a signal's above-the-threshold peaks come quickly, one after another, you may get a more natural sound by lengthening the compressor's release time, so it doesn't switch on and off audibly as each peak comes and goes. If you only have a few peaks here and there that you need to catch, set the release time to a shorter value.

Once the signal's louder parts are under control, you can adjust its overall gain using the compressor/limiter's output control.

Gates

Gates are devices that turn a signal down when its volume falls below a specified level. As its name implies, a gate opens and closes—like a garden gate—to let signal pass through, or not.

When a gate's closed...
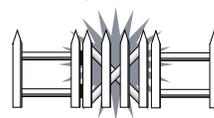
When a gate's open...

...signal can't get through

...signal can pass through

Most commonly, when a gate is closed, it shuts off the signal altogether. Some gates allow you to merely reduce the signal's level when it closes to help you duck unwanted audio without going to the extreme of silencing it altogether.

Gates are very handy for reducing unwanted noise in a signal. For example, you can gate a bass drum signal so that it will fall completely silent whenever the bass drum isn't actually playing—this will prevent other sounds in the kit from being recorded on your bass drum track, making it nice and clean. Similarly, you can gate an electric guitar signal so that when notes aren't being played, the gate will close and shut out any annoying hum.

A gate can also be used as a special effect in which it shuts so fast after a note is played or sung that it chops off the signal's room ambience or reverb, making it fall suddenly away to silence in between notes as a special effect.

Gates typically have five basic controls.

While each has its purpose, you may find that you have to juggle their settings to achieve the desired result when gating signals in which real musicians or singers perform:

- Threshold—A gate's threshold sets the level at which a signal's level will force the gate to shut, reducing the signal's volume. If the signal falls below the threshold level, the gate will close.

- Range—The range setting determines just how much a closed gate will bring down the level of the signal. Setting a gate's range to 0 dB will make the gate completely shut off the signal when the gate closes. You can set it to a higher value to merely reduce the signal's level.

- Attack—A gate's attack time sets how quickly a gate will re-open when a signal exceeding the threshold level occurs. While it may seem that you'd always want the gate to open as fast as it can, there may be times when you'll want to slow down its response time so that it doesn't open for brief peaks in the signal's level that aren't really worth opening the gate.

- Hold—The hold time sets how long the gate will remain wide open when it encounters a signal that exceeds the threshold level. Set the hold time to a value that represents the average amount of time that the wanted audio will last. This setting works hand-in hand with the release time.

- Release—A gate's release time sets how long it will take the gate to close when the signal has fallen below the threshold level and the hold time has ended. The release time can sharpen or soften the sound of the gate closing. When it's set to a short value, the gate will clamp shut hard when the hold time ends, cutting off abruptly. When it's set to a longer value, the gate will close more gradually, in essence fading shut.

Since gating operates by reading the level changes in a signal, signals that are steady in volume—such as sequenced MIDI parts—are easier to gate than those that contain dynamics, such as those produced in a performance by a musician or singer. In either case, a gate's settings interact: when you change one gate setting, you may well have to adjust its partners until the gate is acting as you intend. As you adjust a gate's settings, be patient, and let your ears guide you. The things you want to listen for are:

- If the gate doesn't seem to be working, lower the threshold setting until you hear it start to open and close.

- Make sure that the gate isn't opening—letting audio pass through—when it shouldn't. If this is occurring, raise the threshold setting so that a louder signal is required to open the gate.

- Make sure that the front of the audio you want to hear is not being chopped off as a gate opens too slowly. If this occurs, try lowering the threshold or shortening the attack time.

- Make sure that the gate stays open for just as long as you want it to. If the gate is staying open too long, shorten the hold time (after doing this you may have to lengthen the release time). If it's cutting off the back of wanted audio by a large amount, lengthen the hold time; if it's only chopping off a little bit, lengthen the release time.

- If the gate is shutting too abruptly, lengthen the release time (you may have to shorten the hold time a little to compensate for the change).

Expanders

Expanders do basically the opposite of compressor/limiters: they increase the dynamic range of a performance by exaggerating the differences between its loudest and softest parts. An expander can sometimes add a touch of excitement to a dull performance.

An expander works by reducing the level of signal that falls below a specified threshold, increasing the difference in volume between that signal and any signal that's louder than the threshold level. An expander is thus sort of a cross between a compressor/limiter and a gate.

If your channel strips offer expansion, you'll find the same settings you'll encounter in a compressor, though what they do is a little different:

- Threshold—In an expander, the threshold sets a level below which a signal will be attenuated. Therefore, if signal falls below the threshold level, the expander will push its volume down even further. If a signal is above the threshold level, the expander will allow it to pass unchanged.

- Ratio—The ratio setting determines how much the expander will reduce the volume of signal falling below the threshold level, using the same logic as we found in compressors.

- Attack—An expander's attack time operates the same way that a compressor's attack time does.

- Release—Here, too, an expander's release time operates in the same manner as a compressor's.

Mute and Solo
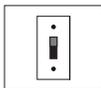
A channel strip's mute and solo features allow you to turn the strip on and off, or isolate the strip for more careful listening by silencing all of your other channel strips.

Mute

A channel strip mute button or software switch provides the most basic channel strip function of all: when you mute a channel strip, you silence the signal passing through it. This provides a handy way to stop a signal without having to go through the trouble of physically disconnecting it from your mixer.

When its channel strip is muted, a signal will be completely turned off—it won't feed any busses to which it's routed, and you won't hear it. When you un-mute the strip, the signal will reappear, and will once again flow into any busses to which you've assigned it.

There are a few reasons to mute a channel. Perhaps:

- you want to turn it off temporarily while you're preparing for a recording or performance so that you can hear some other channel strip's signal.

- the signal is a mic that's allowing performers to talk to you only during rehearsal. During performance, you'll want to silence the mic.

- the signal contains sections of a musical arrangement that you now realize you don't want.

If your mixer offers automation, it's likely that it can capture and play back your channel mutings and un-mutings.

A digital mixer may allow you to assign a set of channel strips to a "mute group." A mute group is a group of channels that can all be muted at once with a single button-press. This can be very handy when you want to quickly silence a group of related signals—for example, a set of vocal mics that you want to silence during a particular part of a song.

Solo

Soloing is a feature that temporarily silences all other signals so that the channel is heard by itself. This allows you to listen to the channel's signal without the distraction of any other signals. This can be helpful when you think you hear a problem in the signal that you need to track down. It can be useful any time you want to work with a signal by listening to it un-obstructed by other signals.

There are a couple of important differences between soloing a channel and simply muting all of the other channels you don't need to hear:

- First of all, it's a much easier and quicker thing to do.
- Second—and more importantly—soloing, unlike muting, doesn't actually turn any channel signals off: it just silences them in your monitors for listening purposes. The silenced channel's signals will continue to flow into the busses to which they've been assigned. This allows you to solo channels in your control room monitors without interrupting any signals being fed to a recorder or stage speaker system—nobody outside of the control room will even be aware that you've soloed anything.

One obvious use of soloing is to isolate an instrument when you want to EQ it. Be careful about EQ-ing a signal when it's soloed. While soloing allows you to hear a channel clearly, remember that your EQ will ultimately have to work along with the mix's other instruments. When you've finished making any changes to a soloed signal's EQ, un-solo it and listen to the signal in context with all of the other signals in the mix to make sure you've got its EQ right.

Some mixers provide a way to solo more than a single channel at a time so that you can isolate related groups of signals.

Level Control
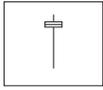
On almost all mixers, each channel provides a fader for the precise control of its signal's level. A fader is a physical device that slides forward, away from you, to increase the level of the channel's signal or back to decrease it. The forward direction is typically referred to as "up," while the other is referred to as "down." A fader derives its name from the idea that you can use it to cause a signal's level to fade away to silence—this is called a "fade-out." In fact, however, a fader is used just as often to make a signal louder.

Signal level goes up

Signal level goes down
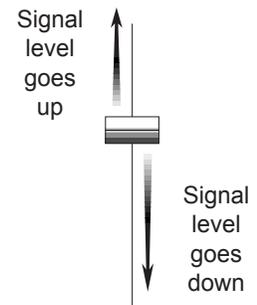
Unlike the trim control we described on Page 29—which sets the level of the signal as it enters the channel strip—a channel fader can control the level of the channel's signal as it flows out of the strip into the main mix busses. On a traditional mixer with standard and Aux busses, the fader will typically control the level of the channel's signal as it goes into a standard bus. A fader can also control the level of any signal being sent to an Aux bus or FlexBus set to receive signal "post-fader."

Your mixer may offer any of a variety of devices that provide you with visual cues that help you keep track of where you are as you move a fader up and down. This can be vital as you experiment with different levels for a channel's signal, and as you move the fader up and down during a recording or mix.

On some mixers, you'll see horizontal lines printed on the mixer's face along the fader's up/down track—these are called "hash marks." You may also see numbers printed alongside the horizontal lines. These show the amount of volume that the fader's adding or subtracting from the channel's signal—a "0 dB" marking may show the position at which the fader is neither boosting or cutting its signal. You may also be able to feel a notch as the fader slides to this position—this notch is called a "detent."

Flying Faders

On some mixers that offer automation, the channel strip faders contain small motors that actually move the fader up and down as the automation is played back so that you can see the changes being made for you. This type of fader is often referred to as a "flying fader."

Inserts

As we first mentioned on Page 12, a channel insert is a point in the channel's signal flow from which you can interrupt the signal, send it to an internal or external effect, and then back into the channel strip to continue its journey. The output of this in-line effect replaces the strip's original un-affected signal. A digital mixer may provide one or more insert points in a channel strip.

Your mixer may allow you to choose the spot in the channel strip from which you want to divert its signal to an insert effect. Typical possibilities for an insert point might be before the channel's delay component (this would be referred to as "pre-delay"), or perhaps just after the delay, right before its EQ module ("pre-EQ"). In either case, after its detour through the selected insert effect, the newly effected signal would be returned to the channel strip at the same place from which it originally came.

On a digital mixer, each insert point on a channel strip may have its own set of controls that allow you to select the internal effect to which the signal will be routed, or to route it to one of the mixer's output jacks. You may also be able to edit the selected insert effect, customizing it to suit your current purposes.
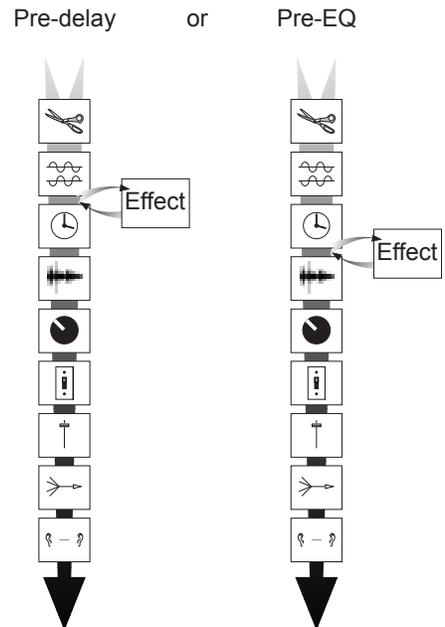
Inserts Beyond the Channel Strips

While not part of our discussion of channel strips, it's interesting to note that some mixers provide insert points for their busses, allowing you to process a bus—that is, all of the signals you've assigned to the bus—through an in-line effect as a single entity. This can be handy, for example, when you want to compress your main mix busses on their way out of the mixer, or run a group of signals through an effect as they're being bussed to a track on a multi-track recorder.

Bus Controls
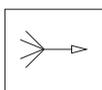
A mixer's bus controls allow you to direct the channel's signal to some other place, which is, after all, the point of bringing the channel's signal into the mixer in the first place. Depending on your mixer, a strip may allow you to route its signal to Aux and/or standard busses, or to one or more of Roland's multi-purpose FlexBusses. You'll also be able to assign the channel's signal to your mixer's main mix busses when you want to create a stereo mix.

Remember that when sending a channel's signal to any bus, your mixer sends a copy of the channel's signal—it doesn't interrupt the signal as it flows through the channel strip. This lets you assign the channel's signal to as many busses as you want from different places in the strip. Perhaps, for example, you'll want to send a signal to an internal effect whose output will be bussed into a mix. Later on down its strip, you may send the signal itself to the mix.

In a mixer with Aux busses, you can use a channel strip's Aux send level controls to send the desired amount, if any, of the channel's signal to each Aux bus. Your mixer may allow you to select the point within the strip from which you'd like to send the copy of the signal, such as, for example, pre-delay, pre-EQ or post-EQ (after the EQ strip's component). In this type of mixer, you can typically assign a channel's signal to a regular bus using a simple on/off switch.

The way in which you control the amount of signal sent to a bus is determined by the point in the strip from which the signal is being copied. If you're sending the signal to the bus post-fader in a traditional mixer with standard busses, the channel's fader will control the amount of signal sent to the bus—in a Roland digital mixer with FlexBusses, the channel's FlexBus send level parameter sets the amount of signal sent. If you're copying the signal from some other point in the strip, the send level will be controlled by the last component the signal passed through before being sent to the bus.

In a Roland mixer with FlexBusses, you can use any FlexBus to send a copy of the channel's signal to any place you like—each bus has all the capabilities required for its use as an Aux or standard bus. In addition to being able to assign the signal to the FlexBus, you'll also be able to select the point in the channel strip from which the bus's copy of the signal will be taken, and be able to independently set the copy's volume as it goes into the FlexBus.

One Bus, Two Bus

Each bus is a separate mono pathway. Your mixer may allow you to link two of them together into a stereo pair. This can be handy when recording a pair of tracks on a multi-track recorder. When you mix, your main mix busses are actually two mono busses linked together as a stereo pair. Once busses are linked, you can position a signal as you wish between the two busses by sending more of the signal to one side than the other using pan or level controls.

Some mixers offer pairs of mono Aux busses permanently linked together as stereo busses—these will typically have their own stereo balance controls.

Your channel strips may also provide controls for Cue busses that you can use to send signal to performers, allowing them to hear what they're doing.

Just a reminder: if you want to send a single channel's signal to its own destination—such as a track on a multi-track recorder—without any other signals, you may not need to use a bus. Your mixer may allow you to use a "direct out" to route the channel strip's output straight to an output jack connected to the desired destination.

Pan Control
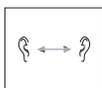
When you're mixing—with all of your channel strips assigned to the main mix busses—you can use the channel strip pan controls to place each channel's signal in the desired position within the overall stereo mix. Pan controls are really level controls—as you pan the signal to one side, you're raising the level of the signal in that bus, and lowering it in the other. On a traditional mixer with separate standard and Aux busses, you can use the channel strip's pan control to balance a signal between two standard busses. (On a Roland mixer with FlexBusses, you can use each FlexBus's level control for this purpose.)

7—Ready, Set Mix

Your Mixer's Manual

You're now ready to go through your mixer's manual to learn how it handles all of the concepts and tools we've discussed. Your mixer may use different terms for some of the things you've just read about, but by applying what you now know about a mixer's underlying structure, you'll be able to identify—and make the most of—the way your mixer does things.

You'll also find that your mixer may offer features not discussed in this book. All of these will build upon the things you've learned. Discovering the clever ways that each mixer goes about its business—and the unique features it offers—is one of the great pleasures of mixing.

This chapter contains additional information you'll want to keep in mind as you begin mixing.

Some Survival Tips

Don't Panic

As you approach your first recording or stage session using your mixer, it may seem that there's an overwhelming amount of stuff to connect and tend to. If you approach things in an orderly manner, you'll find that the whole thing will get done. Here are some ideas to see you through:

• When you set up, you're once again doing one thing over and over. If you take care of one signal at a time—and don't jump around from task to task—you'll have it all set up before you know it. You may find that the performers with whom you're working will interrupt you from time to time. Be polite, but stick with your one-thing-at-a-time method as best you can.

• Before you start setting up, make notes on a sheet of paper that list all of the things you've got to connect, and where you want their signals to wind up. Use these notes as a roadmap, checking off each item as you complete it. If you get distracted during setup by something that comes up, the notes will help you get back on track.

• Turn down your control-room and studio monitors and headphone amplifier, or stage-speaker and stage-monitor amplifier for safety as you make your connections.

• Connect all of your mics, instruments and other cables to your mixing processor or mixer. Once that's done, you can get out of your performers' way and move over to the mixer.

When Panic Strikes

You bring up your first fader to listen to its signal and hear—nothing. This has happened to everyone who uses a mixer at least a few times. If it happens to you, don't worry. Most likely your hardware is fine. What's probably going on is that the signal is getting waylaid somewhere on its way between its source and your ears. Here's what to do:

• We've repeatedly mentioned the importance of doing things in a logical order. This is the moment when you most need to heed that advice. Patiently follow the signal's path from its source into a jack, through a channel strip and on and to the desired destination. Most likely, you'll find that you've forgotten some step along the way. If this doesn't solve your problem, repeat the process.

• Make sure your monitor or stage speaker system is turned up . You'd be amazed how many times this obvious item is overlooked.

Feedback

Feedback occurs when a signal receives its own output. If you're suddenly engulfed in a screeching or howling electronic noise, you'll need to act fast. If you're recording, quickly turn down your monitors and your performers' headphones or monitors. Since it may be something you've just done that's causing the problem, undo the last thing or two you did until the screeching goes away. If it's a routing problem, you may see an overload or clipping indicator on your mixer that'll pinpoint the problem signal for you. Correct the signal's input or output routing.

Setting Basic Channel Input Levels

You'll want to keep each channel fader somewhere around its middle up/down position to allow yourself room to move in either direction, with more room on the bottom in case you need to do a fade-out. In order to do this you'll need to adjust the channel's input trim setting:

1. Have your performers make some noise to give you a sample of their sound as you perform the following two steps.

2. Position the fader about three quarters of the way up. Your mixer may show some form of mark printed beneath the fader in this area—there may even be a detent at this position.

3. Adjust the channel's input trim so that the signal is at roughly the volume you desire without causing the trim's overload indicator to light.

Avoid turning the trim up significantly above its middle position—this can cost you signal clarity. In such a case, increase the signal at its source, or move your fader up.

Some manufacturers refer to this process as "setting unity gain."

Application: A Multi-Track Recording Session

The recording of a song on a multi-track recorder typically involves several processes:

- You'll start by recording the song's basic tracks. Basic tracks can be anything from an entire band playing to a single guide track of a percussion instrument. Typically, a song's basic tracks are comprised of as many instruments as your mics, cables, MIDI channels or mixer channels allow. You can record each performer on his/her own track, or combined with other performers playing related parts—a group of horn players might be one such case. You can also divide complex instruments such as drum kits into individual tracks that allow you to control each component of the instrument separately.

- Once the song's basic tracks have been recorded, you can record the rest of its tracks. This process is commonly called "multi-tracking" or "overdubbing," a term originally coined for a slightly different procedure, as we'll see later.

  While a recorded guide lead vocal can be helpful, it's a good idea to record your final vocals as overdubs to avoid leakage from instruments in your vocal tracks.

- A multi-track recorder will allow you to fix a performance error by re-recording only the portion of a track that contains the mistake, leaving the rest of the track intact. This is called "punching"—when you begin to re-record the section, you're "punching in," and when you stop re-recording the section, you're "punching out."

- If you run out of tracks on your multi-track recorder, or if you'd rather reduce a group of tracks to single track or pair of tracks to make them easier to manage, you can combine tracks by "bouncing" or "ping-ponging" them together.

Use Your Recording Time Wisely

When recording, setting up can take as much time, or more, than the actual recording. Try to record all of your songs' basic tracks one after another, and then move on to your overdubs, punches and bounces. This saves studio time and conserves everyone's energy since performers won't have to wait around as you do and re-do your mixer setups and move mics and cables around and will save you a lot of repetitive work. If jumping back and forth is unavoidable, your Roland mixer may be able to save each song's recording, overdubbing and/or bouncing setups as scenes that you can recall quickly, keeping your mixer setup time at a bare minimum.

In the following sections, we'll refer to the recording studio and control room. If you're working in an environment without separate rooms, think of "studio" as meaning "the recording area," and "the control room" as being the place you have your mixer.

Before recording, make sure that you've properly connected your multi-track recorder's inputs and outputs to the appropriate outputs and inputs on your mixer. If you're going to be sending a headphone feed to your performers, you'll also need to make sure you've connected any Aux, FlexBus or assignable output on your mixer that you intend to use for this purpose to the headphone amplifier to which their headphones are connected.

Setting Up In the Studio

- When you're working with microphones, try to arrange your performers in the studio so that the best sound can be achieved. Try to isolate the sound of each performer as much as possible so that each of your mics picks up only the intended instrument or singer. Unwanted noise from other performers in a mic is called "leakage," and can make getting a clear recording more difficult. You can avoid leakage by placing louder instruments as far away as possible from other mics, or by surrounding them with acoustic separators—also called "baffles," or "gobos." Of course, it's important that your performers be comfortable and able to communicate, so be diplomatic, and be ready to compromise if you have to.

- When working with mics and electric or electronic instruments, encourage their musicians to listen to the instruments through headphones connected to the mixer instead of through amplifiers in the studio. This will prevent their sounds from leaking into your mics.

  If you're recording without mics—in an all-MIDI situation, for example—your performers can be anywhere that's convenient—even right next to you in the control room, if you like—since mic leakage won't be an issue.

- Consider setting up a mic in the studio through which performers can talk to you. This can make the session go more smoothly. You needn't actually record this mic's signal.

- Once you've set up all of your mics and cables, give each musician in the studio a pair of headphones through which they can hear what's being performed and recorded.

Preparing the Mixer for Recording

- If your digital mixer provides input patching, route each of the inputs you used in the studio to the desired channel on your mixer.

- On each channel's writable strip—or using the channel-naming parameter your Roland digital mixer may provide—name the strip with name of the signal it's controlling.

- If your digital mixer has multiple operating modes, select the mode that allows you to adjust channel settings.

- Select the desired source for each channel's input signal.

- Assign all of your channel strips to the main mix, and set your monitors to listen to the main mix. Turn up your studio monitors—you turned them down while setting up, didn't you?—and move from strip to strip, raising each fader a bit to verify that its signal is there. If you're dealing with mics, you'll be able to hear if they work by listening to the studio. If your signals are coming from keyboards or electric instruments—or CDs or turntables—you'll have to get them to make some noise so that you can verify your connections are okay.

- With the exception of the mic your performers use for talking to you, remove all of the channel strips from the main mix and route each to the desired track on your multi-track recorder—you can do this by assigning it to a bus or FlexBus, or by using a direct out.

- Make sure the tracks you're using on your multi-track recorder are record-enabled, and that the multi-track recorder is set to allow you to listen to its inputs so that you can catch any problems quickly. Some people prefer to hear the just-recorded tracks by listening to the recorder's outputs to ensure that they've been properly recorded—with some recorders, however, these signals are delayed slightly, making it hard to catch and fix problems quickly.

- In the control room, turn up the tape-return level of each of the tracks you'll be recording so that you can hear it (if your mixer has MULTI inputs, you'll need to temporarily switch modes to adjust their levels). These levels are for monitoring purposes only—they have no effect on the level of signal recorded on your tracks—so set them however you like.

- Select a Cue bus, Aux bus or FlexBus to use as a headphone feed for your performers and send signal from each channel you want them to hear to the selected Cue bus, Aux bus or FlexBus. Send the signal from as early a point in the channel's signal flow—e.g. pre-fader or pre-EQ—as possible so that your performers won't be distracted by any changes you make to their signals' levels or sound. Be prepared to adjust the send level of each channel as you perfect the headphone mix in accordance with your performers' wishes.

- Set the basic input level for each channel using the method described on Page 46.

- Use each channel's tools to make its signal sound the way you want it to. Apply EQ, add dynamics processing and so on.

  Some people prefer to record without EQ if they're concerned about making mistakes during the session. They apply EQ later on, during mixing.

- Set each channel's signal to the optimal recording level. If your mixer has standard and Aux busses, use each channel fader for this purpose. On a Roland digital mixer with FlexBusses, adjust the amount of signal being sent by each channel to its assigned FlexBus. Consult your multi-track recorder's documentation to learn what its preferred levels are and how to interpret its metering, since these are the meters that count most during recording.

- After you've made all of your settings, set your mixer to monitor the headphone feed to your performers and ask them if it sounds okay to them. If the performers request changes, adjust your send levels to the appropriate Cue bus, Aux bus or FlexBus.

  If your performers desire separate headphone mixes—and if you have sufficient headphone amplifiers available—you can set up multiple Cue busses, Aux busses or FlexBusses for this purpose.

About Adding Effects as You Record

While it makes sense to process a signal with in-line effects—such as dynamics processing, distortion or Roland's microphone modeling—as you record, it's debatable whether or not this is a good idea when you want to add reverb, delay, flangers and other such effects.

Pros: If you add an effect to a signal during recording, the effect will be permanently embedded in the track, and you'll never have to worry again about recreating it. Also, it will free up an effect processor for other uses during final mixing.

Cons: You're absolutely stuck with whatever effect you record—you can't change your mind later without completely re-recording the track. Also, if you're using an effect that you want in stereo, as will frequently be the case with reverb, you'll need two tracks on your multi-track recorder to capture it. If the original signal is mono, this will increase the number of tracks you need.

If you have enough tracks, you can get the benefits of both approaches by recording your dry signals and effects on separate tracks.

Bottom line? It's a matter of personal preference and the needs of your song.

Recording

- Make sure that any required tape or other media is loaded properly in your multi-track recorder, and set the recorder to the location at which you want to begin recording.

- Begin recording by pressing the appropriate transport buttons on the multi-track recorder, or by using your mixer's MMC controls if the recorder supports MMC.

- At the beginning of each take—before the performers start performing—record a slate stating the take's number. If your mixer provides a talkback mic, use it; if not, one of your performers may be able to announce the take. This makes takes easier to identify for playback and mixing.

- During the actual recording, your job is to listen for any problems and generally to make only minor adjustments, if any at all. If you change the level or sound of a track during the recording, you may find mixing to be more difficult because the track won't have a consistent sound. It's better to note the changes you want to make and make them before the next take. If there is no next take, you can make the changes when you mix. An exception: If a signal is clipping, you should remedy the problem immediately.

- If your mixer can record scenes, store the song's final recording settings in case you need to re-record it at some later date.

The First Playback

- Once recording has ended, immediately record-disable all of your tracks to avoid accidental erasure. This will also result in the just-recorded tracks being heard through your tape returns. (When it's time to record again, you'll need to record-enable the tracks.)

- Play back the recording in the control room monitors. If it's desirable to play the recording for your performers, you can play it over the studio monitors, if you have some, or invite them into the control room to listen.

- You can also play the recording for your musicians over their headphones. If your mixer has standard and Aux busses, select your multi-track tape returns as the signal source for all of the channel strips that numerically correspond to the recorded tracks. Make sure that these channels are being sent to any Cue or Aux bus being used as a headphone feed, and adjust their send levels if necessary. Remember to set any channels being used for recording back to their original input sources and headphone send levels before any further recording. If you're using a Roland mixer with MULTI INs, send each MULTI that corresponds to your recorded tracks to any FlexBus being used for headphones.

Fixing Tracks: Punching

Punching can be performed upon basic tracks, or the tracks you overdub later. In either event, there's very little you need to do to set up your mixer for punching:

- Make sure that all of the tracks you don't need to punch are record-disabled.

- Don't change any of your mixer settings! This will ensure that the punched-in portion of the track will match its already recorded parts.

- Consult your multi-track recorder's documentation to learn how to punch in and out.

Adding More Tracks: Overdubbing/Multi-Tracking

• Make sure that all of the tracks you want to keep are not record-enabled.

• Select the channel strip—or strips if you're recording in stereo, or are recording more than one performer—you want to use. If you've finished recording basic tracks, or if you've saved your settings as a scene, you can use any strip; if not, try and find an unused channel so that you don't lose your recording settings.

• Route the channel's signal to the desired track using a bus, FlexBus or direct out.

• Record-enable the desired track on your multi-track recorder.

• Turn up the track's return level on the mixer so that you can hear what you're recording.

• If your performer is using headphones, assign the channel to the *Cue* bus, *Aux* bus, or FlexBus you're using for headphones, and set the desired level.

When a performer is using headphones during overdubbing, he or she will also need to hear the already recorded tracks during overdubbing. Route the necessary multi-track tape returns to the headphone mix.

• Set the multi-track recorder to the desired starting location, and record the overdub.

• When recording is finished, record-disable the newly recorded track.

Combining Tracks: Bouncing/Ping-Ponging

You may decide to combine already recorded tracks because you're running out of empty tracks on your multi-track recorder and need to make room for new tracks, or because you want to combine tracks now to make mixing later on simpler. The process of bouncing tracks involves making a mix of them and recording that mix onto other tracks:

• On a traditional mixer, you'll use the channel strips that numerically correspond to your recorded tracks—depending on the mixer, you can flip the channels or select a tape return as each channel's input source. If you've got a Roland mixer with MULTI INs, patch your tracks to the desired channel strips and select the MULTI In fader control mode.

• Assign all of these channels to a bus or FlexBus routed to the track on which you want the new combination track recorded. (To bounce to two tracks, use two busses or FlexBusses.)

• Record-enable the new track you're bouncing to on your multi-track recorder.

• Turn up the tape return for the track you're bouncing to so that you can hear it.

• Make sure to turn off direct monitoring of the tape returns for the tracks you're bouncing. Failing to do this will mean that you'll be hearing the bounced tracks from two sources simultaneously and will not be able to mix them together accurately.

• Adjust the channel settings of the tracks to be bounced. Set them to their desired levels, EQ and so on. A bounce can include added effects, as well, if you wish. Bouncing doesn't erase the tracks being bounced, so you can do a bounce over and over until you've got it right.

• You can add a new performance to the tracks as you bounce them by bussing a live signal from another channel to the same track as the track you're bouncing to. This is called a "live bounce." Originally, the term "overdubbing" referred to live bouncing.

• Set the multi-track recorder to the desired starting location, and record the bounce.

• When bouncing is completed, record-disable the newly recorded track or tracks.

Application: Mixing a Record

A multi-track recording must eventually be reduced to a final form appropriate to the medium you're using. If you're making a CD, for example, you'll need to create a two-track, stereo mix. Some mixers also support the creation of surround-sound mixes for video games and movie soundtracks. In either case, a mix is where you perfect the sound of your recorded tracks and combine them to create a final product.

Setting Up

You'll need to connect your mixer's main mix bus outputs to the inputs of your mixdown recorder. Your mixer may provide special inputs to which you can connect the recorder's outputs. These inputs can be selected for monitoring, allowing you to listen to the recorder itself when mixing. This is a good idea because it ensures that what you hear is what the mixdown recorder hears, and, thus, what you get.

- On a traditional mixer, you'll use the channel strips that numerically correspond to your recorded tracks—depending on the mixer, you can flip the channels or select a tape return as each channel's input source. If you've got a Roland mixer with MULTI INs, patch your tracks to the desired channel strips and select the MULTI In fader control mode.
- Write the name of each track in the writable area of its strip. If your mixer allows you to assign a name to each channel, do so.
- Assign all of the strips controlling your tracks to the main mix bus.
- Set your monitor to listen to the inputs of your mixdown recorder.

Creating a Mix

The steps below are presented in roughly the order in which they should be performed. However, there's a lot of back-and-forth in mixing since each track's sound is affected by the other tracks in the mix—continual re-adjustments must be made. Expect to re-visit these steps many times as you work your way to a final mix.

- Set the stereo pan position of each channel's signal using the channel's pan control.
- Adjust each track's EQ as necessary. Insert any in-line effects you want to use and tweak the settings of the effects to taste.
- You can solo each track as you refine its sound, but make sure to re-check the track with all of the other tracks playing when you're done.
- To add an effect to a track, send it to an internal or external effect processor by assigning it to an Aux bus or Roland FlexBus routed to the effect processor:
    - If the processor's outputs are connected to Aux returns, assign the Aux returns to the main mix bus. You can also return the outputs of the effect processor to channel strips, and assign the channels to the main mix bus—this is the method you'll always use on a mixer without Aux returns.

- Edit the effect processor's settings: select the desired effect and adjust its settings to suit your needs.
- Using the channel's Aux or FlexBus send level controls, raise or lower the amount of signal sent to the processor until you've got the desired amount of the effect.
- If you're sending more than one channel's signal to the effect processor, you may find the overall level of the effect too loud or soft. While each channel controls the amount of effect applied to its signal, you can set the effect's overall volume by adjusting its Aux or channel return level settings.

- Listen to all of the tracks together, and make any necessary level or sound adjustments.

- Work out and rehearse any changes that need to be made as the song plays.

- Make sure the overall level of your mix is as loud as it can be without clipping the mixer's outputs or the mixdown recorder's inputs. Consult the recorder's documentation to learn what levels it wants, and what its meters mean—they're the meters that count the most when mixing.

  You can run your entire mix through a stereo compressor or some other effect by routing the mix to the effect using the insert points on your Roland digital mixer's main mix busses. This can be a handy way to add just the right final touch to your mix.

- It's a good idea to try out your mix on different systems. If you have more than one set of monitors connected to your mixer, listen to the mix on different speakers to help ensure it'll sound good on any system. If you're using Roland digital monitors, and your Roland digital mixer offers Roland's COSM speaker modeling, you can try out your mix on a variety of different virtual speakers.

- If you have time to make cassette copies of your mix to study on other peoples home and/or car stereos, do it. You want your mix to sound great everywhere.

- When you're ready, record your mix on your mixdown recorder.

Your digital mixer may allow you to record a song's or project's mix settings as a scene. These can help you re-establish the settings at a later date, if necessary. It's also a great way to try out alternate versions of a mix—perhaps, for example, utilizing different approaches to effects—by saving each version as a separate scene.

If your digital mixer allows you to save snapshots, you can save a snapshot for each major settings change in mix, and your mixer will be able to play them back for you at the proper times during the actual recording of your mix.

Your mixer may also be able to capture and play back all of your mix moves during the song, or allow them to be recorded and played back by an external MIDI sequencer.

Check your mixer's documentation to learn more about its various automation capabilities.

Application—Mixing Live Sound

When you're mixing live sound, you're typically responsible for two mixes:

• the mix that the audience hears.

• the monitor mix to which the performers listen as they work.

If your performers require more than one monitor mix, you can create as many monitor mixes as you have busses, outputs and monitor amplification systems.

It's best to perform your mixing tasks from a location within the audience—this is the best way to be sure you're hearing what the audience hears. The optimal location is towards the rear of the room. A central position is best, especially when mixing in stereo.

When you're mixing live sound, your mixer's main outputs—or the main outputs on your mixing processor—should be connected to the main inputs of the house speaker system amplifier. If your stage system is mono, either output will suffice; otherwise, you'll need to use both mix outputs. For monitor mixes, you'll need to connect the outputs of any Cue bus, Aux bus, or Roland FlexBus output you want to use for monitors to a monitor amplifier.

You'll also need a way to connect all of the cables onstage to your mixer:

• In a traditional mixer, this can be accomplished by connecting all of your stage cables to a "snake," a long, bulky cable containing wiring to carry all of your signals back to your mixer.

• In a component-based Roland mixer/mixing processor system, you can place the processor on the stage, and connect your cables to the processor—the processor is connected to the mixer by a pair of thin, easy-to-work-with digital cables.

In either case, take care to run your cabling somewhere it won't get stepped on, bumped, or worse, severed. Running it around one side of the room or overhead will most likely be your best bet.

• There's no substitute for a sound check before the audience arrives. This will allow you to calmly set up, troubleshoot and establish your starting levels without pressure. Try to convince your performers to do a sound check.

• If your mixer provides a white/pink noise generator—we discussed such a generator on Page 20—use it to optimize the frequency response of your stage speakers.

• Assign all of your channel strips to the main mix, and turn up your stage speaker amplifier—you turned it down while setting up, didn't you? Move from strip to strip, raising each fader a bit to verify that signal you want is there. If you're dealing with mics, you'll be able to hear whether they work by listening to the stage. When your signals are coming from keyboards or electric instruments—or turntables or CDs —you'll have to get them to make some noise so that you can verify your connections are okay.

• Ask the performers to run through some of the material they'll be presenting to the audience at the show. This will allow you to hear what they're going to do and establish your settings as they rehearse.

- Set the basic input level for each channel using the method described on Page 46.

- Turn up your stage monitor amplifier. Select a Cue bus, Aux bus or FlexBus to use as a monitor feed for your performers and send signal from each channel you want them to hear to the selected Cue bus, Aux bus or FlexBus. Send the signal from as early a point in the channel's signal flow as possible so that your performers won't be distracted by any changes you make to their signals' levels or sound. Be prepared to adjust the send level of each channel as you perfect the monitor mix in accordance with your performers' wishes. Adjust the send levels gradually to avoid feedback.

- Set each signal to its desired position in the stereo field, or, if you're mixing in mono, pan all of your signals to the side that feeds the output connected to the stage speaker system.

- Adjust each signal's level and EQ as necessary. Insert any in-line effects you want to use and tweak the settings of the effects to taste.

- To add an effect to a signal, send it to an internal or external effect processor by assigning it to an Aux bus or Roland FlexBus routed to the effect processor:

  - If the processor's outputs are connected to Aux returns, assign the Aux returns to the main mix bus. You can also connect the outputs of the effect processor to channel strips and assign the channels to the main mix bus—this is the method you'll always use on a mixer without Aux returns.

  - Edit the effect processor's settings: select the desired effect and adjust its settings it to suit your needs.

  - Using the channel's Aux or FlexBus send level controls, raise or lower the amount of signal sent to the processor until you've got the desired amount of the effect.

  - If you're sending more than one channel's signal to the effect processor, you may find the overall level of the effect too loud or soft. While each channel controls the amount of effect applied to its signal, you can set the effect's overall volume by adjusting its Aux or channel return settings.

- Listen to all of your signals together, and make any necessary level or sound adjustments.

- If your mixer provides fader grouping, take advantage of the feature to group together signals you may need to adjust simultaneously, such as a bunch of backup vocal mics.

- Make sure the overall level of your mix is as loud as it can be without clipping the mixer's outputs or speaker system's amplifier inputs.

- When you've got all your settings the way you want them, a digital mixer may allow you to save them as a scene in case someone changes any of them in-between the sound check and show. If this happens, you can quickly recall the scene to re-establish your settings.

- During the actual performance, your job is to make adjustments as needed. It's not uncommon for musicians' levels to change over the course of the performance, and you'll need to stay on top of these changes to make sure the overall mix doesn't suffer.

- You may need to make subtle, continuous adjustments to a signal's level during a performance—this is called "riding" a level. While a compressor/limiter can help smooth out level fluctuations in a performance, riding levels may still be necessary when working with performers whose mic or playing dynamics would require an unpleasant amount of compression or limiting. You may also need to ride levels to bring up the volume of vocal or instrumental soloists.

Congratulations!

Well, that's it. You now know everything you need to know to understand the workings of any mixer—you'll find the precise details of your mixer's operation in its documentation. In case you want to review anything, here's a quick review of what we've discussed, and where:

- In Chapter 1 (Page 1), we explained the reason that "Basics of Mixing" exists, and described its symbols and conventions. We also introduced a couple of important, yet comforting, thoughts about the process of mastering your Roland mixer.

- Chapter 2 (Page 3) described in general terms exactly what it is that a mixer does anyway. We also introduced some basic terms such as "mix," "signal," "send" and "return." And we explained why a digital mixer is a great thing to have.

- In Chapter 3 (Page 7), we introduced a host of vital mixing concepts, including, among others, channels, busses, and effects.

- Chapter 4 (Page 13) discusses the sections of a typical mixer and what each one does.

- In Chapter 5 (Page 21), we described all of the types of connectors you'll use to bring audio into and out your mixer, as well as some of the connection issues that may arise.

- Chapter 6 (Page 29) detailed the various tools that each channel strip provides in a typical mixer. It also offers lots of tips on how to master such channel tools as EQ and dynamics processors.

- Chapter 7 (Page 45) provided some practical thoughts and tips as you begin to apply all that you've learned in the rest of "Basics of Mixing."

- The three application sections that follow Chapter 7 walked you through typical multi-track recording (Page 47), record mixing (Page 53) and live sound (Page 55) sessions.

- ...and that brings us here!

The remainder of "Basics of Mixing" is a glossary. As you go through your mixer's documentation, you can consult the glossary to look up any terms whose meaning you've forgotten.

Now that you know what your mixer can do for you, you're well on your way to a wonderful creative experience with this oh-so powerful device at your fingertips. Be patient with yourself—and your mixer—as you master the art of mixing. Stay calm, listen carefully, and have fun.

Glossary

ADC	For "Analog-to-Digital Converter," a device that receives analog audio and converts it into digital data, such as analog audio coming into a digital mixer
AES/EBU	For "Audio Engineering Society/European Broadcast Union," a format for sending and receiving digital audio data; typically uses an XLR connection
Ambience	The resonating of the real or imaginary space in which a sound occurs
Amplitude	The technical term for a signal's volume or loudness
Analog	Voltage-based representations of audio
Analog-to-digital	The conversion, using an ADC, of analog audio to digital data
Attenuate	To reduce in volume
Audio	A general term for a stream of sound
Automation	The memorization and playback of changes you make to mixer settings
Aux	Short for "Auxiliary"; a designation for extra busses typically used for sending signal to effects, headphone amps and other destinations
Aux return	An extra input; typically used for receiving a signal from the output of an internal or external effect processor
Aux send	An extra bus that can be used for sending signal anywhere; typically used for sending signal into an effect
Balanced	A type of audio connection that uses the three leads in a cable, connector and jack as part of a phase-cancellation scheme to boost signal and reduce noise
Band	In EQ, a range of frequencies
Bandwidth	In EQ, the width of a band; the number of frequencies that will boosted or cut above and below a selected center frequency
Bass	The lower frequency range of a sound, usually from about 200 Hz down
Bit depth	The resolution at which level changes are represented in a digital audio signal
Boost	To increase the level of something
Bus	A pathway down which one or more signals can travel
Cannon connector	Another name for an XLR connector
Channel	A set of tools for the control and shaping of a single signal
Channel strip	A row of controls on a mixer allocated to the shaping of a single signal
Clipping	The unpleasant thumping or clicking noise made when a signal exceeds the capacity of an audio device
Compression ratio	The amount of gain reduction that will be applied to a signal exceeding a compressor's threshold level setting
Compressor	A dynamics processor that reduces the level of any signal exceeding a specified threshold volume
Condenser Microphone	A type of high-quality mic that requires power

COSM	An abbreviation for Roland's "Composite Object Sound Modeling" technology that shapes audio by applying the sonic characteristics of popular or classic microphones, guitars, guitar amplifiers and studio reference speakers
Cue Bus	A bus—sometimes a stereo pair of busses—dedicated to the providing of signal to performers so they can hear what they're doing
Cut	To reduce the level of something; typically used in the context of EQ
Cycle	In a sound wave, the repeating movement from the greatest amount of air pressure to the least; the pitch of sound waves is measures in cycles; each cycle equals one "Hertz"
DAC	For "Digital-to-Analog Converter," a device that converts digital data to analog audio, such as the audio leaving a digital mixer on its way to an analog device
DAT	Abbreviation for "Digital Audio Tape"; used in reference to this type of tape as well as the recorders that use it
dB	Abbreviation for "decibel," a unit of measurement for the loudness of audio
De-esser	A device that detects and reduces sibilance in vocal signals
Delay	An effect in which a copy of a signal is played back later than the original
Detent	A notch that you can feel as you move a fader up or down; signifies the point at which no level boost or cut is being applied by the fader
Digital	Audio expressed as binary computer data
Digital-to-analog	The conversion, using a DAC, of digital data to analog audio
Dim	A switch that allows you to quickly reduce your monitoring volume
Direct box	A mic-level box that converts a phone connector to an XLR connector
Distortion	Fuzziness or roughness added to a sound
Doubling	The artificial simulation of a second, unison performance that's created using a delay with a short delay time
Dry	A signal to which an effect has not been added
Dynamics	Expressive volume fluctuations in a signal
Echo	A delay-based effect in which copies of a signal are heard trailing off to silence; similar to shouting from a mountaintop and hearing your voice repeat
Effects	Any of a variety of audio processes that can be applied to a signal to modify it, including reverb, delay, flanging, phasing
Effect Loop	The process of adding an effect to a signal by sending a copy of the signal to an effect, and mixing the effect's output with the original signal
Effect return	An input that receives signal from the output of an internal or external effect
EQ	A popular abbreviation for "equalization," the thing an equalizer does
Equalization	The process of altering the levels of frequencies that comprise a signal
Equalizer	A device that boost or cuts the volume of specific frequencies in a signal
Expander	A device that reduces the level of a signal when it falls below a specified threshold to exaggerate its dynamic range
EZ Routing	A re-usable template containing a Roland digital mixer's routings; in some cases, walks you through the creation of a setup using displayed questions

Fade In	A change in level over time increasing upward from silence
Fade Out	A change in level over time falling gradually to silence
Fader	A slider-type device that's used for the precise manipulation of levels
Feedback	The delaying of a delay so that multiple images of the original signal are heard; also, the loud squeal that's heard when a channel is receiving its own output
Filter	A device that removes specified frequencies from a signal
Flanger	An effect that adds a swirling sound to a signal by adding a slightly delayed copy of the signal in which the copy's delay time fluctuates
FlexBus	A powerful all-purpose bus available on Roland digital mixers
Flip	On an in-line mixer, the act of allocating a channel's tools to the control of an input signal or to the control of a multi-track tape return
Flipping	When working with two signals 180 degrees out of phase, delaying one of the signals so that its phase lines up with the other
Flying fader	A motorized fader that automatically moves during automation playback
Frequency	Refers to the number of times per second that a sound wave's cycle repeats, with a greater frequency resulting in a higher perceived pitch; also used as shorthand for describing sound waves in a signal by their pitch
Gain	Another term for level
Gate	A device that turns audio off or down when it falls below a specified threshold
Graphic equalizer	An equalizer with pre-determined editable frequencies, arranged from left to right to visually depict the EQ shape of a signal
Ground	A common zero voltage reference in a system of connected audio devices; when devices have different zero voltage references ground hum may occur
Grouping	A process by which multiple channel strips are joined together under a single level control
Hash mark	A horizontal line along the path of a fader to help identify its up/down position
Headroom	The number of dBs above the point at which a clipping warning appears before clipping actually occurs
Hertz (Hz)	A unit of measurement equal to a sound wave's single cycle
High-pass filter	A filter that removes lower frequencies from a signal, allowing higher frequencies to pass through unaffected
Hum	An undesirable low-frequency tone present in a signal as a result of grounding problems or proximity to a power source
Impedance	The amount of force with which voltage leaves a connector and the amount of resistance to that force in the jack receiving it; they should be equal
In-line	Any effect accessed by interrupting a channel's signal flow, directing its signal to the effect, and returning the effect's output to the channel at the same point from which it came (also called an "insert effect"); also, a mixer whose input and multi-track tape return controls are contained in each of its channel strips
Input	A jack that receives audio
Input level	The level of signal coming into a channel strip

Insert	A point in a signal flow at which an in-line effect can be employed
Insert effect	A two-way journey of a signal from a channel insert point to an effect and back to the insert point
kHz	for "kiloHertz": a thousand Hertz
LCD	For "Liquid Crystal Diode," a plasma-based display used in most digital mixers
LED	For "Light Emitting Diode," a small, sometimes colored, light bulb
Level	A general term for volume or amplitude
Limiter	A compressor set to a ratio of 10:1 or greater
Line level	The high-level signal produced at the outputs of audio equipment such as mixers, recorders and playback devices
Low-pass filter	A filter that removes higher frequencies from a signal, allowing lower frequencies to pass through unaffected
Macro	A digital mixer shortcut that performs a multi-step operation as a single action
Meter	A visual device that shows the level of a signal
Meter bridge	A separate piece of mixer hardware that provides an additional array of meters
Mic	A common nickname for "microphone"
Mic Level	The low-level signal produced by microphones and electric instruments such as electric guitar or bass
Microphone	A device that converts sound waves into audio signals
MIDI	For "Musical Instrument Digital Interface," the wiring and message protocol that allows musical instruments and other devices to communicate
Mix	As a noun, a signal that contains one or more other signals—typically a mix is a pair of stereo signals that contains numerous mono and stereo signals, along with effects, combined together; as a verb, the act of creating such a combined signal, or of using a mixer in general
Mixdown	A common synonym for the noun "mix"
Mixer	A device in which audio signals can be manipulated, enhanced and directed to other destinations, singly or together; also, someone who works a mixer
Modeling	A process by which the characteristics of one signal are applied to another; Roland's advanced COSM modeling creates realistic emulations of popular and classic microphones, guitars, guitar amplifiers and studio reference speakers
Monitor	As a noun, a speaker, or set of speakers, for the purpose of listening to a mix; as a verb, the act of listening when mixing
Mono	A single signal
MMC	For "MIDI Machine Control," the MIDI-based protocol that allows the controls of one MMC-compliant device to affect the transport mechanism of another
MTC	For "MIDI Time Code," a form of SMPTE used for the timing synchronization of two or more MIDI-compliant devices
Mute	A switch that allows you to silence a channel's signal; some mixers provide mute grouping for silencing multiple channels at once

Notching	A form of EQ in which a selected frequency and a specified number of frequencies above and below it—called a "band"—are affected
Optical connector	A connector that transmits digital data as light using fiber-optic technology
Out of Phase	A situation in which the sound-wave cycles in one signal reach their greatest amount of air pressure as the cycles in a similar signal reach their least; the two signals will cancel each other out
Outboard	External, as in an "external device"
Output	A jack that sends out signal
Overload	What occurs when a signal is so loud that it exceeds the capabilities of the device through which it's passing
Pad	A device that lowers the level of a signal
Parameter	A setting whose value can be changed
Parametric	A type of EQ that can be adjusted so that it can boost or cut any frequency within its overall range; may also have a user-definable bandwidth
Patch	A temporary connection made between two audio devices, or within one
Peak	A sudden high-volume burst of signal
Panning	The left/right positioning of a signal within a stereo image
Phasing	The synchronization—or lack thereof—of the sound waves in two similar signals; also an effect in which a swirling sound is added to a signal by creating a very slightly delayed copy of the signal and in which the copy's delay time fluctuates, similar to flanging
Phase Cancellation	The complete cancellation of audio that occurs when two signals are 180 degrees out of phase
Phantom Power	The power required for the operation of a condenser microphone when it's not supplied by internal batteries or a separate power supply
Phone Connector	A 1/4" connector used for the transmission of mic or line-level audio
Phono Connector	A small audio connector used for the connection of line-level signals and S/PDIF-format digital audio connections
Polarity	Refers to whether or not the hot and ground leads in two audio cables are connected to their destinations in the same manner
Post	The designation for accessing audio just after it leaves a particular channel component; for example, "post-fader" grabs audio just after it leaves the channel's main level control, before it gets to the panning control
Pre	The designation for the accessing of audio before a particular module; for example, "pre-EQ" grabs audio before it gets to a channel strip's EQ
Pumping	The undesirable sound of a compressor or expander switching on and off
Q	A synonym for "bandwidth"
Ratio	The mathematical formula that describes the amount of gain change to be applied to a signal that exceeds or falls below the threshold settings of a compressor or expander, respectively
RCA Connector	Another name for a phono connector

Reverb	An effect in which the ambience of a physical space is simulated; a signal is copied many times, and the copies are heard one after another at decreasing levels, so closely together that they are not heard individually
Return	A bus or input jack that receives signal, commonly used for effect outputs
RF	for "Radio Frequency," interference from local radio stations that's sometimes picked up and passed along audio cables
Riding	The process of continually adjusting a signal's level
Roll off	In EQ, the reduction of the level of lower-most or upper-most frequencies
Sample rate	The number of times per second a digital device captures an image of a signal
Scene	All of a song's or project's mixer settings saved in a Roland digital mixer's memory; can be quickly recalled, re-establishing all settings instantly
Send	A bus or output jack that transmits signal
Send & Return effect	The process of adding an effect to a signal by sending a copy of the signal to an effect, and mixing the effect's output with the original signal; also called an "effect loop"
Sequencer	A MIDI recorder that captures MIDI data and can play it back in real time
Shelving	A type of EQ in which all frequencies above or below a selected frequency are affected; low shelving affects all frequencies below the selected frequency; high shelving all those above it
Shielding	The electric or magnetic materials used in a cable meant to protect its signal from unwanted noise
Signal	A general term for audio as it travels through audio cables and equipment
Signal Flow	The journey a signal takes from one place to another
Slapback	Type of reverb whose beginning is slightly delayed to simulate the reflection of sound off of a physical wall; also called- "pre-delay"
Slate	A spoken label recorded at the beginning of a take, such as "Remix, Take 1"
SMPTE	For "Society of Motion Picture and Television Engineers"; synchronization technology used for coordinating the timing of audio and video equipment
Snapshot	A captured group of Roland digital mixer settings that reflect the state of the mixer at a particular moment within a song or project; mixer can recall the snapshot and re-instate its settings at the proper moment during the playback of the song or project
Soft button, knob	A multi-use button or knob on a digital mixer whose function is defined by software
Solo	When monitoring, the isolation of one signal by silencing all other signals
S/PDIF	For "Sony/Philips Digital Interface Format, " a standard for the sending and receiving of digital audio data; typically uses phono connectors
Split	A mixer with a separate section for controlling multi-track tape returns
Stereo	A two-dimensional image created by two signals, each of which is assigned to one of a pair of speakers arranged left and right of each other
Strip	Short for "channel strip"

Sweep	In EQ, to quickly select in sequence all of the frequencies in a signal
Synchronization	Or "sync"; the coordination of timing between audio and/or video devices
Take	An attempt at recording something; each try is called a "take"
Threshold	A designated level that triggers an action in a compressor, gate or expander
Track	A stream of recorded audio data
Transient	A very brief high-level signal
Treble	The higher frequencies in a signal
Unbalanced	A type of connection that utilizes only two of the leads—the high and ground—of a cable, connector and jack
Volume	A general term for a signal's loudness
XLR Connector	A high-quality three-pin audio connector; also called a "cannon connector"; also used for AES-EBU-format digital audio connections
Y cable	An audio cable with one jack on one end, and two on the other

Roland

Basics of Mixing

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