The Basics of

Modern Recording

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Introduction and Table of Contents

Audio recording has changed dramatically in recent years. Excellent digital equipment with vast capabilities is now quite affordable. Technology has put brand new, exciting features in the hands of a huge and growing number of individuals.

Low cost and high technology has meant that many people are leaping directly to sophisticated recording equipment for their first recording experience. Others are moving across from digital sequencing - a very different recording experience that will not necessarily prepare them for some of the issues of audio recording. Both groups need to grasp certain fundamentals to get the most out of modern recording equipment.

This book is designed to introduce the basics of modern recording in a simple format, allowing musicians to 'get up to speed' quickly and easily.

You may wish to read this book through completely, or jump to the page you need. If you are new to the recording process, we would encourage you to read the entire book. Also, you may want to read through the glossary at the end of the book to become familiar with some of the terms that will be used.

Basic Recording / Multitrack Recording

The recording process, whether accomplished with a cassette recorder, digital multi-track recorder, hard disk recorder, or any other recording medium, is essentially the same. The goal is to capture sounds onto a master recording. To do this, recording engineers employ a two-step system:

1. Multitrack Recording - the process of recording and overdubbing various instruments and vocals, each to its own "track."
2. Multitrack Mixdown - the process of simultaneously re-recording these multiple tracks down to one set of stereo tracks (the "master recording") which can be reproduced by a typical playback system, such as a CD player or cassette deck.

Multitrack Recording

Multitrack recording involves "stacking" multiple instruments and/or vocals on top of each other so that when any one of them is played back, all of them can be heard in time with each other. This is made possible by recording products that have the capability of recording many different instruments, each on its own "track." Imagine the tape from 16 different cassette decks, all laid side-by-side and glued together. This would give you a sixteen-track tape (actually 32 tracks, because cassette tape is stereo and has two tracks), with the potential of recording a different instrument on each track.

In other words, let's say you record a drummer, a bass player and a rhythm guitar player playing a song, each instrument being recorded onto its own individual track of a multitrack recorder. Because they are all playing together, their notes are all "in time" with each other, so that upon playback it will still sound like they are playing together, even though their instruments are recorded onto individual tracks. If you want to add a lead guitar to the song, You will play the first three tracks so that the guitar player can "keep time" with the other instruments while recording his guitar onto a fourth track. This process is known as "overdubbing."

Traditionally, recording engineers would record "rhythm tracks" first, consisting of drums, bass, rhythm guitar, keyboards and a "scratch" (to be replaced later) lead vocal, all recorded together. Next, the engineer would begin the overdub process, adding other rhythms, leads, background vocals, any other instruments, and finally re-recording the lead vocal. However, modern recordings are often created a single track at a time, beginning with sequenced instruments, drums loops, or even vocals.

The point is that eventually all of your instruments must be recorded onto various tracks in time with each other. Once this is accomplished, the mixdown process begins.

Basic Recording / Multitrack Mixdown

Multitrack Mixdown

The purpose of the mixdown process is to reduce all of your recorded tracks down to two tracks (stereo) or even one track (mono). This allows your song to be played on conventional playback systems, such as cassette or CD players.

Traditionally, multitrack recorders were connected to multichannel mixers, so that each track has its own channel on the mixing board and can be processed individually. In other words, the individual track outputs from a multitrack recorder were connected to the individual channel inputs of a mixer, which merged all these channels down to a single stereo output. This stereo mixer output was connected to the stereo input of a master tape deck, which recorded the stereo signal.

Along with merging many channels down to two channels, the mixer performed other important processes, such as:

- adjusting the frequency content of the instruments, called “EQ-ing.”
- adding various effects, such as reverb, echo or chorus to the instruments.
- adjusting the volumes of each track so that no single instrument is too loud or too soft.

These processes will be explained in greater detail later in this document.

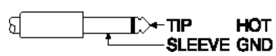
Today, all of these functions, including multitrack recorder, multichannel mixer, EQ and effects, can be found in a single unit. Furthermore, the master recorder may be a CD recorder, DAT tape, or hard drive. What is important is that all of your song’s instruments get recorded, processed and mixed down onto some medium by which they can be heard by your audience.

Common Connections

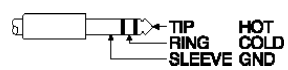
Inputs

Before you begin recording, you will need to connect your instrument or microphone to the input section on your recorder or mixer. You will probably notice that there are a number of different connector types. Variations on the connections include RCA type (the same connectors found on consumer stereo equipment), XLR (most commonly used for microphones) and 1/4-inch (mostly used for instruments).

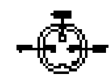
Unbalanced RCA



Unbalanced 1/4-inch



Balanced 1/4-inch tip, ring, sleeve (TRS)



Balanced XLR

Master Out

These outputs are usually connected to your studio monitors (or their power amplifier, if the monitors are not powered) or to the analog inputs of your cassette or DAT machine.

Monitor Out

The Monitor outputs are usually used in the same capacity as your Master Outs (power amp, cassette recorder, DAT machine) but sometimes for the purpose of sending a different mix out of the console. A common situation would be for the Master Outs to be connected to a cassette deck or DAT for recording the mix, and the Monitor Outs to be connected directly to the power amp or studio monitors for listening to your mix.

Digital Out

A Digital Output is specifically made to send signal to another digital device. Their most common use is to send your master mix digitally to a DAT machine. The Digital Outs can also be used to connect to external effects processing gear that has a Digital input.

Digital Optical

Digital Coaxial

Aux Sends

Aux Sends are usually used for sending data out of your console to get processed by an effects processor (reverb, delay, etc.). You can also use Aux Sends to send your mix to a different source, much like you would use a Master Out or a Monitor Out, or as outputs for individual tracks to allow you to transfer your tracks to a different recorder.

Aux Returns

Many consoles have Aux Returns, which bring the signal back in from the external effects processor. These might also be used for just inputting a stereo source, such as a CD player, into your console.

Phones Out

Where a pair of headphones would be connected to the console.

Balanced and Unbalanced Connections

When dealing with the various connections discussed in the previous section, there are two main issues to consider: impedance and balancing. These concepts are important to understand in order to record each piece of gear at its best possible quality.

Impedance

Impedance, also known as resistance, refers to the electronic hardware's inherent resistance to the flow of an AC circuit. In other words, all electronic circuits, including cables, have a natural friction-type resistance to the free flow of electricity, in the same way that a runner encounters resistance from the wind. A runner is stronger when he or she runs with little resistance from the wind. In the same way, a low impedance audio signal is stronger than a high impedance signal because it encounters less resistance.

For practical purposes, impedance between various pieces of gear should always match. Connecting an output that expects a high impedance input into a low impedance input can cause problems because too much current is being sent. For example, a low impedance microphone should be input into a low impedance mixer input. If you need to connect two devices with different impedances, you should use a matching transformer to convert the impedance of one of the devices so that they match.

Impedance Types

Hi Impedance: A circuit whose impedance rating is 1,000 Ohms or greater.

Low Impedance: Any circuit whose impedance rating is 600 Ohms or less.

Balanced vs. Unbalanced

Generally, inputs and outputs of audio gear are either balanced or unbalanced. Balanced cables use an extra wire as a shield to help prevent noise from being picked up along the length of the cable run. Generally, quarter-inch cables and RCA cables are unbalanced, and XLR or stereo quarter-inch cables are balanced (three connecting pins instead of two).

Every piece of gear has either balanced or unbalanced ins and outs. If you are connecting a balanced output to a balanced input, you should use a balanced cable. The connectivity issues can be summarized as follows:

- Unbalanced in/out connected to unbalanced in/out - may as well use an unbalanced cable; using a balanced cable will not hurt but the extra wire will not be used and nothing will be gained by using it.
- Unbalanced in/out connected to balanced in/out - same as previous.
- Balanced in/out connected to balanced in/out - should use a balanced cable; using an unbalanced cable will render the connection susceptible to noise, particularly on cable runs over 10-15 feet.

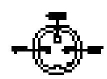
Balanced and Unbalanced (continued)

It’s helpful to note that balanced vs. unbalanced is usually, but not always, related to impedance. For example, XLR cables are almost always low impedance, but quarter-inch cables can be either balanced or unbalanced and either low or high impedance.

Also, if you are connecting a balanced output to an unbalanced input over a long cable run (10 to 15 feet or more), it’s a good idea to use a balanced cable for the majority of the cable run and a direct box or matching transformer right before you connect to the unbalanced input. That way you can take advantage of the higher noise rejection capability of the balanced cable.

Balanced Line
An audio circuit consisting of three wires: High (+), Low (-), and a shield connected to Ground. The High and Low wires are an equal potential difference from the Ground. This is a common design used to help prevent noise and interference for lengthy cable runs.

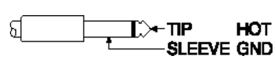
Balanced 1/4-inch
tip, ring, sleeve (TRS)



Balanced
XLR

Unbalanced Line
An audio circuit consisting of two wires: High (+) and Low (-). The High wire carries the signal while the Low wire is connected to Ground. The ground conductor serves as a shield around the other conductor. Because the High and Low wires are not at an equal potential difference to the Ground, they are considered “unbalanced.”

Unbalanced
RCA



Unbalanced 1/4-inch

Microphones

Generally speaking, there are two ways to get a signal into a mixing console, which makes it available to be recorded onto your multitrack recorder. The first way is very simple: direct line input. Direct line input refers to a connection, usually via common guitar-type 1/4" cable, from the output of an electronic instrument to the line input of your mixing console. This is the common method for inputting signals from keyboards, drum machines, sound modules and guitar or bass amps (via direct or line out from the back of the amp). If your mixing console does not have 1/4" line inputs, you can use a Direct Box to translate these line outputs into microphone inputs for your mixer.

The second way to get signal into your mixing console is to use microphones, which connect directly into the microphone inputs of your mixer. Microphones are typically used to record vocals, all acoustic instruments, such as acoustic piano or guitar, and quite often guitar and bass amps. How different types of microphones should be used to record various types of instruments is probably the most critical and difficult aspect of a recording engineer's job. Different microphones *sound* different, and how they are combined, along with how they placed, what angle they are placed at, and the distance from the instrument at which they are placed, are all important factors. Ask ten engineers and you'll get ten different views on micing technique. Although teaching micing techniques is beyond the scope of this document, it is helpful to understand some basic differences between microphones. This will allow you to begin to experiment knowledgeably, which is how all engineers have learned about micing.

Types of Microphones

Microphones, like speakers, are *transducers*. Transducers are devices that convert one type of energy to another type of energy. Microphones convert acoustical energy into electromagnetic energy. How this conversion is done defines what type of microphone it is.

The most common type of microphone is the **dynamic** (or "moving coil") microphone. Dynamic microphones use a coil wrapped around a magnet, which vibrates when sound hits the thin diaphragm attached to it. This vibration results in voltage waveforms which are analogous to the acoustic waveforms which come into the microphone. Your recorder is then able to record these voltage waveforms.

Dynamic microphones are generally very sturdy, relatively inexpensive and can handle high levels of sound. Therefore, they are very useful as the "workhorse" microphones for your studio. Try them on drums, background vocals, guitar and bass amps, and just about anything else. However, dynamic microphones often do not have the frequency response needed for some critical applications, such as lead vocals, cymbals or overhead drums. If you have a condenser microphone available, use it for these types of applications.

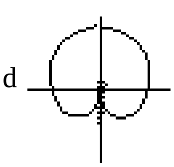
Ribbon microphones are a less common type of microphone, which use a ribbon suspended in a magnetic field rather than a coil. Ribbon microphones have somewhat more high-end frequency response than dynamic microphones, but have traditionally been more fragile as far as handling and sound pressure level. Therefore, they are commonly used for instruments which have a higher-range frequency content, but aren't too loud, such as orchestral instruments, hi-hat or vocals.

Condenser microphones simply use two plates, one of which vibrates according to sound hitting it, with a magnetic field between them. Condenser microphones typically have a very wide frequency response along with a very realistic, transparent sound. However, condenser microphones require a separate power supply and are often many times more expensive than dynamic microphones. Therefore, most small studios will buy one or two of these and only use them for critical applications, such as lead vocals, room mics, cymbals or acoustic instruments.

Polar Patterns

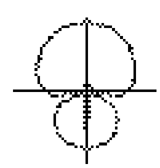
Each microphone has a distinct polar pattern, also called a *pickup pattern*. This defines the area around the microphone in which the microphone will "hear" sounds. It is important to know the polar pattern of your microphone so you will know how to place it effectively. For example, if your microphone only picks up sound right in front of it, you'll need to place it directly in front of your instrument.

Most microphones have a **cardioid** pickup pattern. This means that it will pick up sound directly in front of it, and to a lesser extent along each side.



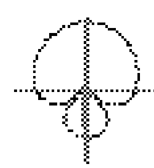
Cardioid

Hypercardioid microphones will pick up sound in front from a greater distance, but less along the sides.



Hypercardioid

Supercardioid microphones will pick up even a greater distance in front and almost none along the sides.

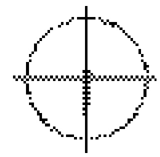


Supercardioid

Microphones (continued)

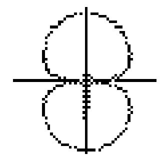
These microphones, also known as **directional** microphones, are perfect for preventing *leakage*. Leakage is the undesired pickup of instruments other than the one you are micing. For example, placing a directional microphone on your snare drum prevents the pickup of the hi-hat sound.

Other microphones, called **omnidirectional** microphones, pick up sound from all directions. These microphones are good for picking up the ambiance of the room, as with overhead drum mics, or for recording entire string sections or choirs.



Omnidirectional

Another type of microphone is a **bidirectional** microphone. These microphones pick up sound on either side, but not from the front or back. There are typically used by placing them between two instruments so that both can be recorded together while maintaining separation between them.

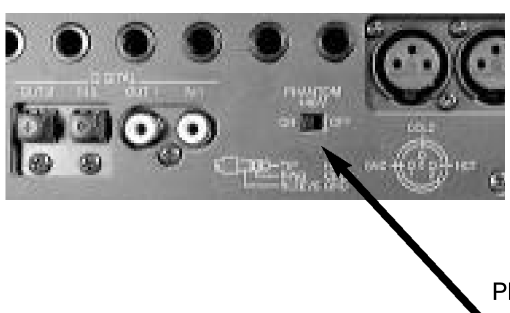


Bidirectional

Mic Pre-amps and Phantom Power

A **pre-amp** is a type of amplifier found on the input section of many consoles, or as a stand-alone piece of outboard gear. The main purpose of the pre-amp is to boost the microphone level signal (-50dBm) up to a line level signal, which is what most consoles operate at. This provides you with greater control of your signal level and provides a certain amount of isolation from outside interference, which could cause noise in your signal path. Pre-amps usually have an output or pre-amp trim knob to adjust the output level. If the pre-amp's output is too high, it can add distortion, noise, and a coloration to the sound. A pre-amp should be used when all of the devices in the recording chain are set at optimal levels and the signal level is still too low.

Also, most condenser microphones contain a type of built-in pre-amp that requires power to operate correctly. This power, referred to as **phantom power**, is usually supplied by an internal battery or by the mixing console. Mixing consoles provide power by sending a voltage (usually +48 volts DC) along the audio cable. On most mic pre-amps and input sections of mixing consoles, there is an on/off switch that determines whether phantom power will be sent or not, giving the condenser mic the voltage it needs to provide a strong enough signal.



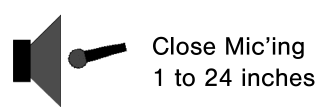
Phantom Power switch
on a Roland VS-1680
Digital Studio Workstation

Basic Microphone Technique

The most important thing to understand about microphone technique is that there is no definitive right or wrong way to do anything. Mic'ing techniques, like microphone selection, are subjective, and individual techniques are developed over time. Furthermore, mic'ing techniques vary according to musical style and tend to evolve or change over time. Therefore, this document will merely provide some basic guidelines and suggestions, and should be used as a jumping-off point for personal experimentation.

The basic microphone techniques can be roughly divided into four categories: close mic'ing, distant mic'ing, accent mic'ing and ambient mic'ing.

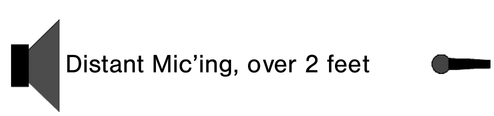
Close Mic'ing
Close mic'ing is the most common type of mic'ing used in typical recording studios. Close mic'ing means that the microphone is placed very close to the sound source, usually about 1 - 24 inches away. Close mic'ing allows the sound source to be recorded relatively free from outside noises (called *leakage*), such as other instruments in the studio. Close mic'ing also provides a tight, "in-your-face" sound, which is generally preferable for multi-instrument songs.



Generally, close mic'ing is done with a directional microphone, such as a cardioid, hypercardioid or super cardioid, by placing the microphone slightly angled near the spot where the sound is emanating. For example, a guitar amp is close mic'd by placing a microphone about an inch away from the spot on the amp grill which is midway between the edge and the center of the speaker, slightly angled toward the center. A trumpet would typically be close-mic'd by placing the microphone about three inches away from the bell of the horn.

However, always be sure to test your placement by moving the microphone around the sound source while listening to the results. When you hear the sound the way you like it, leave the microphone there.

Distant Mic'ing



Distant mic'ing involves placing one or more microphones more then 2 to 3 feet away from the sound source. Distant mic'ing is most often used when recording multiple instruments at once, such as a string section or chorus ensemble. Distant mic'ing allows these groups to be recorded as a whole, so that the overall natural tone balance can be captured without the need for individual mic'ing. Distant mic'ing also allows the sound of the room (called *ambience*) to be recorded, which is often desirable in good sounding rooms, such as a recording studio or church.

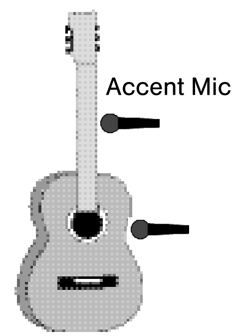
Basic Microphone Technique (continued)

Getting good distant microphone placement takes experimentation. Generally, with a single microphone, the microphone is placed in the center of, for example, a group of singers, about five feet out front. With two microphones, place them equal distance off center, about four feet apart. However, it is especially critical when distant miking to experiment with placement while listening to the results in order to find the "sweet spot."

Accent Mic'ing

Accent mic'ing is often used in conjunction with distant mic'ing, to provide additional pickup of particular elements of a group of instruments. For example, you may want to place a microphone in front of a soloist in an orchestra. However, you must be careful to place the microphone close enough to pick up the solo, yet far enough away so that the balance of the ensemble is not affected when the soloist isn't soloing. Usually this placement is a bit farther away than a close microphone, but again this distance should be finalized by experimentation.

Accent mic'ing can be used when mic'ing even a single instrument. For example, maybe you've gotten the perfect acoustic guitar sound by placing one microphone near the bottom of the sound hole and another up the fretboard. You may want to consider placing a third microphone directly in front of the picking location in order to mix in just a touch of the pick noise, which can dramatically enhance the realism of your recording, especially when the guitar is the only instrument being recorded.



Ambient Mic'ing

Ambient miking is similar to distant mic'ing, except that its main function is to restore the natural reverberation and room sound of a particular recording environment. Ambient mic'ing is particularly important when making a live recording, because the ambiance that these microphones pick up allows your listeners to experience the feel of the live show.

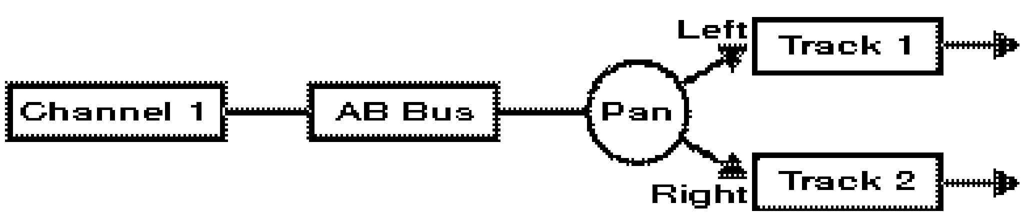
Ambient mic'ing is usually achieved by placing a pair of microphones, either cardioid or omnidirectional, out front of each side of the stage, often near the sound board.

Bussing

Beginner recording enthusiasts are often confused by the concept of how the signal being input to the mixer or recording device gets through the console to its final destination. A very important aspect of signal flow is the concept of the **Bus**. Simply put, a bus is just a pathway for the signal to follow; from an input to a track, from a track to a track, from a track or input to an effects processor, etc.

When “bussing” signals from one place to another, your signals are passing through a type of “routing matrix**.”** A routing matrix is just another name for a summing amplifier; an amplifier used to combine multiple signals while keeping their respective volume levels and pan positions. A good analogy for record busses is the way you control the water in your house. The water usually enters your house at a single location (barring any natural disasters such as floods, hurricanes, tsunamis, etc). It is then routed through a series of pipes to different locations in your house. The path that the water follows is determined by which faucets you turn on or off.

It is common for most mixers to have “stereo busses” which direct your signal to one or two locations. For example, record busses are usually setup in pairs such as tracks 1 and 2 (or A and B), tracks 3 and 4, C and D, etc. The amount of signal sent to each side of the stereo bus is controlled by the pan position. If you assign a signal to the record bus for tracks 1 and 2 and pan that signal all the way to the left, all of the signal will be recorded on track 1 and none of the signal will be recorded on track 2. Setting the pan all the way to the right would have the opposite effect. If you set the pan position to the center, the signal will be recorded equally on tracks 1 and 2. Just think of a stereo bus like a two-lane highway. The pan knob is your steering wheel and it controls which lane you drive in.

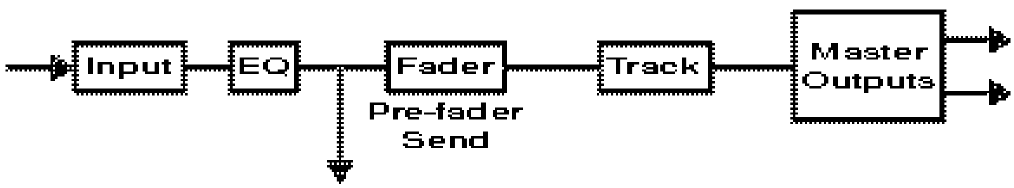


Stereo Bus

Pre-fade and Post-fade

The concepts of pre-fade and post-fade are based upon fader control. Just as their names imply, pre-fade relates to the audio signal **before** it reaches the fader and post-fade is **after** the audio signal goes through the fader.

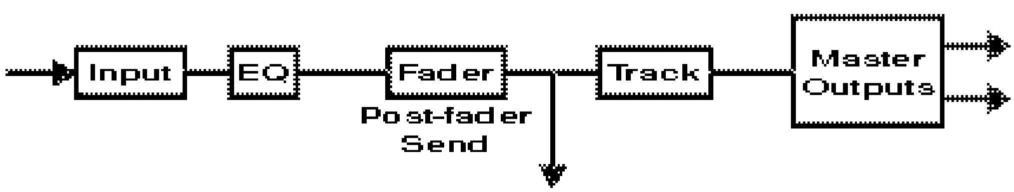
The pre-fader setting is most often used when the audio is to be independent from the fader. You are able to hear the raw level of the audio signal without using the fader to alter the level. This setting is invaluable when monitoring input levels for recording. Pre-fader settings are also preferred when sending a separate headphone mix without altering mixer settings. Using pre-fade in this manner allows any instrument to be turned up in the headphones without affecting the main mix. Pre-fade can also be utilized to lower the fade level of the dry lead vocal while allowing the lead vocal's reverb to



Pre-fader Send

remain.

The post-fade setting is more widely preferred as it renders fader control. Post-fade allows you to monitor the output levels and control over the independent levels of each track. More control over effects are available while using post fade effects, as you have total control over how much effect will be returned



Post-fader Send

to a specific track as well as the level of the original audio signal .

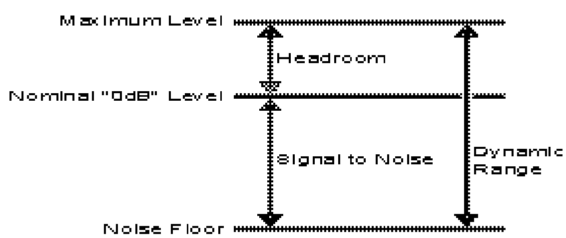
Getting Correct Input Levels

One key to great recording is starting with the right input levels. When you cook, you know that having the temperature too high or too low can either burn or undercook your meal. The same thing is true when you record. Levels that are too hot or too low can destroy an otherwise great performance. If your input level is too low, you will most likely add unwanted noise to your mix when you raise the level of that track during mix down. On the other hand, if your input level is too high, there will usually be distortion on your tracks.

Dynamic Range

When you record, you may notice that the level seems to vary greatly depending on the performance and the style of music you are working on. For example, in a ballad, a drummer may be playing a rim shot during the verse but then goes to a heavy snare during the chorus. This variance in level is referred to as dynamic range. Dynamic range is different than volume. Volume is the amplitude at any one moment in time. Dynamic range is the amount of variation in terms of amplitude.

The DynamicRange of Equipment



Analog vs. Digital

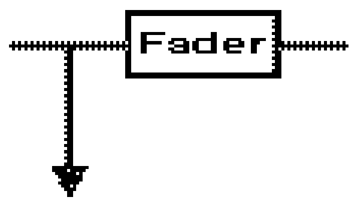
There are many differences between analog and digital recording when it comes to setting levels. With analog recording, input levels can be recorded over 0dB without distorting. Sometimes, recording input levels over 0dB can help to get a hotter or warmer signal to the tape. Higher levels also help mask some of the noise (tape hiss) that is present on most tape based recorders.

In the digital domain, input levels cannot exceed 0dB and, by definition, there is nothing higher. In fact, most digital recorders will not display anything above 0dB. If an input level does exceed 0dB, you risk adding distortion to your recording. This type of distortion is usually referred to as "digital distortion" and it is very undesirable for most recording applications. To get the best results when recording digitally, your input levels should be recorded between -12 and -4dB. Not only will this give you the equivalent of a good analog recording level, but it will give some headroom for mixing after the track is recorded. Digital recorders do not have the noise floor problems associated with most analog recorders and are capable of recording at lower levels without adding noise to the recording. When in doubt, record at lower levels. A track that is recorded too low can usually be fixed whereas it is virtually impossible to remove distortion from a recording.

Pre level and Post level

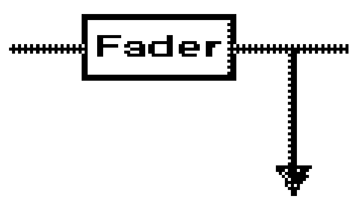
In order to ensure that you are recording at a good level, it is important to understand how the meters on your mixer operate. Most mixers will allow you to view the meters in one of two settings: pre level and post level.

A **pre-fade level** setting shows the signal level that is being sent to the mixer before it passes through the fader. This is the most accurate indication of your signal level as it is input to your mixer. When recording, it is recommended that you set your faders to 0dB, set your meters to pre level, and use your input trims to set a good recording level.



Pre-fade

A **post-fade level** setting displays the level of the signal after it passes through the fader. This is the most common setting for viewing the levels of recorded tracks. After the signal passes through the fader it usually goes to the mix buss or main outputs. This makes a post level setting ideal for viewing levels during mix down.



Post-fade

Monitoring Effects

Effects processing is the process of changing, augmenting or otherwise modifying the audio signal. When an effect is applied to the audio signal it is considered to be “wet.” When the audio signal has no effect applied to it the signal it is considered “dry.”

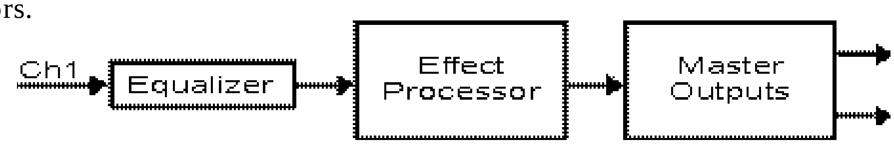
Imagine if you had an aquarium filled with water and you wished to change the color of the water to red. You could apply a red film over the viewing area of the aquarium to give the effect of red water but the water is really clear. You may even want to experiment with various film colors. In regards to effects, this would be the equivalent of monitoring an effect without recording it. Now, lets say you turned the same water red by applying red food dye to the water. The water is now actually red and that cannot be changed. In recording, this is similar to recording a signal “wet” or “printing” a track with effects.

So, when is it best to record wet and when is it better to record dry? The advantage of recording a signal wet is that you no longer need to use your effects processor to create the desired effect. That effect is part of the recording and that frees the effects processor to be used on another track or to create a different effect. The advantage of recording a signal dry is that you maintain the flexibility to change the effect at a later time. This is especially useful if you are unsure what effect(s) you want to use.

Another common recording technique is to record the signal dry on one track and wet on another. This allows you to create a mixture of dry and effected signal and is particularly useful on effects such as reverb and delay. Also, by maintaining a dry track, you have the advantage of changing the effect in the future if desired. This technique can be used if you have lots of available tracks or, if you have a recorder that offers Virtual Tracks.

Effects Routing / Insert vs. Loop

There are two common types of effect routing that are used in audio recording: insert and loop (or send and return). When you insert an effect, you are placing that effect between the source signal (usually an input or a previously recorded track) and that signal's destination (usually a mix or record buss). Inserting an effect changes the physical characteristics of the source signal. Effects commonly used in an insert fashion include compression, distortion, hum canceler, mic simulators and guitar amp simulators.

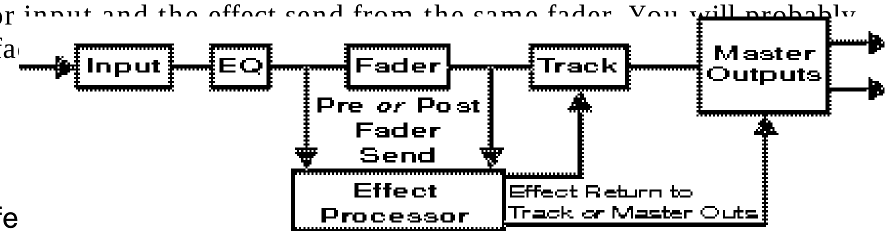


Insert Effect

It is important to note that, when routing an effect in an insert configuration, you are dedicating that effect processor to the source signal that you are sending to it . This means that you cannot use that effect processor on any other input or track while it is routed this way.

When you use an effect in a loop (send and return) configuration, you are splitting your source signal between its destination and the effects processor. This gives you a separate dry signal (the original source signal) and wet signal (the return of the effects processor). This allows you to control the mix of dry and effected signal. The term "loop" is used because the source signal is usually sent from the input section of the mixer to the effects processor and then sent back to the input section, creating a "loop" of the source signal. Effects commonly used in a loop fashion include reverb, delay, chorus and flange.

Most mixing consoles give you the option of sending the signal to the effects processor before it reaches the fader (prefade) or after it passes through the fader (postfade). Prefade allows you to control the level of the track or input without changing the effect send level. For example, to create the illusion of a voice sounding as if it were falling off of a cliff, you might use a prefade send to a reverb effect. That way you could use the fader to lower the volume of the dry voice while the reverb remained at its original level. Lowering the dry signal while maintaining the reverb level creates the illusion of a voice getting farther away. Postfade, on the other hand, allows you to control the level of the track or input and the effect send from the same fader. You will probably use a postfa



Loop Effe

The Compressor

A compressor is an effects processor that compresses the dynamic range of any signal passing through it; basically, making the loud sounds softer and the soft sounds louder. The compressor's main duty is to lower the amplitude of the loudest components of the signal it processes; this process is referred to as gain reduction. Along with lowering the amplitude of the loudest signals, a small amount of amplification is applied to boost the low level signal as well. These actions work together to narrow the dynamic range of the signal.

Here is a brief description of some of the parameters common to most compressors:

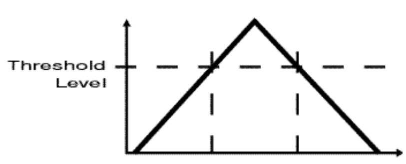
Threshold
The threshold control sets the amplitude level at which the compressor begins to function. When the input level is above the threshold value, the compressor begins to work.

Ratio
The ratio control determines the amount of input level to output level. For example, a 4:1 ratio means that for every 4dB coming into the compressor above the threshold setting, 1dB will be output. Basically, this control tells the compressor how much to compress; that is, the amount of gain reduction.

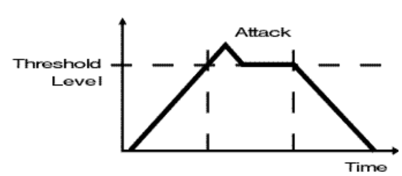
Attack Time
The attack time setting controls how soon the compressor begins to attenuate the signal after the input signal has gone above the threshold level. In other words, the amount of time after the signal crosses the threshold before the compressor begins to work.

Release Time
The release time controls how long the compressor will hold the signal at the threshold level once the input signal drops below the threshold level. In other words, the amount of time the compressor stays on after the signal drops below the threshold level.

Output Gain
The output gain is used to adjust the output level of the compressed signal. Using the output gain, you can bring the overall output level of the compressor to its optimum setting.



Signal at Compressor Input



Signal at Compressor Output

Basics of EQ

EQ—short for equalization—is the process of adjusting the amplitude (volume) of certain frequencies. The tone control of a car stereo, for example, is a very crude EQ control. Usually one EQ can be used to boost or cut more than one single group of frequencies, or *bands*. For example, an EQ that has high, mid and low controls is referred to as a 3-band EQ.

Uses of EQ

EQ can be used for a variety of applications. The most common are:

Correctional EQ is, unfortunately, most often used to compensate for poor sound or recording quality of instruments or vocals being recorded. For example, maybe during mixdown you realize that your cymbals sound dull. EQ can be used to add some high end to the cymbals and allow them to "sizzle." For another example, let's say you are recording a vocalist and you notice that your microphone is a bit nasal-sounding. By cutting out some of the high-mid frequency range, you can get a warmer vocal sound out of your singer. However, *be careful here!* EQ should not be used as a substitute for poor recording techniques. When you have selected the proper microphones and placed them properly, very little, if any, correctional EQ should be needed.

Creative EQ is the process of adjusting frequencies for purely creative reasons. For example, maybe you would like your vocalist to sound like she's singing through the radio. By severely limiting the bandwidth of the vocal frequencies, you can obtain this effect.

Blending – Professional engineers are able to build a "wall of sound" by assigning certain ranges of frequencies to certain instruments, so that these ranges line up to each other to cover the entire frequency spectrum without overlapping each other.

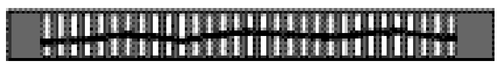
EQ can be applied either during the recording process or, more commonly, during mixdown. Be careful when recording your track with EQ, because it can't be adjusted later. Also keep in mind that adding EQ can often add noise, so it should be applied conservatively (this mainly applies to analog EQ, not digital).

Types of EQ

EQs come in a variety of shapes and sizes, from stand-alone rack-mount units to the EQ controls built into a mixing console. Here are some of the more common categories of EQs:

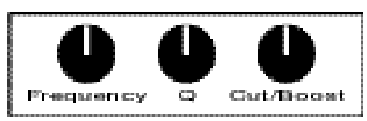
Graphic EQ

The most common type of EQ is the graphic EQ. The graphic EQ offers boost/cut controls for a series of pre-selected frequencies that collectively cover a wide range of frequencies. The graphic EQ can be easily recognized because the controls are typically a series of sliders that are physically lined up next to each other, providing a graphic representation of the overall EQ curve. These EQs are commonly used to adjust the overall sound of a mix, and vary according to the frequency characteristics of the room you're in.



Parametric EQ

The parametric EQ allows you to select specific frequencies to boost or cut. For example, the EQ section on your mixing console may have two controls for each band: one control to select the frequency and one control to boost/cut the selected frequency. Note that this is actually a description of a *sweepable* or *semi-parametric EQ*, and not a full parametric EQ, although you will often hear it referred to as such. A full parametric EQ has one additional control per band: the Q adjustment. This control allows you to define how wide or narrow the band of frequencies is. This will make more sense if you understand that when an EQ boosts or cuts a selected frequency, it is actually affecting a group of frequencies (the *band*) around the selected frequency. The Q adjustment defines how wide the band is. Semi-parametric and full parametric EQs are generally used to make specific frequency adjustments without affecting the overall sound.



Shelving EQ

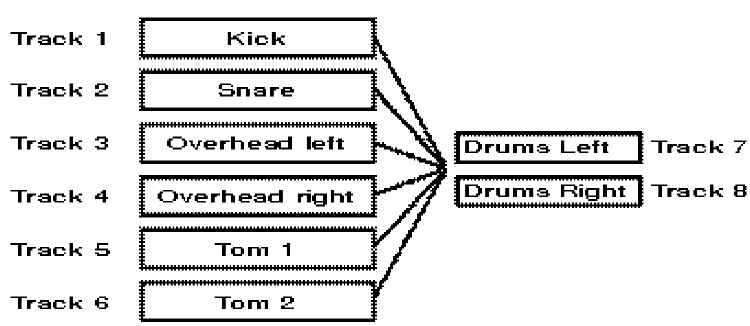
Another type of EQ typically found on mixing consoles is known as a *shelving EQ*. This type of EQ typically consists of a single boost/cut control for either a high or low band. When you boost/cut the high or low band, you are boosting/cutting a fixed frequency (typically 10KHz or 12KHz for the high band and 80Hz or 100 Hz for the low band) *as well as all the frequencies above/below it.* Shelving EQs are useful for adding brightness or "bottom end" to a particular track.

Bouncing Tracks

It is often necessary or desirable to combine the data of previously recorded tracks to a single track or a stereo pair of tracks in the same project - this process is commonly referred to as **track bouncing**. It is very important to understand that track bouncing is a *recording* process; instead of recording a live instrument or vocal onto a track or tracks, you are recording previously recorded tracks onto other tracks.

Engineers usually track bounce for several reasons: to combine the audio on several individual tracks to fewer tracks, to re-record an already recorded track with EQ settings, to “print” (record the audio output) effects applied to a track with the original track itself, etc. Track bouncing has many benefits. For example, if you bounced 8 individual drum tracks down to a stereo pair of tracks, you would end up with a stereo drum mix that would sound the same as the original 8 tracks but only uses 2. You then could re-record other instruments on the first 8 tracks. Also, sometimes you might need more effects in your song than there are effects processors in your studio. In this case, you can bounce a track or tracks elsewhere while effects are applied to them, and record both the original tracks and their respective effects. After the effects are recorded, the processors are free to be used on other tracks.

When track bouncing, make sure that you adjust individual settings such as level, EQ, and effects before you bounce tracks. Once your individual tracks are “bounced” down and you record over the original tracks, you will be able to adjust only the mixed version of those tracks. For example, if you bounced 8 tracks of drums down to a stereo pair, you would be unable to add reverb to the kick drum without adding it to the rest of the drums. If you have the need to bounce tracks, you might want to use a recorder that lets you keep your original tracks after you bounce. This will give you the best of both worlds; you can free up tracks for recording by bouncing while retaining individual control over your original tracks. This concept is often referred to as “virtual tracking” and we will discuss it in more detail in the next section.



Bouncing 6 Tracks to 2 Tracks

Virtual Tracks

What Are Virtual Tracks?

In the “old days” most albums were recorded in studios with expensive tape recorders with lots of tracks. Often the artist needed many tracks in order to have several different versions of the lead vocal, or the guitar solo. Or, maybe they wanted to have several background vocals on different tracks so they could mix them later. They needed lots of tracks because they didn’t want to throw away any of their recorded takes. Bouncing tracks, as we discussed previously, would free up tracks for recording but, in order to record additional tracks, it was necessary to record over the original tracks. Today there is a new solution to the “I never have enough tracks” dilemma—Virtual Tracks.

Virtual Tracks give you the ability to bounce tracks and still keep all of your original takes for later comparison, editing, or re-mixing.

How Virtual Tracks work

Picture several stacks of file cards. Each stack has one card on top, with several others underneath. By shuffling the cards, you can bring any individual card to the top of the stack at any time. You can write on or read a card by bringing it to the top of the stack.Virtual Tracks work in the same manner. Each track has one Virtual Track selected. You can listen to anything that is already recorded on that track or you can record new takes to that track. Recording on one Virtual Track does not erase things that were recorded on the other Virtual Tracks.

You can select any Virtual Track for playback. In fact, you can even combine pieces of multiple Virtual Tracks into a single composite or “best of” track. You can also mix multiple Virtual Tracks down to a single track. This is great for creating fuller sounding string sections or thickening vocal tracks. Virtual Tracks are great for trying new ideas because you can record new takes without erasing something that you want to keep.

Here are just a few ideas on how you can use Virtual Tracks in the recording process:

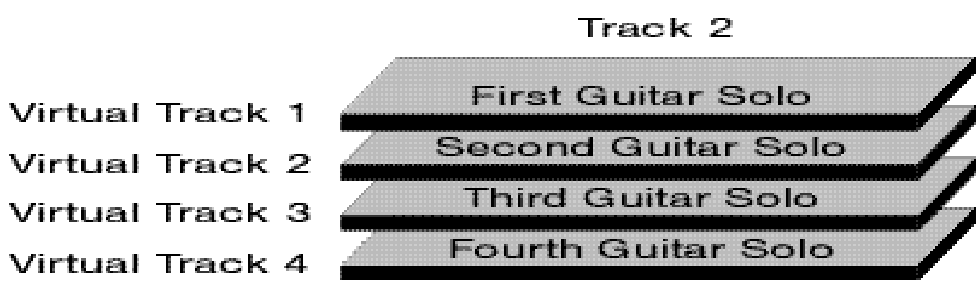
Recording a guitar solo with Virtual Tracks

When recording a guitar solo, try recording different takes of that solo on different Virtual Tracks within the guitar track. You don’t have to erase previous takes or lose other tracks, just change Virtual Tracks for each take. This helps to keep the creativity flowing while you are recording. You can decide which solo (or parts of solos) you want to use at a later time.

Virtual Tracks (continued)

Recording background vocals with Virtual Tracks
Record several takes of background vocals on different Virtual Tracks within one playback track. When you are finished, temporarily assign each of these Virtual Tracks to the various playback tracks so that you can hear all of them at the same time. Keep one track open for recording then adjust the levels of the other tracks as desired and bounce them to the open track. This will give you a “mixed” version of all your background vocal takes on a single track. You can now recall your original Virtual Tracks to be played back with the new background vocal track. If you need to adjust some of the original takes, don’t worry, they’re still safely stored as Virtual Tracks.

Recording a dry and processed guitar at the same time using Virtual Tracks
Try splitting your guitar signal so that you route the output of the guitar to one input and the output of your effects processors to another input. Then, record your guitar part to two different tracks. After you are finished recording, mix the “dry” and “wet” tracks to your liking and bounce them together to any available Virtual Track. You can use that track for playback and keep the “dry” track on a spare Virtual Track. That way, if you want to experiment with different guitar effects at a later time, you can recall the “dry” guitar track and apply new effects to it.



Virtual Tracks

Track Management

In a perfect multitrack recording environment, each instrument or vocal is recorded onto a separate track, so that it can be given its own particular EQ, effects, panning and volume. Most of this processing is done during the mixdown, when the goal is to blend all of these instruments together in order to create a stereo master recording that sounds exactly the way you want it to.

However, for most recording engineers, there is no perfect multitrack recording environment. Although 64- and even 128-track recording environments do exist, they can usually only be found in large professional studios at upwards of $400 per hour. Therefore, most of us work on recording systems that have a limited number of tracks available. However, this does not mean that we can't produce lush, multi-layered recordings that sound professional, it just means we have to plan ahead. The process of creating a track plan for each instrument and vocal is sometimes referred to as track management. If you don't do this ahead of time, you may find yourself out of tracks and options for finishing your song.

Bouncing Tracks

One of your most useful tools will be a process called "bouncing tracks." Bouncing tracks is essentially the process of doing mini-mixdowns during the recording process (for more detailed information, see page 22). For example, let's say you have recorded your drums onto individual tracks, using up eight tracks of your sixteen-track recorder. You can "bounce" these eight tracks down to one stereo pair, freeing up six tracks. However, you must be careful when bouncing tracks because, if you record over your original tracks, you lose your option to individually mix these tracks during the main mixdown.

Of course, bouncing tracks works much better on digital recorders than it does on analog recorders, because digital bounces do not lose sound quality. Also, some digital recorders allow you to record new tracks without erasing the previous tracks. This concept is sometimes referred to as "virtual tracking" and it affords you the luxury of bouncing tracks without losing control of the independent tracks (for more detailed information, see page 23). If you are using an analog recording system, you should be aware that each bounce loses some audio quality. Generally, more than two or three bounces for any given track renders the track too noisy to be very useful.

Stacking Tracks

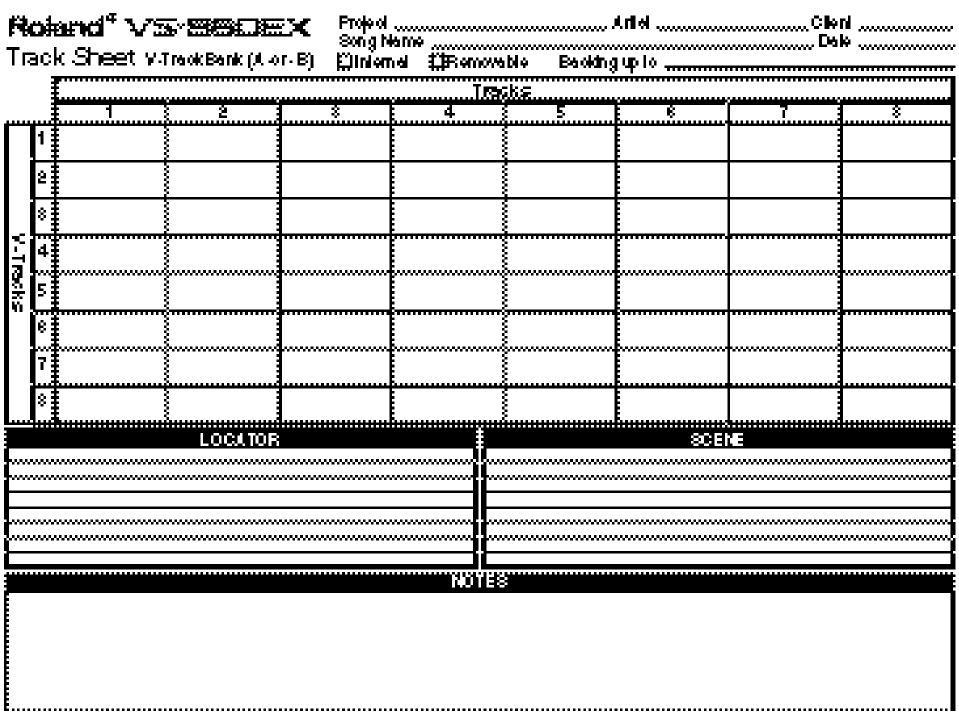
Another useful trick is to place more than one instrument on a particular track, sometimes known as "stacking tracks." For example, let's say you have a piano solo during the intro of your song, and a guitar solo during the fade-out. In this case, you can record them both onto the same track because they do not overlap.

Track Management (continued)

When stacking tracks, remember to use instruments that need little or no effects processing or at least use similar types of processing, so that you don't have multiple changes to worry about during mixdown. If you need lots of processing for a track, consider recording the processing along with the track, so that you don't have to do it during mixdown. Also try to select instruments that are as far apart in the song as possible, so that you have plenty of time between them to make any changes you do need to make.

Track Sheets

It should be clear by now that these types of complex track management tools should be well thought out ahead of time. Most engineers use tracks sheets, which are sheets of paper that have a separate box for each track where instruments that are going to be assigned to that track can be listed, along with other track notes. By using track sheets, you can map out what will happen on each track ahead of time. If you don't have any tracks sheets simply draw one out before you start. The small investment in time at the beginning of your session will potentially save you countless hours and headaches during your recording process.



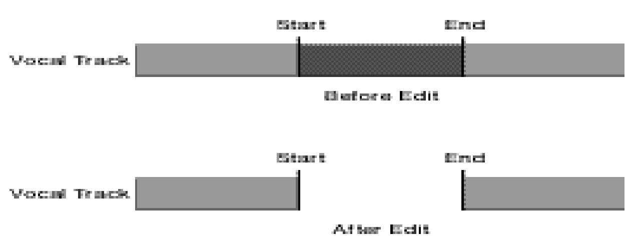
Non-Linear Editing

In analog days, the only way to edit the audio on tape was to actually break out the razor and cut and splice the tape itself–not exactly the most efficient means of editing. Imagine cutting the tape in the wrong place or not putting it together properly. You could destroy countless hours of work with a single misplaced edit.

With the introduction of digital hard disk recording, these problems are now virtually non-existent. A hard disk recorder uses a hard disk to store your recording instead of tape. When you edit audio on a hard disk, you are editing in a temporary memory buffer and you are not actually changing your original recording. This type of editing is referred to as nondestructive. Think of it this way, if you create a memo on your computer's word processing program and then make some changes to that memo a few days later, the changes don't become permanent until you instruct your word processor to save the changes in your document. Nondestructive editing works in a similar fashion; you can make your edit, listen to the result, and then decide if you want to keep the edited version or restore the original version. Some hard disk recorders take this a step further by storing your original versions as "Undo" levels even after you save the new version (see page 31).

Setting your Edit Points

Digital editing sounds easy but can get a little tricky if you do not understand the basics. Almost every edit you do will need to have a **Start** and **End** point. For example, if you want to erase part of a track, you will need to determine what portion of the track you want to erase. The start point would be where you would want to start your erase and the end point would be where you would want to stop erasing. In the example below, the start point is placed on the vocal track where the vocalist came in too soon and the end point is placed after the mistake. When you perform an erase, the area marked by the start and end point is erased and the timing remains unchanged.



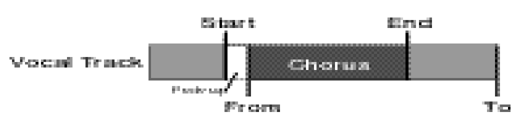
Start and End Points

Non-Linear Editing (continued)

In addition to start and end, some edits require additional edit points. These points are usually referred to as the From and To points. From and To points are used in conjunction with Start and End points. Most edits that involve moving or copying audio to a different track or a different time will require a From and a To point.

When performing this type of edit, Start and End points are used to select the portion of the track that will be moved or copied whereas From and To points are used to determine how and where that portion of the track will be moved to. In other words, the To point is simply a destination for the material between Start and End. The From point is a location between Start and End that will line up with the To point after the edit is made. For most edits, the From point will be the same as the Start point.

If you have a key reference point (e.g., the downbeat of a measure, a sound effect, etc.) that is not at the beginning of the section you want to edit, you can use the From point to align that reference point with the desired time. For example, let's say you want to copy the lead vocal from the first chorus to the second chorus but the vocalist starts singing a little before the chorus actually starts. The example below shows how to use the From and To effectively to achieve the proper edit.



Before Edit

After Edit

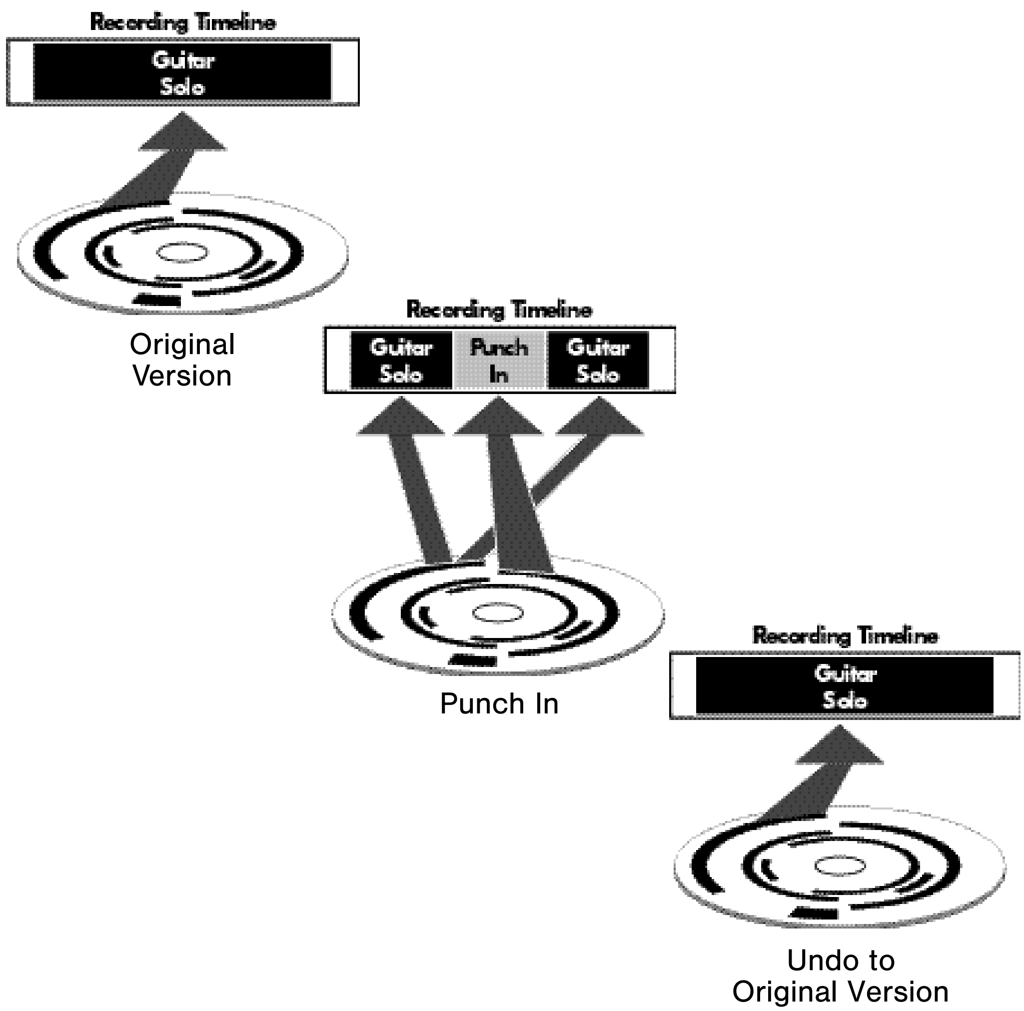
Using From and To Points

Undo

.Many current hard disk recorders have a function known as **Undo.** This function allows you to cancel the last performed operation (usually this is limited to the recording of a track or a track edit function) and return your song to its previous condition.

Some hard disk recorders contain multiple levels of Undo, allowing you to actually "go back in time" through your recording session. This is particularly useful in situations where a track was accidentally recorded over or edited out, a track was recorded at too low or high a level, etc. For example, say you record a guitar part. Then you 'punch in' some new guitar playing to the middle of the track. If you realise that you made the punch too early, just Undo it!

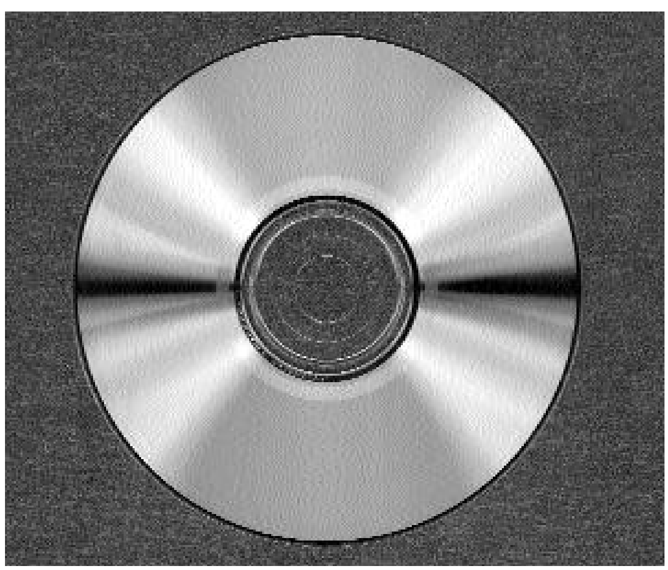
Use the **Undo** function to remove the 'punch in' guitar edit, leaving you at the place and time of the session where you just had your original guitar track.



Mastering

"Why don't my recordings sound as loud as the ones I buy in the store?" This question has been asked by the vast majority of recording enthusiasts struggling to make their project sound as professional as possible. The main reason is that nearly all of the CDs and/or tapes you buy in a music store have been professionally mastered. So, what is it that a mastering engineer does that makes everything sound so much better? A mastering engineer is usually responsible for aspects of the recording process such as maximizing and balancing volume levels, adjusting the timing between tracks, and correcting any frequency type problems that are common to all tracks. The mastering process is also the last chance to fix any remaining problems with the mixed version of the songs by using tools such as EQ and noise reduction.

The mastering engineer works exclusively with two track songs that have already been mixed. The main tools of the mastering engineer (aside from his or her ears) are compression and EQ. Effects applied during the mastering process are applied to the entire mix, not individual tracks. Compression is generally used to help increase the overall level as much as possible without causing distortion. In digital recordings, 0dB is the loudest a signal can be without distorting. EQ is usually used to shape the overall tonal balance of the songs. For example, if the mixed version of your songs lack good bass response, EQ can be used to boost the lower frequencies. The main advantage of using a professional mastering engineer is that most pros have many years of experience. They know what tools to use and, more importantly, when to use them. If you are going to master your own recordings, it is probably a good idea to read a few articles and/or books on the mastering process to provide some insight on the various tips and tricks used by the professional mastering engineer. Also, listen to some already mastered CD's so that you can get a better idea of the sound you want before you start.



Backup

Having a single working copy of your platinum-selling breakthrough masterpiece is never a good idea. Musicians are not immune to Mr. Murphy's law and it is usually not until we lose some important, irreplaceable data that we start to learn that lesson. A backup is a second copy of your recording that can be used in the unlikely event that your working copy is lost or damaged. To make a backup of a tape based recorder you will most likely need two recorders, one for recording and one for playback. Most hard disk recorders make the backup procedure a bit easier. Usually backup functions are built-in and allow you to store a copy of your song to storage mediums such as DAT tape, magneto optical drives, and CD-R. Of these, CD-R and CD-RW disks are rapidly becoming the most popular, and for good reason. They are inexpensive, reliable, and readily available.

A final thought on backing up your recordings. If you make a backup copy of a song, and then erase or record over the original version, you no longer have a backup. You must have two copies of your recordings to have a backup. If you are going to erase a song from a hard drive or record over a master tape, make sure you make two backup copies of that song first. When it comes to making frequent backups of your recordings, the shoe people have the right idea—"Just do it!"

A Roland VS-CDRII

Recording Overview

Now that we've discussed the recording process in general, let's review the steps involved in creating a multitrack recording.

1. Connect your instrument or microphone to an input on your mixing console and/or recorder.
2. Route that input to an available track for recording.
3. Adjust the input level for your instrument or microphone to a good recording level.
4. Record the track.
5. Repeat steps 1-4 to record additional tracks as desired.
6. Apply effects and/or EQ to your tracks as desired.
7. Adjust the levels of your tracks to create a balanced stereo mix.
8. Record your mixed tracks to a stereo pair of tracks or a two track recorder such as a DAT or cassette recorder.
9. Make any final adjustments (such as EQ and/or compression) to your stereo mix and record your stereo mix to its finished medium (e.g., CD, cassette, vinyl).

Summary

So, there you have it. A complete history of everything you need to know about the recording process. O.K, well, not exactly. But, we hope this book has provided you with some of the fundamentals you need to start making your own recordings.

We've quickly covered a wide variety of topics that are important to understand before you can start using your recording gear to its fullest potential. If you are interested in learning more about these subjects, there are many books and articles covering every aspect of the recording process--from amateur to professional.

The best teacher, however, is experience. This is certainly true when it comes to making great recordings. We encourage you to take the knowledge gained from this book and start applying it to your recordings. Remember, no one gets in a car for the first time and heads off for Indianapolis the next day. Go slowly and take one step at a time. As you progress, you will start to find the techniques that will produce the types of recordings you are striving for. Until then, remember to have fun along the way!

Glossary

The following section contains definitions for some terms that you will likely encounter in the recording process.

Aux Send / Return Normally refers to the output Bus of a mixer used to "send" a signal to an external processing device. The Aux Return is the return input to the mixer used for the return signal from the effects device.

Balanced An audio circuit with 3 wires; two wires carry the signal, high (+) and low (-), and the third is a shield which is connected to a chassis or system ground.

Bass The low audio frequency range, normally considered to be below 500 Hz.

Bus A signal path to which a number of inputs may be connected for feed to one or more outputs.

Compressor A compressor is a dynamic effect that decreases gain as the level of the input signal increases to reduce the dynamic range of the audio. A compressor may operate over the range of input levels, or it may operate only on signals above or below a given level (the threshold level).

Condenser Microphone A microphone utilizing a capacitor (condenser) as a pickup element. Electronics are usually contained in the micro phone body and a polarizing voltage is necessary, so external or battery power is required, and output levels are usually higher than other types of microphones. Condenser microphones are commonly used for recording vocals and acoustic instruments.

DAT (Digital Audio Tape) A recording medium that records audio signals to tape digitally, via a hardware recording device called a DAT recorder.

dB The "dB" (decibel) is a unit of measurement for ratios of sound level, power, voltage, and other quantities.

Dynamic Microphone A type of microphone which converts acoustical to electrical energy by means of a permanent magnet and a moving coil. Dynamic microphones do not require external power. Some dynamic microphones have very high quality and are commonly used in recording and sound reinforcement.

Effects Return A mixing console input that receives the signal from an effects device. Many mixing consoles have a level control to adjust the amount of effects signal returned to the mix; this control is called the effects return control.

Effects Send A mixing console output that sends a signal to the input of an effects device. Most mixing consoles have an effects send level control for each input (channel).

Equalizer (EQ)
An electronic device that will amplify (boost) and/or attenuate (cut) certain portions of the audio frequency spectrum. There are many different types of equalizers.
Graphic
An equalizer which operates simultaneously at a number of preset frequencies, any of which may be boosted or cut independently of the others.
Parametric
An equalizer where the center frequency is continuously variable over a given frequency range, and where the "Q" (slope rate) is usually adjustable.
Shelving
A boost or cut characteristic which has a response curve resembling a shelf. Maximum boost or cut occurs at the indicated frequency and remains constant at all points beyond that frequency.

Fader
A potentiometer that controls the signal level for a console input position or output channel.

Gain
The amount an amplifier increases the power of a signal, usually specified in dB.

Headroom
"Headroom" refers to the difference between the nominal operating level and the maximum level at any point in an audio system or device, usually expressed in dB.

Hertz
Abbreviated "Hz," the unit of measurement for frequency; 1 Hz is equal to one cycle per second (cps).

Impedance
The total opposition to the flow of alternating current in an electrical circuit. Impedance is measured in ohms.

Insert
When a track is routed to a Bus such as an effects Bus using an INSERT path, the audio goes to the effect Bus and then directly back to the channel in the mixer. Therefore, there is no "dry" sound of the original without the effect because the only audio path is through the effect and then to the mixer. INSERTS are used when you don't want to hear the original sound without the processing (e.g., when you use a compressor on a voice).

kHz
Abbreviation for kiloHertz, or one thousand cycles per second.

Locator
A position marker placed in your song to help you find sections quickly. Locator locations can usually be accessed directly from front panel buttons.

Marker
A marker is any temporary mark placed within a song to indicate a particular location you wish to return to. For example, just like a bookmark points to a page in a book, a marker points to a song location to make it easy to find later.

Media
Media is the term used to indicate the actual surface or device your song is recorded on. For example, cassette tape, reel to reel tape, hard drives, and removable media are recording media.

Glossary (continued)

Microphone A device for converting sound waves into corresponding electrical signals. Microphones can be categorized in several ways: their sensitivity patterns, the method by which they convert sound to electrical energy, or other characteristics.

MIDI Musical Instrument Digital Interface. A digital communications language that allows musical instruments and related equipment to ‘talk’ to each other.

Mix The procedure whereby two or more signals from live and/or recorded sources are combined to achieve a desired balance.

Mixdown The process whereby signals from a multi-track tape recorder are routed to a mixing console and recombined to make a stereo or monaural master tape.

Mixer A device or system in which two or more signal sources (mic or line level) can be combined and fed to another device or part of the audio system. Larger mixers are often called “mixing consoles.”

Mute Reducing an audio signal to off (full attenuation).

Ohm A measurement of electrical resistance.

Omni-Directional Equal sensitivity in all directions. Usually refers to non-directional microphones.

Overdubbing A recording procedure utilized in multi-track tape recording. A performer (or performers) listens to previously recorded musical tracks, typically with headphones, while recording one or more additional tracks.

Pan Pot The control that places a signal in stereo perspective to appear acoustically between the left and right speakers.

Phantom Power A method of remotely powering the preamplifier or impedance converter which is built into many condenser microphones by sending voltage along the audio cable. Many professional mixing consoles supply phantom power. In the cases where phantom power is required, but not available from a mixer, you can use an external power supply for the microphone.

Pre-Fader / Post-Fader Audio signals from an external source or a recorded track come into a mixer and are then routed to a Bus such as the Mix Output. If the audio signal is routed to the Mix Output Bus *Pre*-Fader, then the faders won’t have any effect on the level of the sound in the mix. If the source or track is routed *Post*-Fader, then the fader will control the level to the Mix Output Bus. Generally, tracks will be routed to the Mix Output Bus or the Aux Bus Post-Fader.

Glossary (continued)

Punch In
A procedure in multi-track recording that is essentially an overdub, but instead of recording the new part on an adjacent track to already recorded material, the new part is recorded by erasing a previous part from a given track. The punch in is initiated, while the song is playing, by entering record mode at some precise instant. For example, a punch in at bar XX, or a punch in after a given word. (Punch out refers to either stopping the recording or switching from record back to play mode for the track or tracks involved.)

Sample Rate
The rate at which samples are taken in the analog to digital conversion process, usually specified in Hertz (Hz).

Status
A mode selection to choose the monitoring and recording condition of each channel (e.g., Record, Source, Play, Mute).

Stereo Bus
In a mixing console, the Bus or channel which is used to feed a program to a stereo tape recorder, 2-channel sound system or other stereo equipment. A stereo Bus actually consists of two Buses: left and right.

Tracking
Refers to the original recording of live music when the first tracks (usually rhythm instruments) are recorded.

Treble
The high audio frequency range, normally considered to be above 5000 Hz.

Unbalanced
An audio circuit with 2 wires; one wire carries the signal, high (+) and the second carries the low (-) and also is connected to chassis or system ground.

Undo
A command to eliminate a previous action, command, or recording.

Virtual Tracks
Virtual tracks are additional areas to record your takes and ideas. They are found on some hard disk recorders. You can usually edit several Virtual Tracks together to create a composite track. On some recorders, you can even bounce several Virtual Tracks to another Virtual Track to make a mix of these tracks. For example: If you need to bounce some background vocals together to make room for some guitar parts, you can use the Virtual Tracks to store the original vocal parts that you have bounced. Unlike a tape based system, you can then add new material on the tracks you have bounced from and still keep the originals in case you want to re-mix the bounce later.

XLR
Describes any of several varieties of audio connectors having 3 or more conductors plus an outer shell which shields the connection and locks the mating connectors. 3-pin XLR-type connectors are commonly used to make balanced mic and line level connections in professional audio systems.

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