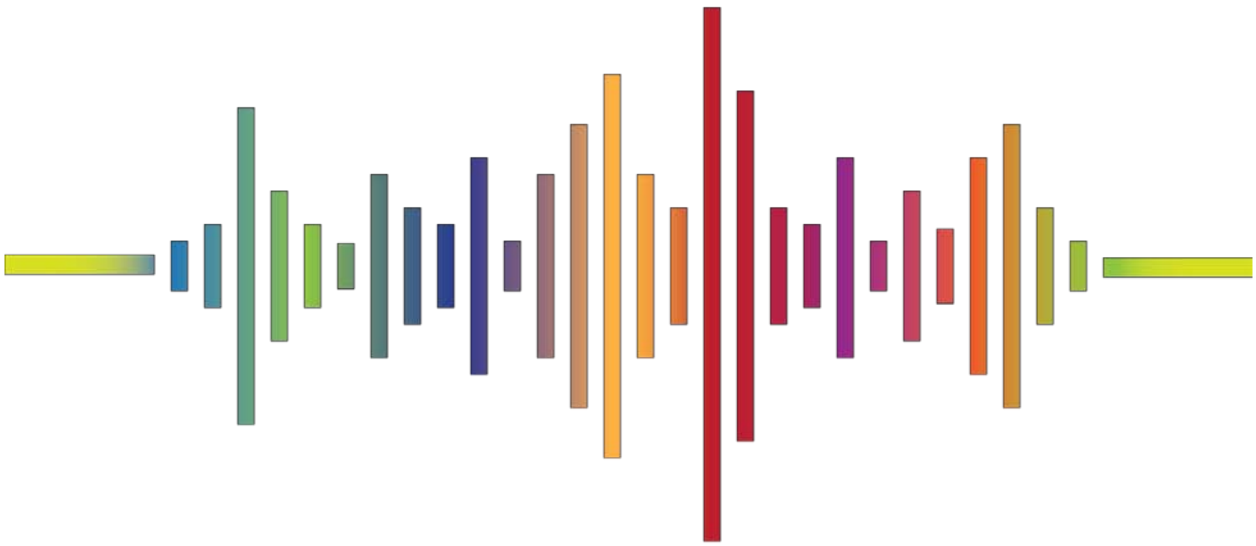
# MUSIC



**RECORDING • MIXING • MASTERING**

# Transposition Chart

| | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| 0 | C | C# | D | D# | E | F | F# | G | G# | A | A# | B |
| 1 | C# | D | D# | E | F | F# | G | G# | A | A# | B | C |
| 2 | D | D# | E | F | F# | G | G# | A | A# | B | C | C# |
| 3 | D# | E | F | F# | G | G# | A | A# | B | C | C# | D |
| 4 | E | F | F# | G | G# | A | A# | B | C | C# | D | D# |
| 5 | F | F# | G | G# | A | A# | B | C | C# | D | D# | E |
| 6 | F# | G | G# | A | A# | B | C | C# | D | D# | E | F |
| 7 | G | G# | A | A# | B | C | C# | D | D# | E | F | F# |
| 8 | G# | A | A# | B | C | C# | D | D# | E | F | F# | G |
| 9 | A | A# | B | C | C# | D | D# | E | F | F# | G | G# |
| 10 | A# | B | C | C# | D | D# | E | F | F# | G | G# | A |
| 11 | B | C | C# | D | D# | E | F | F# | G | G# | A | A# |
| 12 | C | C# | D | D# | E | F | F# | G | G# | A | A# | B |

# TABLE OF CONTENTS

# Overview

This book is a reference book for recording, mixing and mastering.

# Terms

| | |
|---|---|
| **Bus or Signal Bus** | A signal pathway within a mixer. Many mixers have several busses. The primary signal bus is usually referred to as the Main Bus. In addition to the Main Bus, there can be one or more Aux Busses. There may also be one or more signal busses for effects. |
| **Decibel** | A unit for measuring the relative level of power voltage, current or sound intensity. A tenth of a bel. |
| **Dry Mix** | A signal that hasn't been altered by effects or processing. |
| **EQ** | Equalizer, Equalization |
| **Harmonics** | An extra quality to the sound beyond the frequency of the note(s) generated by an instrument. |
| **IDAM** | An acronym for - Inter-Device Audio + MIDI<br>Allows the inter-connection of iOS devices. |
| **MIDI** | MIDI is an acronym for (Musical Instrument Digital Interface) is a technical standard that describes a communications protocol, digital interface, and electrical connectors that connect a wide variety of electronic musical instruments, computers, and related audio devices for playing, editing and recording music. |
| **Wet Mix** | A signal that has been altered by effects or processing. |
| | |

# RECORDING

## Signal Levels

In the audio world, there are four signal levels that we deal with: **mic**, **instrument**, **line**, and **speaker**. These levels all have different meanings, so it is important to know the differences between them.

### Differences Between Signal Levels

#### Mic Level

Mic level is the voltage of signal generated by a microphone. This is the lowest, or weakest, level signal of the four and requires a preamplifier to bring it up to line level.

#### Instrument Level

Instrument level signals fall between mic level (lower) and line level (higher) signals. These signals refer to any level put out by an instrument, commonly from an electric guitar or bass. A preamplifier is required to bring the signal up to line level.

#### Line Level

Line level signals are the highest level signals before amplification. This is the type of signal that typically flows through your recording system after the preamplifier stage and before the amplifier that powers your speakers. The two types of line levels are consumer and professional. Consumer line level is rated around **-10dBV** and is what you'll find in products like a CD player. Professional line level is rated around **+4 dBu** and can be found in equipment like mixing desks, preamplifiers, and signal processing equipment.

#### Speaker Level

Speaker level signals are post-amplification. After a line-level signal enters an amplifier, it exists to the speakers at what is called speaker level. These signals are much higher in voltage than line level and require speaker cables for safe signal transfer.

# Digital Audio

Digital audio is a digital representation of analog wave forms.

## Sampling Rate

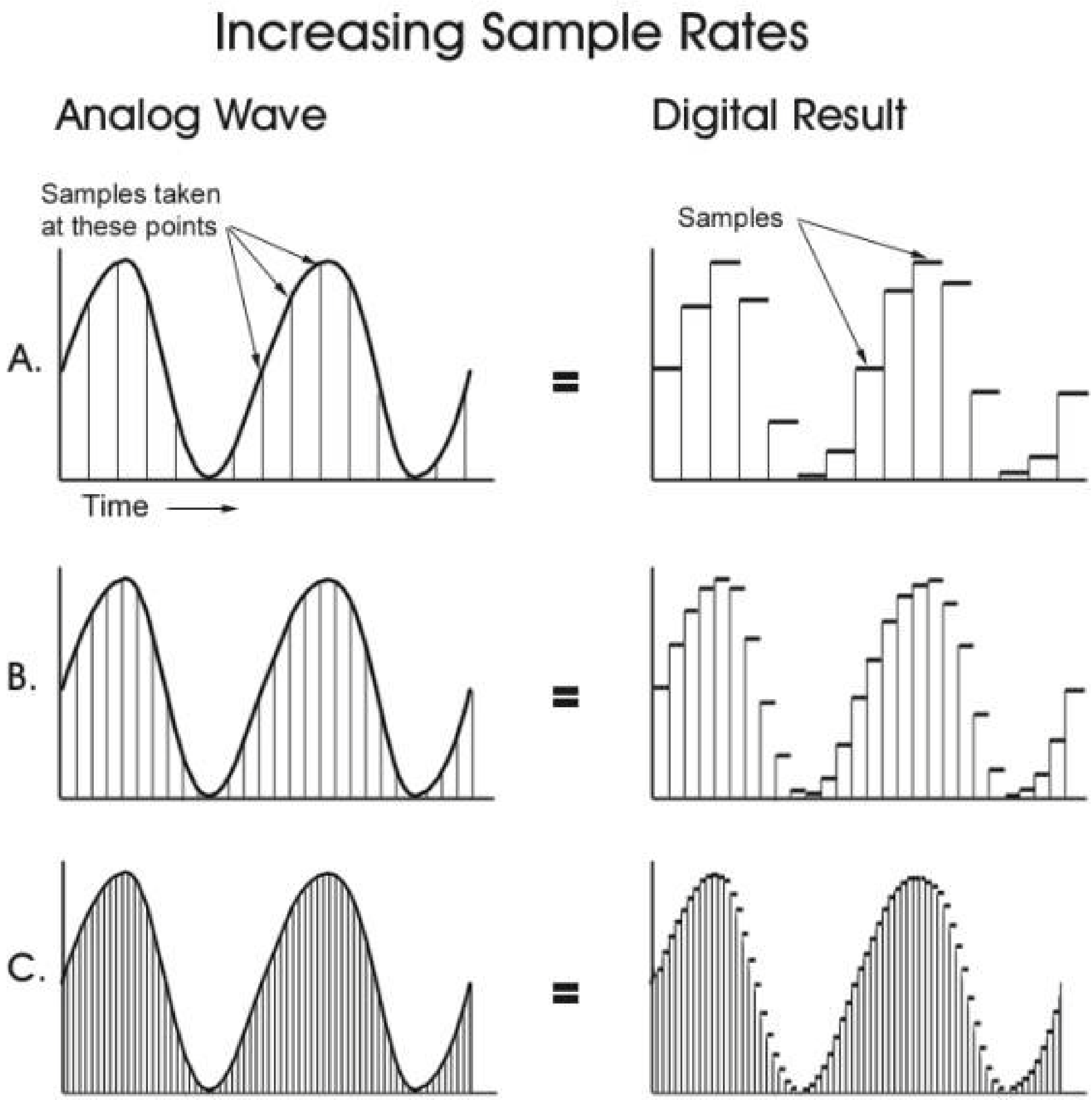
A commonly seen unit of sampling rate is Hz, which stands for Hertz and means “samples per second”. The most common sample rate you’ll see is 44.1 kHz, or 44,100 samples per second. This is the standard for most consumer audio, used for formats like CDs.

This is not an arbitrary number. Humans can hear frequencies between 20 Hz and 20 kHz. Most people lose their ability to hear upper frequencies over the course of their lives and can only hear frequencies up to 15 kHz–18 kHz. However, this "20-to-20" rule is still accepted as the standard range for everything we could hear.

While 44.1 kHz is an acceptable sample rate for consumer audio, there are instances in which higher sample rates are used. Some were introduced during the early days of digital audio when powerful anti-aliasing filters were expensive. Moving the Nyquist frequency even higher allows us to place the filter further and further out of human hearing, and therefore impact the audio even less.

48 kHz is another common sample rate. The higher sample rate technically leads to more measurements per second and a closer recreation of the original audio, so 48 kHz is often used in “professional audio” contexts more than music contexts. For instance, it’s the standard sample rate in audio for video. This sample rate moves the Nyquist frequency to around 24 kHz, giving further buffer room before filtering is needed.

Some engineers choose to work in even higher sample rates, which tend to be multiples of either 44.1 kHz or 48 kHz. Sample rates of 88.2 kHz, 96 kHz, 176.4 kHz, and 192 kHz result in higher Nyquist frequencies, meaning supersonic frequencies can be recorded and recreated. Low pass filters have less impact on the sound and more samples per second, which results in a more high-definition recreation of the original audio.




41.1kHz (CD Quality) or 48kHz or 88.1kHz or 96kHz or 176.4kHz or 192kHz in Logic Pro X

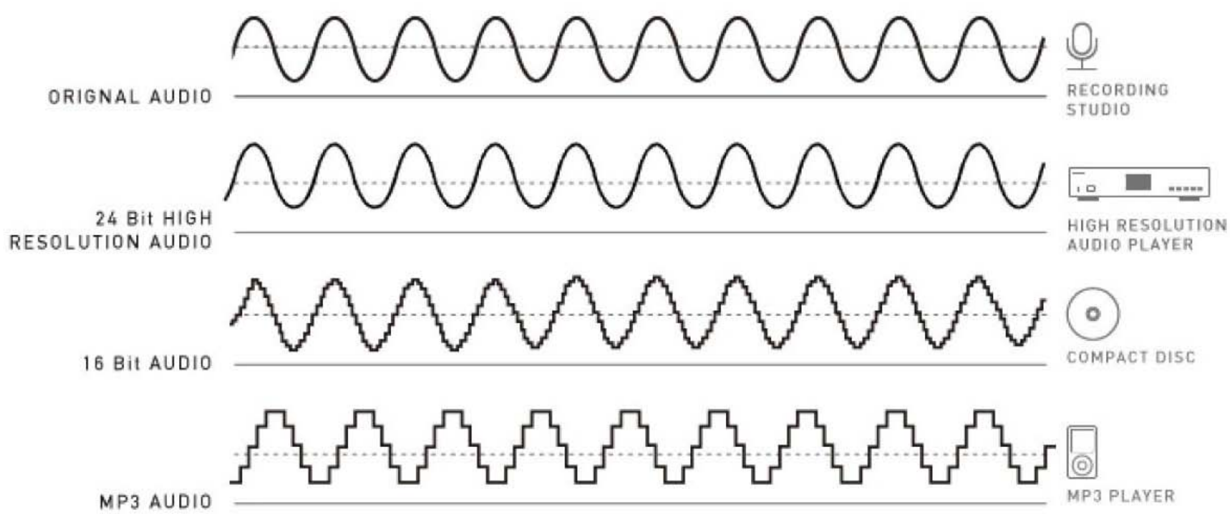
## Bit Depth

Analog audio is a continuous wave, with an effectively infinite number of possible amplitude values. However, to measure this wave in digital audio, we need to define the wave's amplitude as a finite value each time we sample it.

The bit depth determines the number of possible amplitude values we can record for each sample. The most common bit depths are 16-bit, 24-bit, and 32-bit. Each is a binary term, representing a number of possible values. Systems of higher bit depths are able to express more possible values:

| Bit Depth | Values |
|---|---|
| **16** | 65,536 |
| **24** | 16,777,216 |
| **32** | 4,294,967,296 |

With a higher bit depth—and therefore a higher resolution—more amplitude values are available for us to record. As a result, the continuous analog wave's exact amplitude is closer to an available value when sampled. Therefore, a digital approximation of the amplitude becomes closer to the original fluid analog wave.




Increasing the bit depth, along with increasing the sample rate, creates more total points to reconstruct the analog wave.

# Connector Types

### XLR

Used mainly for Microphones and Powered Monitors. Most mixing boards have XLR-Female inputs for microphones and XLR-Male outputs for powered monitor speakers.

### TRS

TRS is an acronym for Tip, Ring, Sleeve. This is a stereo jack. Pay careful attention to the number of rings on a jack. A jack with two rings is a stereo jack, while a jack with one ring is a mono jack. There are generally two sizes or TRS jacks, 1/8 inch (3.5 mm) and the larger 1/4 inch (6.35 mm) jack. Almost all mixing boards, if they have stereo inputs, use the larger 1/4 inch TRS jack.


### TS

TS is an acronym for Tip, Sleeve. This is a mono jack. There are generally two sizes or TS jacks, 1/8 inch (3.5 mm) and the larger 1/4 inch (6.35 mm) jack. It's really rare to use the smaller jack. The most common use for the 1/4 inch TS jack is as an instrument patch cable. This is the type of jack used for an electric guitar, bass guitar, and keyboards. Most if not all stereo output keyboards have two TS jacks, one for the left channel and one for the right.

**Note:** Pay close attention to the number of rings on the 1/4 inch jacks you're using. Most patch cables used by musicians are TS cables. Even the stereo on keyboards consist of a left and right output that use two TS cables.

### RCA

RCA connector, originally called a phono connector, is a type of electrical connector commonly used to carry audio and video signals. The name RCA derives from the company Radio Corporation of America which introduced the design. The connectors male plug and female jack are called RCA plug and RCA jack. They are not as prevalent as they used to be, but sometimes mixing boards, like the ZEDi 10FX, have a set of RCA outputs. The standard color coding: Red = right, White = left and Yellow = video.

### MIDI

Most MIDI today is transported over USB cables, but some older keyboards and interfaces use the original five-pin DIN connector.

## Microphones

### Dynamic

There are many types of dynamic microphones that are applicable from broad to specific applications. Mainly what you'll encounter are dynamic instrument mics and dynamic vocal mics. One of the more common brand and models are the Shure SM58 vocal mic and SM57 instrument mic.

### Condenser

Condenser microphones are more sensitive than dynamic microphones. They are really good for capturing the nuance of the human voice, acoustic guitar and many other acoustic instruments. These are the type of microphones often seen in what's called a shock mount. A shock mount is a structure that is holds the microphone suspended by rubber band like matrix. The purpose is to keep the microphone from moving due to vibrations. Condenser microphones also often use something called phantom power to boost their signal.

## Ribbon

While these mics are no longer as popular, Ribbon mics were once very successful particularly in the radio industry. The light metal ribbon used in these mics allows it to pickup the velocity of the air and not just air displacement. This allows for improved sensitive to higher frequencies, capturing higher notes without the harshness while retaining a warm vintage voicing. These days, interest for Ribbon mics have returned, especially since modern production ribbon mics are now sturdier and more reliable than their old counterparts, making them viable for live multi-instrument recording on venues where noise level is manageable. You can also use them for recording if you're looking for vintage vibe, or you can set it up in combination with dynamic or condenser mics for a more open sounding track.

## Mic Level

Microphones have comparatively small output voltages, on the order of thousandths of a volt (0.001V) ranging up to tenths of a volt (0.1V). Mic outputs can range from very low to very high depending on the mic type and design.

### Low-output Mics

Some mics, particularly dynamic and ribbon mics, have extremely low output levels and need a lot of amplification, commonly called gain, which is the job of the mic preamp. Sometimes these low-output mics require as much as 50dB to 70dB of gain, depending on the sound pressure level (SPL) generated by the sound source and the distance from the mic. Clearly a mic placed in front of a quiet singer will require more amplification than the same mic on a loud guitar amp. Examples of mics with low output levels are the AEA R84, Royer R121, or the Shure SM7B.

### High-output Mics

Condenser mics such as the RODE NT1A, the Manley Reference Cardioid, or the AKG C414 XLII will have hotter outputs requiring drastically less amplification (less preamp gain) to achieve suitable signal levels, sometimes as little as 10dB to 30dB of gain. The reason for this is that condenser mics have amplifiers built right into the mics (sometimes called head amps) that provide the voltage for the mic’s output.

If you anticipate using microphones with low output (called low sensitivity) on quiet sources such as finger picked acoustic guitar, then you will need a mic preamp with a lot of gain. Preamps that only offer 40dB–50dB of amplification will not provide enough gain to record at optimum levels. Most importantly, mic output levels are too low to connect directly to inputs that expect to see our next level — line level.

**Phantom Power**

To know what phantom power is, you need to understand how condenser microphones work. Condensers work on the principle of variable capacitance. In Britain, they are actually called "capacitor microphones." Sound waves vibrate a diaphragm (usually gold-sputtered mylar) that is stretched in front of a metal plate (called the backplate). As the diaphragm vibrates, the distance between the diaphragm and the backplate changes, which changes the capacitance. When that happens, the acoustic variations are converted into tiny electrical variations which must be amplified before they leave the mic. This amplification method is what's known as phantom power.

Phantom power, commonly designated as +48V or P48, was designed to power microphones without using bulky external power supplies such as the ones required for tube microphones. It's a way of sending the DC electrical current required through a balanced XLR cable. We need that voltage to power the diaphragm and the mic's internal amp. When phantom power is turned on, DC current is sent through the XLR cable and delivers the voltage necessary to power the microphone. It's most widely used as a power source for condenser microphones, which have active electronics. In addition, true condenser microphones (as opposed to electret) require a voltage for polarizing the microphone's transducer element, and phantom power provides a voltage for both of these purposes.

Dynamic microphones do not require phantom power. However, there are some very low output dynamic microphones, such as the Shure SM7B (which has an output level is -69dBV), which require a preamp to boost the signal. If the preamp you connect the microphone to is active, you will then need to turn the phantom power on.

# Microphone Response Area

## Omnidirectional

An omnidirectional (or nondirectional) microphone's response is generally considered to be a perfect sphere in three dimensions.

## Unidirectional

A unidirectional microphone is primarily sensitive to sounds from only one direction.

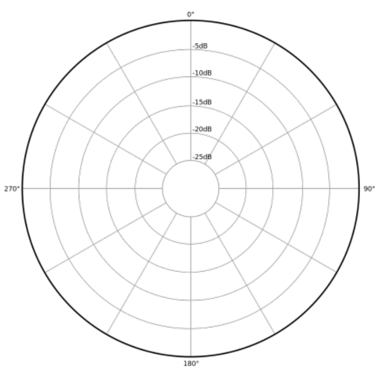
## Cardioid, hypercardioid, supercardioid, subcardioid

The most common unidirectional microphone is a cardioid microphone, so named because the sensitivity pattern is "heart-shaped", i.e. a cardioid. The cardioid family of microphones are commonly used as vocal or speech microphones since they are good at rejecting sounds from other directions. In three dimensions, the cardioid is shaped like an apple centered around the microphone, which is the "stem" of the apple. The cardioid response reduces pickup from the side and rear, helping to avoid feedback from the monitors. Since these directional transducer microphones achieve their patterns by sensing pressure gradient, putting them very close to the sound source (at distances of a few centimeters) results in a bass boost due to the increased gradient. This is known as the proximity effect. The SM58 has been the most commonly used microphone for live vocals for more than 50 years demonstrating the importance and popularity of cardioid mics.

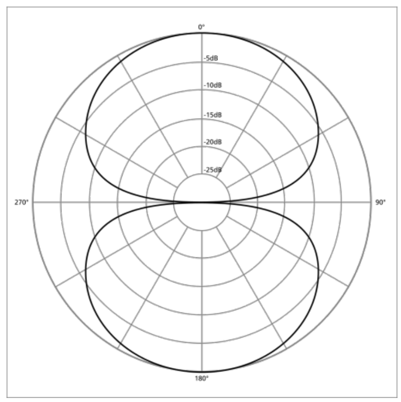
## Microphone Polar patterns

Microphone polar sensitivity. Microphone is facing towards the top of the page in diagram, parallel to the page.

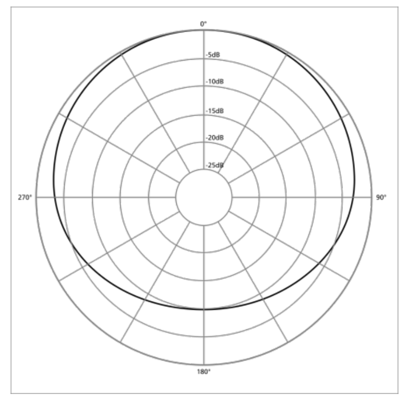
### Omnidirectional



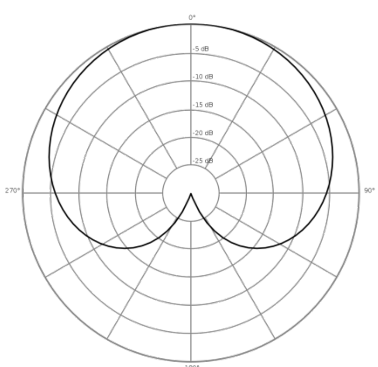
### Bi-directional / Figure 8



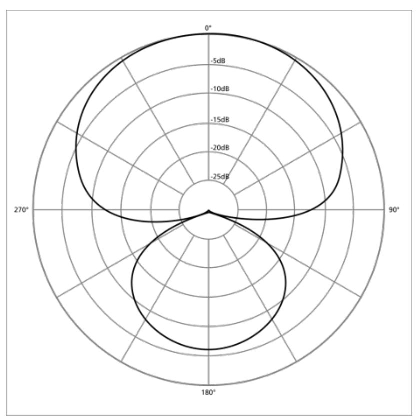
### Subcardioid



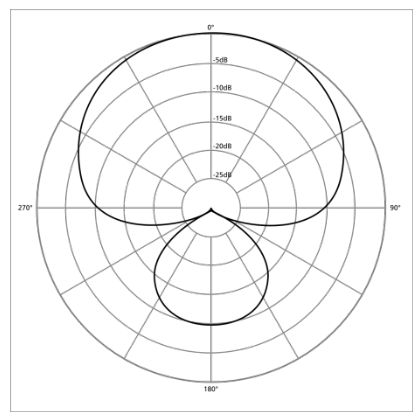
### Cardioid



### Hypercardioid



### Supercardioid



## Direct Input Recording

Electric instruments such as electric guitars, electric basses and electric keyboards can be plugged directly into the mixing board or some other type of audio interface. Always make sure that the levels and gain are turned all the way down before plugging in.

# Gain Staging

In audio engineering, a gain stage is a point during an audio signal flow that the engineer can make adjustments to the level, such as a fader on a mixing console or in a DAW. Gain staging is the process of managing the relative levels in each step of an audio signal flow to prevent introduction of noise and distortion, feeding the inserts, such as equalizers and compressors with the right amount of signal, particularly in the analogue realm. Ideal gain staging occurs when each component in an audio signal flow is receiving and transmitting signal in the optimum region of its dynamic range.

For example, let's say you have an electric guitar[1] that is plugged into an effects board[2] which is plugged into an amp[3] whose cabs are miked by a dynamic instrument microphone which is plugged into a mixing board[4] which is connected via USB to a DAW in which an audio track[5] has been set up to record. At each of these 5 points (stages) in this chain, there is the option to increase or decrease the level of output. The end objective is to achieve the required level of output with the least amount of signal noise.

## Record Input Levels

Too low of a signal from your source means lots of background hiss and noise. Too high a signal from your source results in nasty pops and crackles known as clipping. Getting the levels right when you're recording a track is important. Understanding how to set recording levels will keep those nasty digital crackles and pops out of your projects:

- First, keep your input signal peak between **-12db** and **-6db**. This will typically provide the best signal to noise ratio. It will also assure that your source signal has enough peak room to stay out of the clipping range.
- Second, start from the beginning. Here's what I mean. Make sure to set the input level on your audio interface at this same point (**-12dB** to **-6dB**). It is important to keep this level at the source and work your way back to the **DAW** input. Your signal should be set properly at the input of every device in your audio chain.
- Are you using a preamp before your audio interface? Make sure to set the level properly here first. Next check the input signal of your source connected to your audio interface. Make sure the level is set properly here as well.
- Some audio interfaces do not have a level meter with more than 3 led signal bars (some only have 1). Often these same units will be **green/yellow** when the signal is present and then turn **red** when it clips. If this is the case with your audio interface there is a simple way to properly set your levels.
- Push your input source signal until it shows clipping, then back off the input about **6 dB**. From there just make sure your meter never turns red even at the loudest points in your audio source.

Knowing how to set recording levels from the beginning of your source to the recording signal on your DAW is simple yet critical. Get this part correct and you will have plenty of good clean signal to process in post production.

# Mixing Console (aka Mixing Board)

a mixing console is an electronic device for combining sounds of many different audio signals. Inputs to the console include microphones being used by singers and for picking up acoustic instruments, signals from electric or electronic instruments, or recorded music. Depending on the type, a mixer is able to control analog or digital signals. The modified signals are summed to produce the combined output signals, which can then be broadcast, amplified through a sound reinforcement system or recorded.

Mixing consoles are used in many applications, including recording studios, public address systems, sound reinforcement systems, nightclubs, broadcasting, television, and film post-production. A typical, simple application combines signals from microphones on stage into an amplifier that drives one set of loudspeakers for the audience. A DJ mixer may have only two channels, for mixing two record players. A coffeehouse's tiny stage might only have a six-channel mixer, enough for two singer-guitarists and a percussionist. A nightclub stage's mixer for rock music shows may have 24 channels for mixing the signals from a rhythm section, lead guitar and several vocalists. A mixing console in a professional recording studio may have as many as 96 channels.

In practice, mixers do more than simply mix signals. They can provide phantom power for condenser microphones; pan control, which changes a sound's apparent position in the stereo soundfield; filtering and equalization, which enables sound engineers to boost or cut selected frequencies to improve the sound; dynamic range compression, which allows engineers to increase the overall gain of the system or channel without exceeding the dynamic limits of the system; routing facilities, to send the signal from the mixer to another device, such as a sound recording system or a control room; and monitoring facilities, whereby one of a number of sources can be routed to loudspeakers or headphones for listening, often without affecting the mixer's main output. Some mixers have onboard electronic effects, such as reverb. Some mixers intended for small venue live performance applications may include an integrated power amplifier.

## Mixer Applications

There are several applications for mixers. Some are made for multipurpose while others are specific to a particular task.

**Live Application**

While it’s possible for small venues that a band will have very minimal amplification, such as a four piece rock band (drums, bass, guitar, singer) where the drums are not amplified and the guitar player and bass player have amps and the singer a PA system. In larger settings everything has to be amplified and mixed together in order for it to sound good to the audience. To do this, a mixer is required.

**Recording Applications**

In the early days of recording the number of tracks was very limited. Often only a two-track was used. Most of those early recordings were essentially live recordings in a controlled environment without an audience. Once mutitrack became more ubiquitous, the recordings would be separated into at least a track per instrument and vocal. Often there would be multiple tracks per instrument by using multiple microphones placed in different locations to pick up on different sonic qualities. Many modern mixing boards include a USB interface that can be used to record directly into a DAW**.**

## DAW

DAW is an acronym for Digital Audio Workstation. A DAW is essentially a computer with recording software loaded on it and an audio interface. A DAW with an interface allows you to record digital audio onto a computer. Some examples of common DAW software are ProTools (the industry standard), Cubase, Logic Pro X, Reason, Studio One, Reaper and Garage Band

## DAW Mixers

DAW Mixers are virtual and are only limited by the computer hardware and software limitations. A DAW mixer can often have hundreds of tracks available for use. Logic Pro X for example has 256 tracks available. DAW mixers are often made to mimic the look of a hardware mixer.

## Mixer Busses

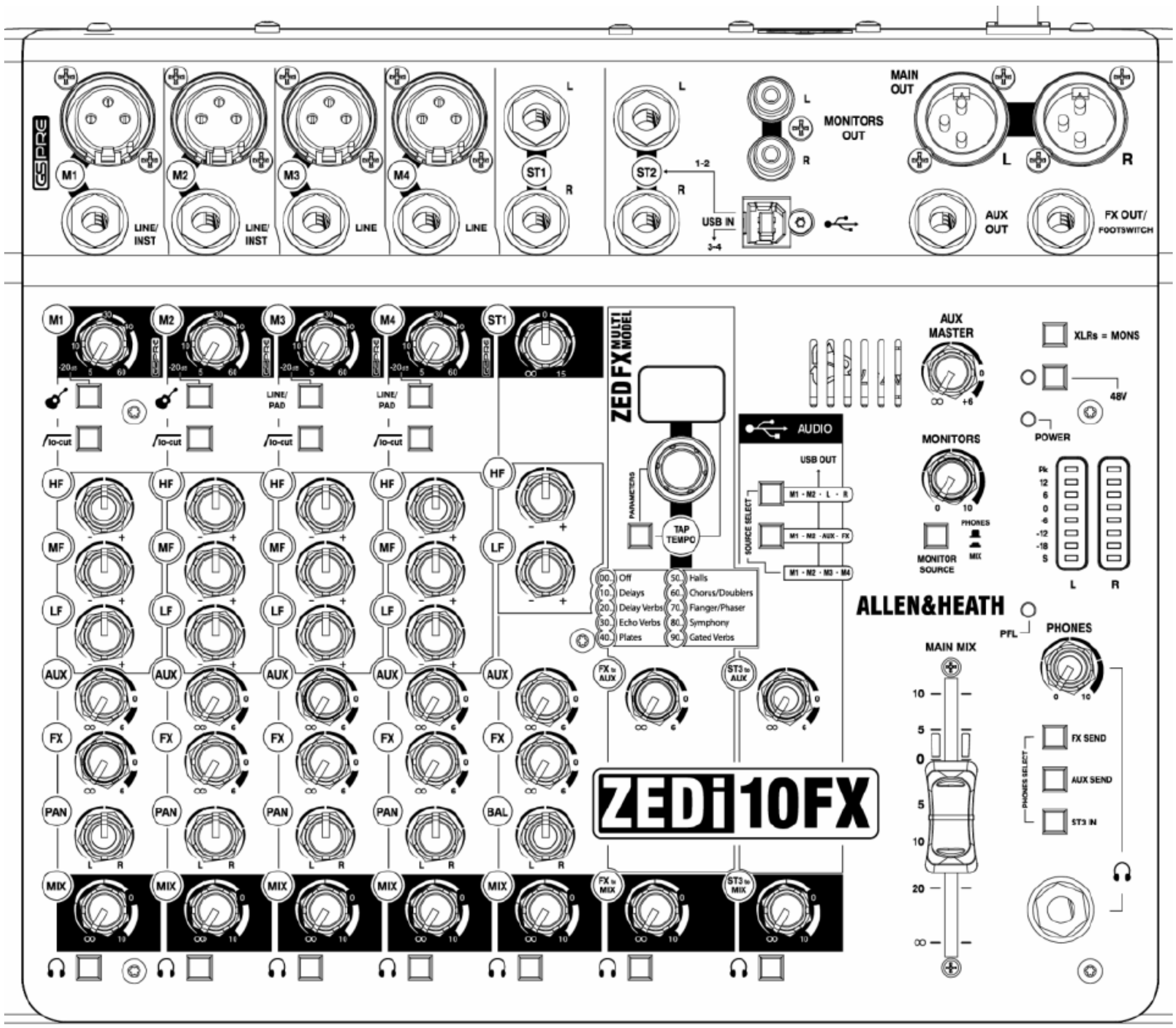
A bus is merely a signal path. The signal path that allows each channel/track on the mixer to add to the mix is often referred to as the Main Bus. Mixers often include one or more busses besides the main bus. The purpose of extra busses is to process signals, for monitoring and/or for recording. A physical mixing console may distinguish busses by use such as FX (for effects) and AUX (auxiliary) busses.

## DAW Busses

DAWs have virtual busses and many more than are usually available on a commercial mixing console. In a professional DAW like Logic Pro X, you have 256 available busses. Within the virtual track there is the ability to apply effects, which is akin to an FX send on a physical console.

# Allen & Heath ZEDi 10FX

The Allen & Heath ZEDi 10FX mixing board is a 10-channel mixer with four mono tracks (M1-M4) and three stereo tracks (ST1-ST3). The third stereo track is the 3-4 return from the USB.




This is just a little mixer for mixing a small band in a small venue or for doing some home recording. In addition to the Main bus, it has a single AUX bus and a single FX bus. In addition to onboard effects, it has an effects out jack with return option on ST2. It's USB interface allows for four tracks out and four tracks returned which makes it a great mixer for doing home recording.

### Standard Small Mixer Controls

| | |
|---|---|
| **Input** | XLR (Mic), TS (Instrument) |
| **Gain** | Gain increases the signal level. Certain input sources like guitars and vocals need more gain than line level inputs like keyboards. |
| **HF** | This adjusts the High Frequency of the signal. |
| **MF** | This adjusts the Mid Frequency of the signal. |
| **LF** | This adjusts the Low Frequency of the signal. |
| **AUX** | This adjusts the level of signal that is sent to the AUX bus. |
| **FX** | This adjusts the level of signal that is sent to the FX bus. |
| **PAN or BALANCE** | This adjusts the level of signal that is sent to the right or left channel of the MAIN bus. Pan is for Mono tracks, Balance is for Stereo tracks |
| **MIX** | This adjusts the amount of signal that is sent to the MAIN bus. |

### Mixer Busses

| | |
|---|---|
| **Main Bus** | Routes to the Main Out XLR Out Jacks. Controlled by the “Mix” knobs on each track and by the “Main Mix” fader. |
| **AUX Bus** | Routes to the AUX Out. Controlled by the “AUX” knobs on each track and the “AUX Master” knob. |
| **FX Bus** | Routes to the FX Out. Controlled by the “FX” knobs on each track |

### USB Interface Configuration for ZEDi 10FX

This particular mixer's USB interface allows for 4 channels to be routed to the DAW and 4 channels to be returned from the DAW.

### In Source:

The return channels are stereo channels **1-2** and **3-4**. Channels **1-2** route to **ST2** and channels **3-4** route to **ST3**. Anything plugged into the ST2 jacks overrides the 1-2 USB signal.

### Out Source:

The signals out are determined by the setting of the **USB OUT SOURCE** buttons. The following are the options for the **USB Out** channels. The "L" and "R" output is the Main Bus output.

| | | | | |
|---|---|---|---|---|
| **1** | **M1** | **M2** | **L** | **R** |
| **2** | **M1** | **M2** | **AUX** | **FX** |
| **3** | **M1** | **M2** | **M3** | **M4** |

To record a guitar, vocal and piano, you could use configuration 1. The important thing to remember is to zero out the main mix faders on M1 and M2 or they will be included in the piano track which is the main mix.

## Monitoring

When recording tracks there are a couple things that the musician should be able to hear:

- Previously recorded tracks and/or a rhythm click track.
- The current instrument(s) and or vocals that are being recorded.

### Monitoring Options for ZEDi 10FX

- Headphones plugged into mixer's headphone jack.
- Headphones connected to a headphone amp that's connected to the RCA **MONITORS OUT**. This option allows for multiple headphones to be used.
- Powered monitors connected to the RCA **MONITORS OUT**
- Headphones connected to a headphone amp that's connected to the **AUX OUT**. This option allows for multiple headphones to be used.
- Powered monitors connected to the **AUX OUT**

## Monitors Out Select Source

**Headphones** When selected the monitors receive the same signals as the monitoring source selected for the mixer headphones.

**Mix** When selected the monitors receive the main out signal.

## Headphone Select

**Main Mix** When no buttons are selected the signal to the headphones is the Main Bus.

**FX Send** When this is selected the signal to the headphones is the FX Bus.

**AUX Send** When this is selected the signal to the headphones is the AUX Bus.

**ST3 In** When this is selected the signal to the headphones is the ST3 channels. This is the return from the USB for channels 3-4.

## Monitoring the previously recorded tracks and or a click track

In the DAW, set the output of the recorded tracks or click track to **3-4** which routes to **ST3**. Set the **Headphone Select** to monitor the **ST3** signal.

## Monitoring the instrument being recorded

Turn up the **AUX** signal output of the track being recorded for the **AUX bus**. This will send the signal to the **AUX Bus** in addition to the signal being sent out to the DAW via USB. Set the Headphone Select to monitor the **AUX Bus** signal in addition to monitoring the **ST3** signal.

# Prepare DAW For Recording

## Create a project

You may be prompted to create either an **Audio Track** or **MIDI Track**. There may be other options beyond those two.

| | |
|---|---|
| **Set Tempo** | In BPM (Beats Per Minute) |
| **Set Key** | Set the key of the song. |
| **Set Time Signature** | Beats Per Measure |

## Recording an Audio Track

Most all modern DAW applications allow you to record both MIDI and Audio. Audio is what you'd chose to record vocals, acoustic guitar, electric guitar, electric bass guitar, etc. With MIDI instruments you basically just play the notes on a controller and it records which notes, when and how hard you played the note. With audio, you need to be concerned with the input levels and the gain and a lot of other factors.

## MIDI verses Audio

When you create a track in a DAW, you're often given several choices for what type of track to create. The two most prominent choices are MIDI and Audio.

## MIDI

A MIDI track is not audio, but is a set of instructions that indicate which notes are to be played, at what level and when they are to be played. It also sends information about note modulation like pitch bend and control changes. MIDI is recorded using a MIDI controller or it is programmed.

## A few of the advantages of MIDI:

**Assignable** You can easily change the sound timbre of the notes being played by switching the MIDI instrument. A piano sound can be changed to an electric piano sound or a trumpet sound or a cello sound, etc…

**Editable** Since MIDI notes are just data, it can be edited. You can add, delete and move notes. You can transpose music easily into another key or octave.

## Recording MIDI Tracks

MIDI tracks can be created by recording MIDI data generated by playing a MIDI Controller or by programming the track directly in via a MIDI editor.

## MIDI Controllers

MIDI controllers are primarily keyboards, but there are also other types of controllers like MIDI Guitar Controllers, Sax-Like MIDI Wind Controllers, MIDI Drum Kits, Pad Controllers and more. There are dedicated MIDI Controllers (keyboards that make no sound of their own, but just transmit MIDI data), and non-dedicated MIDI Controllers which in addition to sending MIDI data, also receive MIDI data and make sounds.

### Keyboard Controller

### Surface Pad Controller

### Wind Controller

### Keytar Controller


### Guitar Controller

### Clip On Controller

## MIDI Instruments

MIDI Instruments are virtual instruments that can be played by MIDI files/tracks or a MIDI controller keyboard. There are endless MIDI instrument choices. Most DAW apps come stock with an array of MIDI instruments to choose from. There are also premium instruments that you can purchase with sounds like pianos, electric pianos, bass guitar, guitar, horns, strings, woodwinds, percussion, and many more. There are even virtual synthesizers with programmable controls

# MIDI Plug-in Technologies

## VST

Virtual Studio Technology is an audio plug-in software interface that integrates software synthesizers and effects units into digital audio workstations. VST and similar technologies use digital signal processing to simulate traditional recording studio hardware in software. Thousands of plugins exist, both commercial and freeware, and many audio applications support VST under license from its creator, Steinberg.

**There are three types of VST plugins:**

- **VST Instruments** generate audio. They are generally either Virtual Synthesizers or Virtual samplers. Many recreate the look and sound of famous hardware synthesizers. Better known VST instruments include Discovery, Nexus, Sylenth1, Massive, Omnisphere, FM8, Absynth, Reaktor, Gladiator, Serum and Vanguard.
- **VST Effects** process rather than generate audio—and perform the same functions as hardware audio processors such as reverbs and phasers. Other monitoring effects provide visual feedback of the input signal without processing the audio. Most hosts allow multiple effects to be chained. Audio monitoring devices such as spectrum analyzers and meters represent audio characteristics (frequency distribution, amplitude, etc.) visually.
- **VST MIDI** effects process MIDI messages (for example, transpose or arpeggiate) and route the MIDI data to other VST instruments or to hardware devices.

## AU

Audio Units (AU) are a system-level plug-in architecture provided by Core Audio in Apple's macOS and iOS operating systems. Audio Units are a set of application programming interface (API) services provided by the operating system to generate, process, receive, or otherwise manipulate streams of audio in near-real-time with minimal latency. The version at the time of this writing is Version 3 and the plug-ins are referred to as **Auv3 Plug-ins**

## IAA

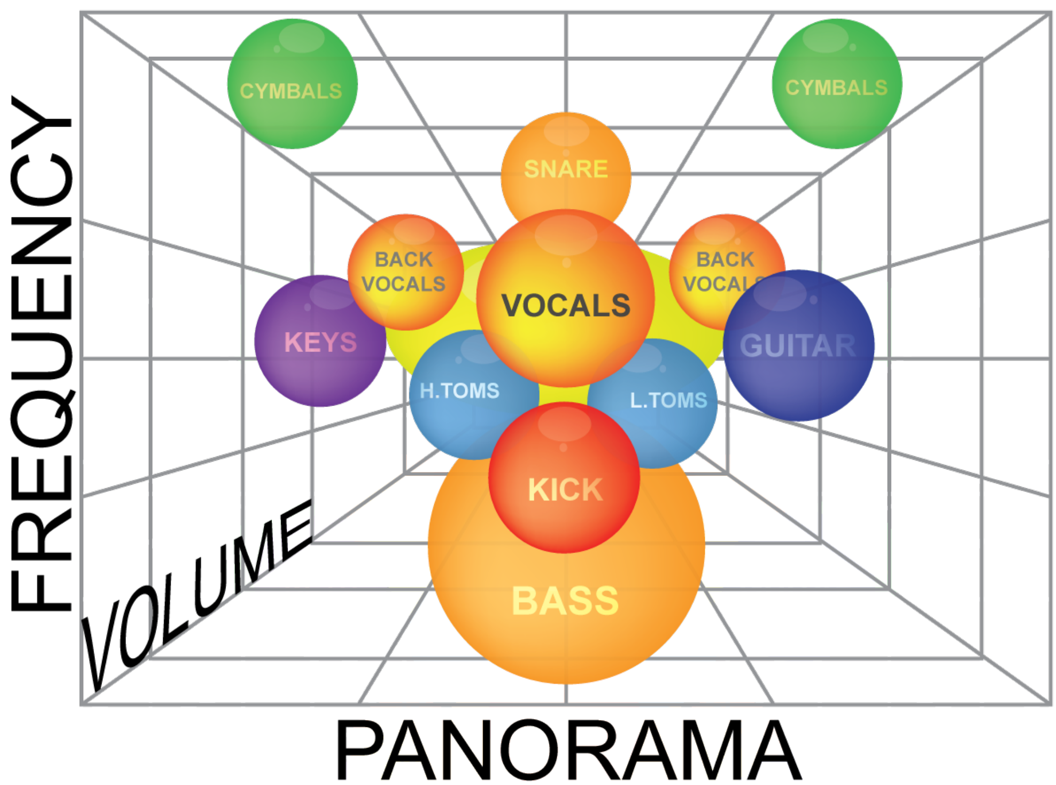
Inter-App Audio (IAA) is a technology developed by Apple Inc and designed to route audio and MIDI signals between various applications within devices based on the iOS mobile operating system. The technology was first introduced in 2013 in the seventh version of iOS.

## Recording Order

There are no rules on the order in which you record the tracks for a song, but it is important to make sure they all sync together. Whatever instrument you start with should be played to a rhythm track of some sort, even if it's just a click track.

## Layer Arrangement

When arranging the instrumentation for a song, there are a few important guidelines to creating a good mix. The art of mixing music , is creating a space within the soundscape where each instrument can exist without clashing with other instruments. A good arrangement fills the soundscape without conflicting instruments competing in the same sonic space. The image below shows how the instruments fit into the three dimensional soundscape.



# Frequency Of Music Notes

The following equation gives the frequency f of the nth key, as shown in the table:

$$f(n) = \left(\sqrt[12]{2}\right)^{n\ 49} \times 440Hz$$

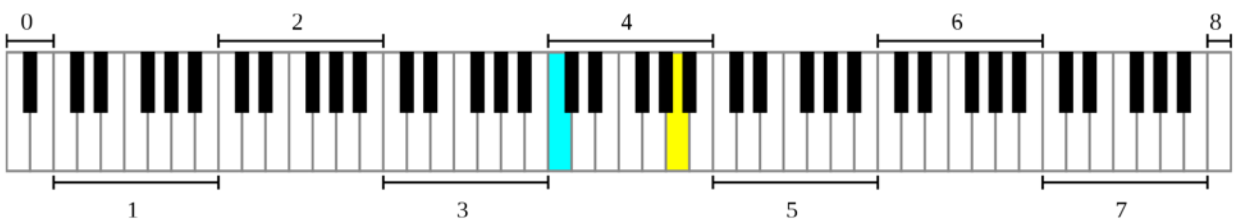
(a' = A4 = A440 is the 49th key on the idealized standard piano)

Alternatively, this can be written as:

$$f(n) = 2^{\frac{n\ 49}{12}} \times 440Hz$$

Conversely, starting from a frequency on the idealized standard piano tuned to A440, one obtains the key number by:

$$n = 12\log_2\left(\frac{f}{440Hz}\right) + 49$$




## Note Frequencies

A 108-key piano that extends from C0 to B8 (Low Notes to High Notes). A standard 88-key keyboard ranges from A0 at 27.5 Hz to C8 at 4.1 kHz.

| | | | |
|---|---|---|---|
| C0 Double Pedal C | 16.3516 | C1 Pedal C | 32.7032 |
| C♯0/D♭0 | 17.32391 | C♯1/D♭1 | 34.64783 |
| D0 | 18.35405 | D1 | 36.7081 |
| D♯0/E♭0 | 19.44544 | D♯1/E♭1 | 38.89087 |
| E0 | 20.60172 | E1 | 41.20344 |
| F0 | 21.82676 | F1 | 43.65353 |
| F♯0/G♭0 | 23.12465 | F♯1/G♭1 | 46.2493 |
| G0 | 24.49971 | G1 | 48.99943 |
| G♯0/A♭0 | 25.95654 | G♯1/A♭1 | 51.91309 |
| A0 | 27.5 | A1 | 55 |
| A♯0/B♭0 | 29.13524 | A♯1/B♭1 | 58.27047 |
| B0 | 30.86771 | B1 | 61.73541 |

| | | | |
|---|---|---|---|
| C2 Deep C | 65.40639 | C5 Tenor C | 523.2511 |
| C♯2/D♭2 | 69.29566 | C♯5/D♭5 | 554.3653 |
| D2 | 73.41619 | D5 | 587.3295 |
| D♯2/E♭2 | 77.78175 | D♯5/E♭5 | 622.254 |
| E2 | 82.40689 | E5 | 659.2551 |
| F2 | 87.30706 | F5 | 698.4565 |
| F♯2/G♭2 | 92.49861 | F♯5/G♭5 | 739.9888 |
| G2 | 97.99886 | G5 | 783.9909 |
| G♯2/A♭2 | 103.8262 | G♯5/A♭5 | 830.6094 |
| A2 | 110 | A5 | 880 |
| A♯2/B♭2 | 116.5409 | A♯5/B♭5 | 932.3275 |
| B2 | 123.4708 | B5 | 987.7666 |
| C3 | 130.8128 | C6 Soprano C (High C) | 1046.502 |
| C♯3/D♭3 | 138.5913 | C♯6/D♭6 | 1108.731 |
| D3 | 146.8324 | D6 | 1174.659 |
| D♯3/E♭3 | 155.5635 | D♯6/E♭6 | 1244.508 |
| E3 | 164.8138 | E6 | 1318.51 |
| F3 | 174.6141 | F6 | 1396.913 |
| F♯3/G♭3 | 184.9972 | F♯6/G♭6 | 1479.978 |
| G3 | 195.9977 | G6 | 1567.982 |
| G♯3/A♭3 | 207.6523 | G♯6/A♭6 | 1661.219 |
| A3 | 220 | A6 | 1760 |
| A♯3/B♭3 | 233.0819 | A♯6/B♭6 | 1864.655 |
| B3 | 246.9417 | B6 | 1975.533 |
| C4 Middle C | 261.6256 | C7 Double high C | 2093.005 |
| C♯4/D♭4 | 277.1826 | C♯7/D♭7 | 2217.461 |
| D4 | 293.6648 | D7 | 2349.318 |
| D♯4/E♭4 | 311.127 | D♯7/E♭7 | 2489.016 |
| E4 | 329.6276 | E7 | 2637.02 |
| F4 | 349.2282 | F7 | 2793.826 |
| F♯4/G♭4 | 369.9944 | F♯7/G♭7 | 2959.955 |
| G4 | 391.9954 | G7 | 3135.963 |
| G♯4/A♭4 | 415.3047 | G♯7/A♭7 | 3322.438 |
| A4 A440 | 440 | A7 | 3520 |
| A♯4/B♭4 | 466.1638 | A♯7/B♭7 | 3729.31 |
| B4 | 493.8833 | B7 | 3951.066 |

| | |
|---|---|
| C8 Eighth octave | 4186.009 |
| C♯8/D♭8 | 4434.922 |
| D8 | 4698.636 |
| D♯8/E♭8 | 4978.032 |
| E8 | 5274.041 |
| F8 | 5587.652 |
| F♯8/G♭8 | 5919.911 |
| G8 | 6271.927 |
| G♯8/A♭8 | 6644.875 |
| A8 | 7040 |
| A♯8/B♭8 | 7458.62 |
| B8 | 7902.133 |

The reason for listing the frequencies of each note is because it's good to be able to translate a note range into a frequency range for audio mixing. It's also a good idea to keep in mind with broad range instruments like a piano, is to perhaps know what range it's part within the song is playing.

## S A T B

In music, **SATB** is an acronym for **Soprano**, **Alto**, **Tenor** and **Bass**, defining the voice types required by a chorus or choir to perform a particular musical work. Pieces written for **SATB** (the most common combination, and used by most hymn tunes) can be sung by choruses of mixed genders, by choirs of men and boys, or by four soloists.

**SATB** can to also refer to ensembles of four instruments from the same family, such as saxophones (soprano, alto, tenor and baritone) or recorders.

## Operatic Voice Classification

Within the operatic systems of classification, there are six basic voice types. The ranges given below are approximations and are not meant to be too rigidly applied.

| | |
|---|---|
| **Soprano** | the highest female voice, being able to sing C4 (middle C) to C6 (high C), and possibly higher. |
| **Mezzo-soprano** | a female voice between A3 (A below middle C) and A5 (2nd A above middle C). |
| **Contralto** | the lowest female voice, F3 (F below middle C) to E5 (2nd E above Middle C). Rare contraltos possess a range similar to the tenor. |
| **Tenor** | the highest male voice, B2 (2nd B below middle C) to A4 (A above Middle C), and possibly higher. |
| **Baritone** | a male voice, G2 (two Gs below middle C) to F4 (F above middle C). |
| **Bass** | the lowest male voice, E2 (two Es below middle C) to E4 (the E above middle C). |

## Musical Instrument Frequency Ranges

Approximate Frequency Range of Vocals and Musical Instruments

**Vocal**

| | |
|---|---|
| Soprano | 250 Hz – 1.0 kHz |
| Contralto | 200 Hz – 700 Hz |
| Baritone | 110 Hz – 425 Hz |
| Bass | 80 Hz – 350 Hz |

**Woodwind**

| | |
|---|---|
| Piccolo | 630 Hz – 5.0 kHz |
| Flute | 250 Hz – 2.5 kHz |
| Oboe | 250 Hz – 1.5 kHz |
| Clarinet (Bb or A) | 125 Hz – 2.0 kHz |
| Clarinet (Eb) | 200 Hz – 2.0 kHz |
| Bass Clarinet | 75 Hz – 800 Hz |
| Basset Horn | 90 Hz – 1.0 kHz |
| Cor Anglais | 160 Hz – 1.0 kHz |
| Bassoon | 55 Hz – 575 Hz |
| Double Bassoon | 25 Hz – 200 Hz |
| Soprano Saxophone | 225 Hz – 1.0 kHz |
| Alto Saxophone | 125 Hz – 900 Hz |
| Tenor Saxophone | 110 Hz – 630 Hz |
| Baritone Saxophone | 70 Hz – 450 Hz |
| Bass Saxophone | 55 Hz – 315 Hz |

**Brass**

| | |
|---|---|
| Trumpet (C) | 170 Hz – 1.0 kHz |
| Trumpet (F) | 300 Hz – 1.0 kHz |
| Alto Trombone | 110 Hz – 630 Hz |
| Tenor Trombone | 80 Hz – 600 Hz |
| Bass Trombone | 63 Hz – 400 Hz |
| Tuba | 45 Hz – 375 Hz |
| Valve Horn | 63 Hz – 700 Hz |

**Strings**

| | |
|---|---|
| Violin | 200 Hz – 3.5 kHz |
| Viola | 125 Hz – 1.0 kHz |
| Cello | 63 Hz – 630 Hz |
| Double Bass | 40 Hz – 200 Hz |
| Guitar | 80 Hz – 630 Hz |
| E. Guitar | 80 Hz – 1.2 kHz |
| E. Bass | 40 Hz – 400 Hz |

**Keyboards**

| | |
|---|---|
| Piano | 28 Hz – 4.1 kHz |
| Organ | 20 Hz – 7.0 kHz |

**Percussion**

| | |
|---|---|
| Celeste | 260 Hz – 3.5 kHz |
| Timpani | 90 Hz – 180 Hz |
| Glockenspiel | 63 Hz – 180 Hz |
| Xylophone | 700 Hz – 3.5 kHz |

**Drums**

| | |
|---|---|
| Kick | 50 Hz – 450 Hz |
| Floor Tom | 80 Hz – 100 Hz |
| Toms | 100 Hz – 600 Hz |
| Snare | 120 Hz – 250 Hz |
| Cymbals | 3.0 kHz – 5.0 kHz |

30 RECORDING

# MIXING

Mixing is the process that you do after all the recording is done and before you Master the project. Mixing is the art of carving out sound to allow instruments in a mix to fit together.

## Pre-Mix Setup

| | |
|---|---|
| **Make a Session Copy** | Do not work on the originals. |
| **Tweak the Track Timing** | Look up the section on Adjusting Timing. |
| **Check the Fades** | Check to be sure that nothing is getting cut off too soon. Check each fade-out on elements such as background vocals or doubled instruments to be sure the release time is the same. |
| **Eliminate Noises** | Although noises might not sound too bad when played with all of the tracks in the mix, something that is obscured pre-mix and pre-master can come out in the master. Also by eliminating all of the extraneous noise, the tracks will sound cleaner and more distinct. |
| **Trim Heads and Tails** | Add fade-ins and fade-outs to eliminate any edit noises. |
| **Cross-Fade Your Edits** | Forgetting to cross fade edits can result in clicks and pops in the mix . |
| **Comp Your Tracks** | While comping shouldn't be done in the mixing stage, it should occur in the overdub tracking stage, if there is somewhere where it needs to be done, now is the time to do it. |
| **Tune Your Tracks** | If there are any notes out of tune, you can use pitch-correction plugins like Auto-Tune, Elastic Audio. |
| **Decide What Stays and** | In the digital age of recording we're not as limited as the days of a four- |

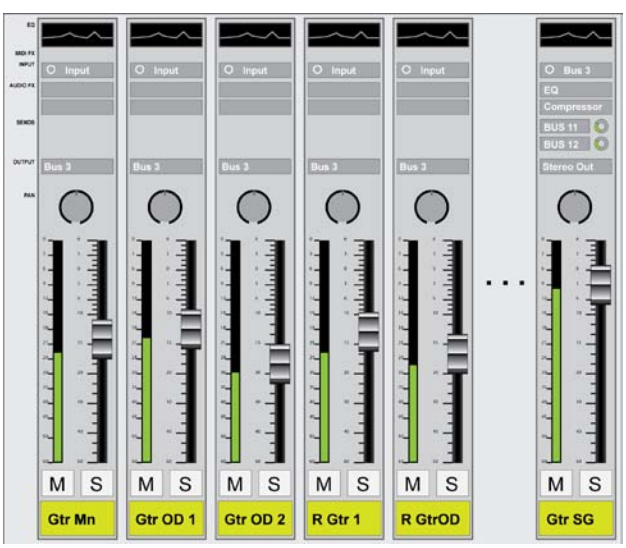
| | |
|---|---|
| **What Goes** | track recording system. It's easy to build up a lot of tracks with endless overdubs and takes. At some point you need to try and keep the good and toss out the bad and mediocre. |
| **Delete Empty Tracks** | Remove any empty tracks. Often in the process of recording a lot of tracks will be created that are never used. |
| **Deactivate and Hide Unused Tracks** | Often in the process of recording, some tracks are created to create another track. For example, a midi track used to create an audio track. While it might be a good idea to hold onto the track in case you want to change the audio track, it has no use in the mix so it's a good idea to deactivate it and hide it. |
| **Reorder Tracks** | Reorder tracks into logical groupings such as grouping similar instruments together. |
| **Color-Code Tracks** | This just makes it easier to see which tracks are in which groups. |
| **Correctly Label Tracks** | Be specific. What instrument, what part of song, who's playing. If necessary, use abbreviations or acronyms to help it fit within the space available. |
| **Insert Section Markers** | Time markers, sometimes called memory locations. Example of time markers might be: Intro, Verse 1, Chorus 1, Verse 2, Chorus 2, etc... |
| **Create Groups and Subgroups** | Subgroups are a way to pre-mix a number of channels before they're sent to the main mix. This not only allows you to control the level of several channels with a single fader, but also to be able to apply EQ and effects to those channels as a group. |

# Creating Subgroups

When recording a song, the number of tracks with multiple overdubs and parts can add up and become hard to manage. Before mixing, it's a good idea to organize the tracks by type of track. It also helps visually to color code the tracks.

## Creating Subgroups

A subgoroup is where you set up a bus as the subgroup destination and then set the output for each member of the subgroup to that bus. In the following illustration, there are five tracks in the subgroup that route to Bus 3 which is labeled “Gtr SG” for Guitar Subgroup.



After the levels have been adjusted in each track of the subgroup, the levels and any insert effects on the bus track are collectively controlled on the bus track. To make things more manageable, you can hide the subgroup members leaving only the subgroup track.

### Groups:

Groups aka Console Groups are different from subgroups. A group is a series of tracks linked together in the DAW to act as one. Generally groups work well with stereo instruments such as piano and organ Leslies, guitars recorded with multiple mics, drum overheads and the like. Subgroups work best with large groups of instruments such as the drum kit, background vocals, and string sections.

### Groups within Subgroups

It is possible to create groups within subgroups.

### DAW Tracks

A track in a DAW is virtual and so are it’s busses and bus sends. There are many options for routing signals. The limitation is just the limits defined by the DAW software.

| | |
|---|---|
| **Input** | Audio, MIDI, Bus |
| **Sends** | Busses. Multiple busses with send levels on each. |
| **Output** | Main Bus (Stereo Out), Bus |

## Effects and Processing

An effect is a device that treats the audio in some way, then adds it back to a dry or untreated version of the sound. Echo and reverb are obvious cases, and you can use pitch-shift and pitch modulation in a similar way. Processors, by contrast, generally are those devices that change the entire signal and don't add in any of the dry signal. Things like compressors and equalizers fall into this category. Processors can often be used as effects in their own right, or as part of an effect chain.

#### Delays, Echoes and Reverb

Time-based effects include delays, echoes and reverbs. At their core, they all basically do the same thing: capture an input signal, delay it slightly, then play it back. The timing and varying complex processing of the resulting signal(s) is the main thing that differentiates them.

The terms **delay** and **echo** often refer either to a single repeat of a sound or a series of repeats utilizing feedback (a control where some of the output signal is routed back to the input) to produce a series of repeats, each one sometimes slightly degraded in quality from the one before.

**Short Delays:** Short delays of approximately 50 ms (milliseconds) or less cannot be perceived by the human ear as separate sounds. Instead, we hear them as a kind of room ambience or a doubling of the input signal.

**Medium Delays:** Slightly longer delays (in the 75 to 250 ms range) with little or no feedback are often called slap-back. This refers to a tape-based technique developed by legendary engineer Sam Phillips back in the 1950s.

**Long Delays:** Long delays of 250 ms or greater are perceived as discrete repetitions of a sound. These can be used to create polyrhythms in the music, or to add emphasis to specific words being sung.

A subset called **tap delays** simulate the action of analog tape-based units that utilized multiple playback heads spaced a few inches apart for the creation of multiple discrete echoes, not just a single echo being repeated. These are useful for creating wild new rhythms and layers and are commonly used in dub and techno genres.

**Tips & Tricks: Delay**

**Tempo Delay:** Most plug-in and hardware delays now allow you to automatically sync delay times to MIDI clock and then specify the interval of the repeats in terms of note values rather than milliseconds. A trick here is to use two simultaneous tempo-based delays with, say, a triplet delay setting, panned hard left, and a straight-note delay panned hard right. Things can get more interesting still if you apply this technique using ping-pong delays, so that alternate repeats bounce from one side of the stereo spectrum to the other. To create a true 3D effect, play around with the amount of original signal left in the middle. Depending on the intervals between your repeats, you can turn simple guitar and synth lines into complex, arpeggiator-like patterns or totally spaced out ambient pieces.

**Ostentatious Delays:** If you're making very rhythmic music of any kind, it makes sense to use tempo-synced delays, to avoid undermining the main pulse. However, simple tempo-synched delays tend to be masked by the main rhythmic stresses, so they sink into the background of the mix unless mixed very high in level, which makes it difficult to create ostentatious delay effects in rhythmic music without swamping your mix. One solution to this problem, very common in trance music, is to set a delay to a three-16th-note duration, which means that although the delay repeats never step outside the 16th-note grid, they'll often miss the main beats and therefore remain clearly audible.

**Keep It Reel:** Perhaps because a humble tape echo was the first effect I ever owned, delay has always been my primary effect. Whether to liven up repetitive loops or add apparent complexity to simple solos, it's worth getting to grips with delay the old-fashioned way. This means daring to switch off MIDI sync and manually setting delay time, driving feedback to the brink of madness, or routing the pure delay output through equalizers, filters and so on.

**Non-synced Delay:** We are so used to perfectly synced delays that it's easy to forget that manual sync and a pair of ears has a charm all of its own. Even delay times that bear no obvious relationship to the tempo can add dynamic movement and feel to a track: check out some early King Tubby if you need reminding of this.

**Subtlety:** You don't always have to make longer echo or delay effects obvious in the mix for them to be effective. Once you've set up the delay times and panned them to suit your song, try dropping the delay levels until you scarcely notice them during most of the mix (listening on headphones often helps set the most suitable level). This generally results in intriguing little ripples of repeats that you notice at the end of verses or during pauses, that add interest and low-level detail to the mix.

### Reverb

Early reverb chambers, plates and springs have now given way to digital solutions, which fall into two main groups: synthetic and convolution. Synthetic reverbs take an algorithmic approach, setting up multiple delays, filters and feedback paths to create a dense reverberation effect similar to what you might hear in a large room. Though these often sound a bit 'larger than life', they've been used on so many hit records that we now tend to accept their sound as being the 'correct' one for pop music production. Most can approximate the sound of rooms, halls, plates and chambers, but in comparison with a real reverberant environment, the early reflections often seem to be too pronounced. The advantage of a synthetic reverb is that the designer can give the user plenty of controls for altering the apparent room size, brightness, decay time and so on.

### Convolution Reverb

In recent years, convolution reverbs have become both affordable and commonplace. These differ from synthetic reverbs insomuch as they work from impulse responses (or IRs), recorded in real spaces to faithfully recreate the ambience at the microphone's position when the IR was made. Sometimes these are referred to as sampling reverbs but there's no sampling involved as such, even though the process seems akin to sampling the sonic signature of a room, hall or other space.

Because IRs can be recorded in virtually any space, convolution reverbs generally come with a library of IRs ranging from small live rooms to famous venues, top studio rooms, forests, canyons, railway stations and just about anything else you can think of. They sound very convincing, and there's plenty of variety to be had, but once the IR is loaded, there's only a limited amount of editing you can do without spoiling the natural sound. Usually you can apply EQ and also change the envelope of the reverb decay to make it shorter, and adding pre-delay is not a problem, but after that you pretty much have to take what you get. Some companies, such as Waves, have managed to create additional controls but, as a rule, the further you move from the original IR, the less natural the end result.

Ironically, the sound of certain synthetic reverbs is now such an established part of music history that most convolution reverbs come with some IRs taken from existing hardware reverb units or from old mechanical reverb plates. Also, if you have a convolution reverb, it is worth checking the manufacturer's site, as additional IRs are frequently available for download.

All serious reverb units have a stereo output to emulate the way sound behaves in a real space and, in the case of convolution models, the IRs are often recorded in stereo, using two microphones. Some surround reverbs are also available.

Reverb creates a sense of space, but it also increases the perception of distance. If you need something to appear at the front of a mix, a short, bright reverb may be more appropriate than a long, warm reverb, which will have the effect of pushing the sound into the background. If you need to make the reverb sound 'bigger', a pre-delay (a gap between the dry and wet signals) of up to 120ms can help to do this without pushing the sound too far back, or obscuring it.

Though reverb increases the sense of stereo width, it dilutes the sense of stereo position. If you want to pinpoint the placement of something in a mix, you should consider using a mono rather than a stereo reverb, and panning this to the same place as the dry sound.

Most synthetic reverbs allow you to balance the level of the early reflections and the later, more dense reverb tail. If you want to keep the sense of space but without the reverb tail taking up too much space in your mix, you can increase the early reflection level and reduce the tail level.

As a rule, you don't add much, if any, reverb to low-frequency sounds, such as bass guitar or kick drums. Where you need to add reverb to these sources, short ambient space emulations usually work better than big washy reverbs, which tend to make things sound muddy. Taking this a step further, you can also make a mix sound less congested by EQ'ing some low end out of your reverbs.

**Modulation**

Modulation effects such as tremolo, vibrato, flanging, phasing, and chorusing utilize a component called a Low Frequency Oscillator, or LFO for short. These effects differ in terms of how the LFO is used, as follows:

**Tremolo:** If the LFO is used to vary the amplitude (volume) of a signal, the result is tremolo – a trembling effect that makes a sound more rhythmic, percussive or stuttering. If it's used to vary the pitch of a signal, the result is vibrato.

**Flanging:** If the LFO is used to periodically change the timing of a slightly delayed signal (in the 0.1 - 10 ms range), the result is flanging. If it instead sweeps a series of notch filters (components that sharply reduce extremely narrow frequency ranges), the result is phasing. Both flangers and phasers typically provide a feedback control, which intensifies the effect; at extreme settings you can get uncontrolled "howling" noises. Flangers and phasers are both instantly recognizable for the 'swooshing' and swirling effects they create. As you increase the speed of the LFO, they often take on a ‘watery' quality.

**Chorus:** If the LFO is utilized to periodically change the pitch of the incoming signal and shifting the timing of delays, you end up with chorusing. As its name implies, it's like the effect of hearing a real-life choir, with many people singing. Even the best trained choral singers don't hit every single note exactly at the same time, or exactly on key, which is why the end result is a glorious spread of sound, with the slight changes in pitch and timing adding richness. Stereo chorus effects split an incoming signal in two and use the LFO to slightly shift the panning of each channel, further adding to the envelopment. Applied to electric piano and arpeggiating guitars, chorusing creates the impression of two instruments playing the same part – the embodiment of the classic '80s sound.

**Distortion / Saturation**

Distortion / saturation effects typically simulate the sound created by overdriven analog components such as vacuum tubes, transistors, and magnetic tape. Though they are commonly used for adding grit (with saturation being the more subtle of the two), either of these effects can fatten up thin sounds and/or help them better cut through a dense mix when applied in small doses. A little bit applied to a bass, 808 kick, or lead vocal works wonders to make the part stand out. Just be sure to not be too heavy-handed and overdo it; if everything in your mix is too harmonically rich, there's a chance nothing will stand out! Read here to learn more about using analog-style harmonic distortion in a mix.

**Amp Modeling**

Amp modeling plugins can provide accurate emulations of the world's best-loved guitar amps and speaker cabinets. These kinds of plugins can come in very handy during mixing. The most obvious applications are to reamp the DI signal of a keyboard or bass, or to rescue a badly recorded guitar part – even one that's been played on an acoustic guitar – but amp modeling plugins can also help mask other problems such as clipping. Waves GTR3 Amps offers many guitar and bass amps, including clean, overdriven and high gain models.

So don't use flanging when phasing will work better. Don't use distortion when saturation is called for. Don't use tremolo when what you really want is vibrato. Always choose the effect that best complements the music.

**How To Connect Effects**

Reverbs and time-based effects like delays should almost always be accessed via an aux send and return, with their Mix controls set to 100 percent. This not only allows you to route multiple signals to a single effect but also ensures that only effected "wet" sound arrives at the aux return. You can then blend the desired amount in with the unaffected "dry" sound to taste.

Other kinds of effects like saturation, tremolo or flanging should usually be accessed via channel or buss inserts – this way the sound can be effected completely without the added gain or potential phase issues by blending an effected duplicate. But, depending on the situation, running effects in parallel can certainly be used to yield creative results if that is what's desired.

**Do Not Pile Them On To Every Track**

Yes, effects are cool. Yes, they sound great. And there are literally thousands of them out there. Which is exactly why many newbie’s fall into the trap of using them on way too many tracks.

Experienced mixing engineers often strive to add the least amount of processing. To use a well-worn metaphor: mixing is like cooking, and any chef will tell you that few dishes are improved by tossing in dozens of different seasonings. The same goes for effects. A good rule of thumb here is: if you're adding effects to more than five or six tracks during mixing, there's a good chance you're overdoing things. Take the effects off a few of them, and focus on balancing the sounds instead. Your listening audience will thank you for it.

**Don’t Limit Just One Effect Per Track**

Okay, so you've decided to heed our advice and add effects to just a handful of the tracks you're mixing. That doesn't mean that each of those tracks should receive just one effect each, however. In fact, you can often get better – and certainly more interesting – results by chaining two or three effects together.

For example, try these chains:

- Flanger or phaser before (or after) delay / echo.
- Delay or echo into tremolo.
- Pitch-shifter (set to just a few cents) before distortion, saturation, or amp model… or vice versa.
- Pitch-shifter before delay. (As a bonus, if you then feed some of the output of the delay back to the input of the pitch shifter, you can create delays that keep climbing or falling in pitch as they recirculate.)

Adding effects to reverb sends or returns can yield some amazing results. Flanging, phasing, chorusing, and harmonizing all work well for this. The intermingling of effects with dynamics processing can yield some very cool sonics, too. For example, apply a gate or expander with a medium attack time before a delay. This will chop off transients and make the delayed sound much smoother and easier to blend in.

**Synchronize Time-Based Effects**

Most modern echo plugins allow you to automatically sync repeat times to the song's tempo, then specify the interval of the repeats in terms of note values rather than milliseconds. This is a powerful tool, and one that should usually be employed whenever the tracks you're mixing were recorded to a click track or aligned to the grid in your DAW; fail to sync them and you'll be undermining the main rhythmic pulse of the song.

Another cool trick is to apply two simultaneous synchronized echoes to a sound – one with, say, a triplet setting panned hard left, and a half-note panned hard right. Route the original dry signal up the middle and experiment with varying amounts of "dry" versus "wet" signal – you can even automate those changes to create a constantly shifting soundstage. Or route the echoes to a pitch shifter to turn simple guitar or synth lines into complex arpeggiator-like patterns.

Notice, however, that we said that sync should just be employed most of the time. That's because echo times that bear no obvious relationship to the tempo have a charm of their own and in certain circumstances can add a nice random "feel" to a mix. The key here is experimentation. Often times, offsetting a BPM-synced delay by just a few milliseconds will yield results that sound perfectly less-than robotic, but not so completely random that they feel out of place.

**Don’t Put “echo-echo-echo-echo” On Anything From Start to Finish**

Here's another common mistake made by mixing newbie’s: applying an echo to a sound – any sound, but most especially lead vocals, all the way through a song. Do that and you're guaranteed to: a) end up with a sloppy mess that obliterates all the other components in your mix, and b) bore the listener to tears.

Instead, apply echo to just individual notes to give them emphasis, or to individual words within a vocal line as a way of 'underlining' certain lyrics; these are often called delay throws. (In the analog world, send faders were "thrown" to selectively send audio to an outboard delay unit.) An easy way to set this up in your DAW is to automate a track's send to a delay on an aux track, leaving the send volume down when delay is not desired and moving the send volume up over the waveform of the sound or word you'd like to highlight.

The classic example of how to do this tastefully and effectively is embodied in the Pink Floyd song "Us and Them" from their mega-gazillion selling, uber-multi-platinum Dark Side of the Moon album. The first line of every verse always consists of three, and only three syllables ("Us and them"/"Me and you"/"Black and blue," etc.). And only the first and third syllables ever receive echo, specifically seven whole-note repeat echoes that run the full course of the two bars of relative emptiness after each syllable, each at a slightly lower level than the one before. (The sole exception is the opening line of the final verse: "With, without." Guess which word does not receive the echo?)

If that same echo had been applied to Gilmour's lead vocal from start to finish, it would have been a gimmicky distraction. Instead, this selective usage of an effect becomes an integral part of the song, a piece of ear candy that immediately grabs the listener and keeps their attention. There's a powerful lesson to be learned here!

## Effect Chain Order

The order of your plugins in an effects chain does matter. A plugin will affect the sound differently depending on its position along the effects chain. Knowing how the order of plugins influence sound can help you make mix decisions and troubleshoot problems.

## Signal Processing Order

**1 Gain Staging**

Gain staging is managing levels at each stage of the signal path. Maintaining the gain structure gives your mix sufficient headroom and dynamic range for mixing. It also ensures the audio signal flow is at an optimum level without clipping as it passes through various processors and mixer stages.

Inserting a gain plugin first in the chain allows you to adjust your levels before further processing. You can also insert a gain plugin anywhere needed along the effects chain to prevent clipping. Avoid running a "hot" signal into a plugin. Overloading a plugins input signal will give you poor results.

**2 Saturation**

Saturation adds presence, character, warmth, excitement, punch, and cohesion. Driving sounds through tubes, transistors, and circuitry has long been the key ingredient in achieving analog sounding mixes.

Saturation enhances sounds by adding even and odd order harmonics. These harmonic frequencies make a digital-sounding mix sound full, fat, and warm. It also helps sound translate on small speakers that can't reproduce lower frequencies like earbuds, laptops, and phones.

Inserting a saturation plugin before an EQ allows you to cut or minimize unwanted harmonic frequencies created by saturation. However, it's also suitable to saturate after EQing.

**3 Subtractive EQ**

Subtractive EQing removes problematic frequencies, creates clarity, and adds presence. The goal is to clean up the sound and minimize problematic frequencies that might get boosted by the compressor.

Use a parametric EQ to cut unwanted sub frequencies and harsh resonances. Avoid boosting in this first stage of equalization. It's often better to boost with a second EQ after making corrective moves with subtractive EQing and compression.

**4 Compression**

Compression smoothes out the dynamic range of a sound to maintain constant levels. It also adds loudness, cohesiveness, punch, and helps shape the tone. However, used incorrectly can reduce the impact of the sound, so only apply compression when needed.

This first stage of compression focuses on corrective work. Use a compressor to control dynamic range and catch the loudest peaks. Inserting the compressor after a subtractive EQ prevents unwanted frequencies from triggering the compression.

5 **Additive EQ**

After doing corrective work with subtractive equalization and compression, try shaping the tone with additive equalization. Tonal equalization adds presence and character. For example, it's common to boost the high end to add presence in pop and electronic music.

The best way to shape tone is with an analog modeled EQ. Analog modeled equalizers emulate the circuitry and musical character of their classic hardware counterparts. Analog modeled EQs also add, warmth, presence, punch, and sheen to your music.

It's common practice to use a combination of digital and analog-style EQs. For example, digital EQs are excellent for corrective work because they are precise, flexible, and transparent sounding. Whereas analog-modeled EQs are ideal for tonal work and sweetening. They excel at emphasizing or attenuating certain frequency bands in a broad, musical manner to achieve clarity and punch.

6 **Modulation**

After achieving an ideal frequency balance and dynamic range, add modulation effects if needed. Common modulation effects include chorus, flanger, phaser, tremolo, stereo width, and auto-pan. These creative audio effects add movement, depth, width, and character to sounds.

It's common to insert creative effects after corrective work in post-production. Some prefer to get the basic sound down before modulating or adding ambiance. However, it's also common to apply modulation effects before corrective effects during the sound design process.

Again, there are no rules. It might sound better to insert a modulation effect before the subtractive EQ and compressor. This workflow is logical if you want to clean up the sound and tame dynamics after creative processing. Use your ears!

7 **Reverb and Delay**

Reverb and delay effects are typically inserted towards the end of the chain or on return tracks. These time-based effects add a sense of space, dimension, and fullness to your mix. They also provide that integral polish that immerses listeners.

The order of these two effects will also make a difference. For example, when using delay and reverb together, it's common to insert the delay before the reverb. This way the reverb doesn't wash out the sound before it hits the delay.

**8 Limiting**

Limiters are essential dynamic processing tools used for mixing and mastering. At a basic level, limiters reduce the peaks in a waveform and prevent them from exceeding a digital ceiling. The result reduces the dynamic range of an audio signal and increases perceived loudness.

Insert a limiter towards the end of your effects chain to reduce excessive transients, control dynamics, glue sounds together, boost levels, and add power. However, not every sound requires limiting. Only apply a limiter if it's needed.

**9 Sidechaining**

Sidechain compression creates separation between elements and helps them punch through the mix. It also minimizes phasing and frequency masking between the two elements.

A sidechain compressor uses an external sound source to quickly reduce the volume of a sound. For example, sidechaining a bass track will lower the bass volume every time the kick strikes. This technique allows the kick to cut through the mix more clearly. It will also give the kick more of an impact.

Applying sidechain compression last in the effects chain also helps control reverb and delay tails. In addition, it gives you more flexibility to adjust the amount of sidechaining without affecting the signal input of other effects plugins.

**10 Volume**

Inserting a gain plugin last in the chain allows you to adjust your levels further if needed. It's also recommended to insert a gain plugin at the end of the effects chain if you plan on using volume automation.

Automation is an essential process that helps make your music sound more compelling, exciting, and dynamic. It's better to apply automation using a gain plugin instead of the tracks volume fader. This method keeps the volume fader free to make further gain staging adjustments.

## Automating Effects and Processing

The automation features in DAWs give users comprehensive control over virtually every aspect of their mixes. Not just volume and pan, but also effects parameters, sends, and mutes can be controlled and precisely edited with automation.

### Automation Modes

Every DAW implements automation somewhat differently, but there are fundamental concepts that are common to virtually all of them, including these four automation modes: Read, Touch, Latch and Write. There are a couple of different ways to apply automation. You can draw it in manually on a track, or you can record it into your DAW in real time using an on-screen fader or an external controller.

### Touch Mode

Touch is excellent for making tweaks because as soon as you release the fader, knob or button, it returns the automation to its previous level or position.

### Latch Mode

Use Latch mode for changes to stay at the new level for extended periods. Let's say you have a synth part that you want to make louder from the third chorus until the end of the song. You wouldn't want to use Touch mode, because that would require that you hold the fader in the new position until the end of the song. In that scenario, Latch mode would be the better choice, because all you'd need to do is push the fader to the new level at the start of the third chorus and then release it.

### Write Mode

Use Write mode to erase and redo a whole track of automation. With Write mode activated, your DAW will start erasing and rewriting as soon as you hit play. Be careful, as it will wipe any automation data that the transport passes over (of the parameter you're controlling on the channel you're working on).

### Read Mode

This is the most passive mode. It simply responds to the automation.

### Volume Automation

Although you can control virtually any parameter, volume is probably the one you'll spend the most time automating. Volume automation gives you precise control of the levels of all your tracks, allowing you to program adjustments on any track in any part of the song. Let's look at some of the things you can do with it:

**Master Bus Automation**

Automate master buss volume to tweak your song's dynamics. You can change the intensity of any song section by automating the volume of the master buss. If the chorus or bridge isn't jumping out enough, push everything up a couple of dB. Keep the changes on the subtle side to avoid sounding unnatural. You want it to simulate what it would sound like if the musicians and singers dug in a little more (or less) on a given song section. Subtle master buss volume automation adjustments can add dynamic variation to your song.

**Automate Track Volume**

Automate a track so it pokes out around another. Sometimes you'll have an instrument that needs to be accentuated in spots and then come back down, such as fills around vocal lines. If the instrument in question isn't popping through enough, use Touch mode (or automation editing) to push it up in the spaces.

One of the most common applications of this technique is with drum fills. You can make them really pop by pushing them up and then back down after the fill ends. You'll either have to create a drum sub master (or VCA) and do your automation on its volume, or create an edit group of the drum tracks so that the they'll respond together and the automation will be written to each of them. You'd be surprised how much more energy you can create that way. The drum fills are accentuated by pushing up the overall drum level during the fills, via a drum sub-buss channel.

The drum fills are accentuated by pushing up the overall drum level during the fills, via a drum sub-buss channel.

**Humanize Programmed Parts**

Make programmed parts more dynamic. Volume automation is helpful for humanizing parts that were programmed, not played. Add dynamics similar to what a musician would do if playing the part, or just humanize it by adding sporadic (slight) changes to the volume.

**Automation vs. Compression**

Use more automation, less compression. While it's tempting to insert a compressor on a track when you want to get its dynamics under control, it's easy to over-compress and make an instrument or vocal part sound less natural. Consider compressing more lightly and using volume automation to handle much of the dynamic heavy lifting. If you have the time to devote to it, you can even out the dynamics of virtually any part, only with automation.

### Fix Plosives

Fix plosives with volume automation. Popped consonants, especially "p" and "b" sounds can be reduced using volume automation. Zoom in until you see the plosive in the waveform, then draw in a fast volume-automation fade under the plosive (see image below). You might have to experiment with the angle of the fade until you get the results you want.

### Automating Effects and Virtual Instruments

In addition to controlling volume, you can use automation in useful and creative ways to control effects parameters.

### Automate Plugin Parameters

Increase control over your plugins. You can usually automate almost any of your plugins' parameters, although, depending on your DAW, you may first need to configure which ones to automate. Effects parameter automation opens up a lot of creative options. The possibilities are endless.

### Tempo-Synched Pan Effects

Create tempo-synced auto-pan effects. Most DAWs allow draw-in automation using different waveshapes or geometric shapes, creating a repeating pattern. If you're automating pan automation on a stereo track, for example, drawing in shapes constrained to the rhythmic values related to the song's master tempo can create some very cool auto-pan effects. You'll need to put the track's snap on to the rhythmic value that you want before drawing in the pan. From there you can easily custom-manipulate rhythmic changes to go with the composition.

### Automate Effect Sends

Automate your effects sends. Want to make the reverb more intense on the lead vocal in the chorus? You could automate the effects-send level from the vocal track to a higher amount in the choruses and lower in the verses. Or, how about automating the pan control on an effects return to create a wider reverb in one song section for added drama.

### Automate EQ

Automate your EQ. Adding highs to a track can help bring it further forward in the mix and make it sound more exciting. If you want to, say, make the vocals a tad more exciting on the choruses, you could automate a small high-end boost during choruses and bring it back during verses.

**Automation Editing Tips**

You might use your DAW's mixer faders for writing initial automation passes, but you'll likely find yourself doing a lot of manual editing of the automation data. Although break-point automation gives you a lot of control, it can be awkward to work with. It's very easy to, for example, edit one section and accidentally affect the levels somewhere else in the song. If you delete the breakpoint, you'll see the line collapse to zero in the rest of the song.

**Here are some tips:**

Learn your DAW's break-point editing idiosyncrasies. Every DAW implements its automation editing a little differently. Besides just drawing lines, you need to learn how to raise the level of a selected section of automation without affecting the rest, add and delete breakpoints, and switch between drawing tools if you want to edit successfully.

Try the freehand drawing tool if your fades sound unnatural. You might be tempted to use the straight-line tool when drawing in a fade, but you'll probably find it to sound too perfect because it's so straight. Often you get a better sounding fade out with a curve, or with little bit of irregularity in the line you draw. It's sometimes hard to see accidental changes in automation levels, especially if you have the track height too low. The track on top is actually a duplicate of the one below, but the actual volume change in the automation looks lower because the track height is reduced.

Watch for accidental level changes. If you're editing one end of an automation line, it's easy to accidentally make the side you're working on lower or higher than the end, causing a slow ramp up or down over time in the parameter you're automating. The further you're zoomed out on the track, and the lower the track height, the less you'll notice these differences. Especially with volume, be very careful that a track that's supposed to start and end at the same level, actually does. Typically, your DAW will display the level numerically when you select a breakpoint, and you can use that to check that both ends of the automation are in the right place.

## Create Effects Channels

Many mixers have a standard set of go-to effects that they'll set up before they begin a mix since they know that at some point or another they'll be needed in the mix.

## Two Reverb One Delay Setup

The following is a quick and easy setup that works well using two reverbs and a delay:

| | |
|---|---|
| **Drums** | Use a reverb set to dark room sound with about 1.5 seconds of decay and a pre-delay of 20 milliseconds. |
| **Other Instruments** | Use a plate with about 1.8 seconds of decay and a pre-delay of 20 milliseconds. |
| **Vocals** | Use a delay of about 220 milliseconds with two repeats. |

## Two Reverb Two Delay Setup

Another common setup uses two reverbs and two delays:

| | |
|---|---|
| **Short Reverb** | A room program with the decay set from 0.5 to 1.5 seconds of decay with a short pre-delay timed to the track. |
| **Long Reverb** | A plate or hall program with a decay set from 1.5 to 4 seconds of decay and a pre-delay of as little as 0 or as much as 150 milliseconds timed to the track. |
| **Short Delay** | A delay of about 50 to 200 milliseconds. |
| **Long Delay** | A delay from about 200 to 400 milliseconds. |

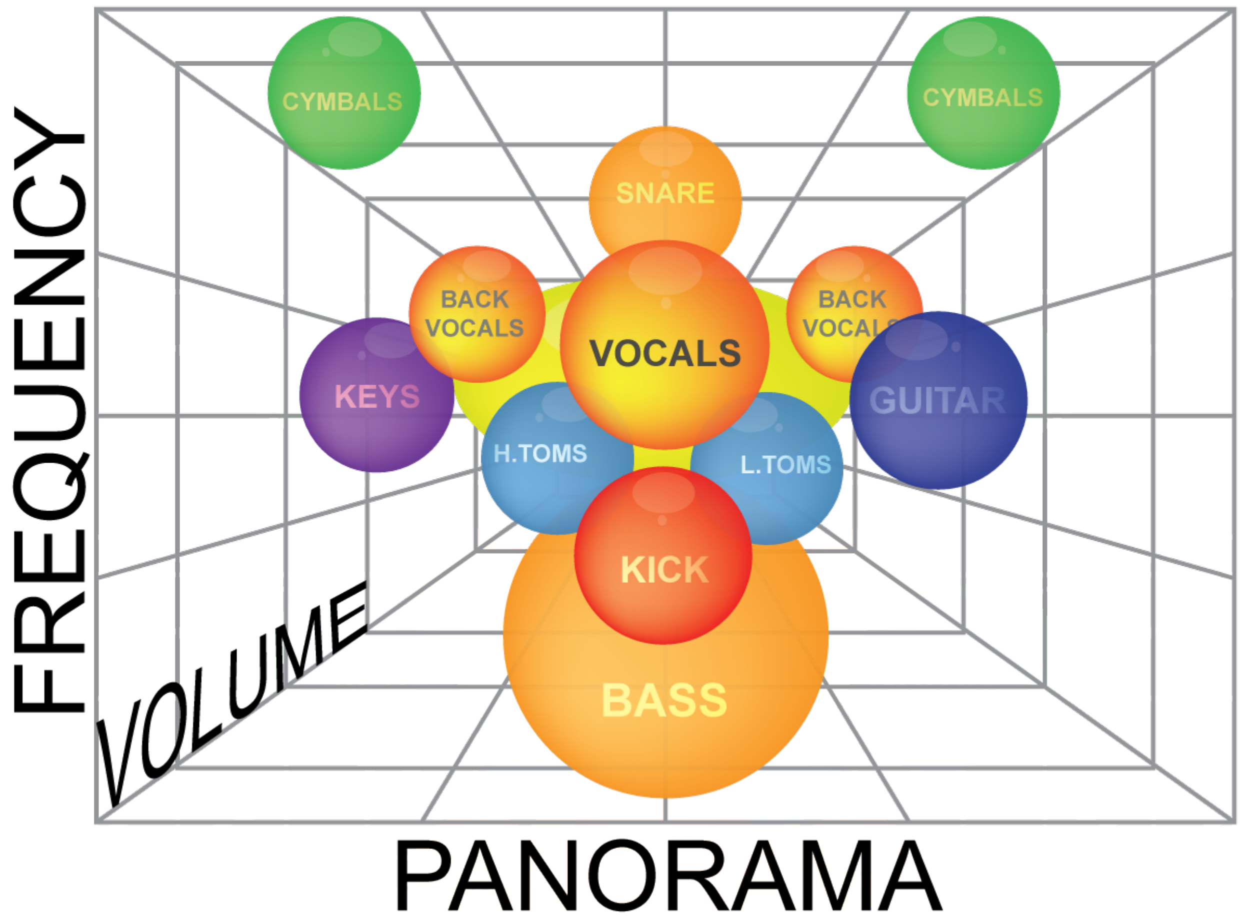
## Assign Channels

Assign to the effect channels tracks that will use certain effects. Such as the case where the drums or snare will use a short reverb. It's best to assign those channels to the appropriate sends and pan them accordingly, but make to that the send is set to off before mixing begins.

## Insert Compressors and Limiters

In most modern mixes at least a few channels, such as the kick, snare, bass and vocal, will usually need a compressor during the mix in order to modify the track's dynamic range.

# THE SOUNDSCAPE




## Tall, Deep and Wide

**Tall** The vertical dimension, which is governed by frequency, is the least variable and the most difficult to control.

**Deep** The front-to-back dimension, governed by ambience, volume and brightness is less variable and is trickier to control.

**Wide** The horizontal plane, which you adjust with your pan pots offers the most variability and is the easiest to control.

Mixing is the process of fitting each instrument of the mix into this three dimensional soundscape without any of them existing in the same space at the same time.

## Elements of a Mix:

| | |
|---|---|
| **Balance** | The volume level relationship between musical elements. |
| **Frequency Range** | Having all audible frequencies properly represented. |
| **Panorama** | Placing a musical element in the sound-field. |
| **Dimension** | Adding ambiance to a musical element. |
| **Dynamics** | Controlling the volume envelope of an individual track or the entire mix. |

## The Arrangement

When your track has more than a few instruments, learn to avoid clutter and create space in your mix. By using filtering, EQ and other ways to organize the sound field, you'll get everything to shine through the mix with definition and clarity.

One of the prime directives when mixing is to make sure each vocal and instrument track can be clearly heard. Accomplishing that can be challenging if your song has more than just a few tracks. If this is the case, you'll have to use more than just panning to achieve separation.

In this article, we'll look at some of the techniques you can use for separating elements in a mix. We'll describe various methods for placing elements in the soundstage, and talk about ways to reduce clutter, which is the enemy of a good mix.

### Think in 3D

It's beneficial to envision your mix is as if you're sitting in front of a stage where all the various instruments and vocals are located. Each element will be positioned not only horizontally, but also front to back, and bottom to top. Those three dimensions are not equally variable or controllable:

### Pan tracks in the same frequency range apart

If you have two elements that are centered around the same frequency range, it's best to try to pan them apart from each other. If you had, say an acoustic rhythm guitar and a keyboard that are both playing chords in the same register, they'd sound a lot more distinct if they were in different positions in the left-right spectrum. The opposite is also true: Instruments that inhabit different frequency ranges will not obscure each other as much when panned together. Think about these concepts when you're making panning decisions.

## Reduce width of stereo elements

A stereo instrument often sounds fantastic on its own, or in a simple arrangement such as a song with acoustic guitar and vocal. But when combined with a bunch of other tracks in a busy mix, its width might do more harm than good. Often, stereo instruments fit better if you reduce their width (by panning them toward one side or another). The narrower they are, the easier it will be to find a unique place where they'll shine through. Too many stereo instruments in the mix can cause sonic clutter as they all overlap each other. Another benefit to converting stereo elements to mono is better mono compatibility for your entire mix.

## Create depth with loudness and frequency

One of the critical factors in how your ears perceive the distance of a sound is its volume. Try this: snap your finger right in front of your face, and then extend your hand out as far as your arm will reach and snap it again similarly. The farther-away snap sounds quieter and less present. The same principle applies to your mix. Tracks that are louder and brighter seem as if they're more up front, and the converse is also true. If you're trying to, say, make a guitar part feel like it's further back, turn it down, or roll off some of its high end.

## Create depth with ambience

A sound that you hear from a distance sounds more ambient than one you hear close up because it contains a higher ratio of reflected sounds to direct ones. The farther away the sound source is from your ear, the more room reflections you'll hear. Therefore, adding reverb or delay, which simulate room reflections, make a track seem to move backward in the mix. The higher the ratio of reverb to direct sound, the farther back it will seem.

## Use pre-delay to highlight a vocal or instrument

Imagine being in a large room like a gym. If you clap your hands, there's a delay before you hear the echo of that clap, due to the time it takes for the sound waves to bounce back to your ears. Virtually every reverb processor, like H-Reverb, Abbey Road Reverb Plates, Manny Marroquin Reverb and others, offer a pre-delay parameter, which simulates that natural delay.

The brief duration of the pre-delay time allows a direct sound to cut through the mix more, because the beginning of that sound is dry before the reverb kicks in. Optimal pre-delay times depend on the tempo of the song, how much reverb you're applying and the reverb's decay time setting. Experiment with those variables to find the right settings for a given situation.

## Use automation to make an instrument pop

You can use automation to make an element pop out of the mix periodically. For example, if you've got an arpeggiated synth line under a vocal, you could automate it to come up in volume between the vocal lines. The ability to precisely record and edit automation in a DAW gives you a lot of control over volume along with many other variables.

## Define height with frequency

The greater the pitch of a sound, the higher it appears to be in your speakers. So when you're arranging your mix soundstage, factor in frequency as another way to locate and separate elements. To use an extreme example, if you have a piccolo and an upright bass, the former will sound higher than the latter. This principle is also useful for choosing which instruments can be layered in the same panning position without masking each other (see #9).

## Carve out frequencies to create space

The term "frequency masking" refers to two elements inhabiting the same frequency range in a mix and thus obscuring each other to some degree. Panning is the best cure for it, but if you have to keep the tracks near each other in the panorama that masking occurs, you can also use EQ to differentiate them. The idea is to boost one track in a particular frequency range while cutting the other one by the same amount in the same range.

Make sure to choose frequencies that are flattering for the track you're boosting. For example, let's say you have a piano and acoustic guitar that are both playing in the same octave and masking each other. You could increase the upper midrange frequency (for example 3.5 kHz) on the acoustic to give it a little extra sparkle and cut the piano there. Then you could boost the piano at a lower-midrange frequency (say 650 Hz) and reduce the guitar similarly.

## Unmask the kick with sidechain compression

Most bassists try to play on the same beats as the kick drum whenever possible. When the two instruments lock in, it sounds tight and helps the groove. But it can cause some masking issues in the mix because the kick and bass are in the same frequency range and are just about always panned straight up the middle. Fortunately, you can use sidechain compression to help the kick poke out from the bass when they hit together. Here's how it works: You set up a compressor on the bass track using its sidechain mode, and for the sidechain source, you assign the kick drum. That way, every time the kick plays, the compressor will attenuate the bass by a small amount, allowing the kick to come through more.

## Arrange your song with mixing in mind

There's some truth to the saying that a well-arranged song will almost mix itself. If you think about the principles discussed here, particularly frequency masking, it makes sense that if you arrange your music so that similar instruments play in different registers as much as possible, you won't have to work as hard to find space for all of them.

For example, if you have an electric rhythm guitar and a piano playing chords simultaneously, try to make sure they're not playing primarily in the same frequency range. Perhaps you could move the piano part up an octave, or have the guitar play higher chord inversions. Try to envision in advance how various parts are going to fit into the mix.

## Use high-pass filtering to reduce clutter

Most instrument and vocal tracks carry low-end information that's not needed for them to sound good and sit well in the mix. Those frequencies can cumulatively add a lot of mud to your mix, negatively impacting its overall clarity.

Many engineers use high-pass filters (aka "low-cut" filters) to roll off extraneous lows. Virtually any EQ plugin provides filters. You could apply a single high-pass filter on the master buss that's set to roll off below 30 Hz. One caveat: I wouldn't suggest this approach if you're mixing dance music, hip-hop or some varieties of pop, where the sub-frequencies are essential.

A more precise way to go for any type of music is to apply the filters on individual tracks, where you can tailor them to the frequency response of the particular instrument or voice. A useful method is to turn the frequency adjustment of the filter up slowly until you hear the track start to audibly thin out, and then back it off to just before that point.

## Avoid reverb-tail muddiness

A reverb with a long decay can sound really cool when you hear it on a single instrument or voice, but multiple tracks with such effects can create a mess of overlapping reverb tails. Think about it; if you've got a reverb with, say, a three-second decay, inserted on an effects return, and you're sending multiple tracks to it, there's going to be gobs of reverb decay washing over your mix throughout the song. On a slow song it's much less problematic, but if the song is fast, you'll need to keep the decay times shorter to avoid reverb-tail clutter. It's also advisable to use the EQ or high-pass filter parameters on your reverb to cut out some of the low frequencies of the reverberated sound, which aren't usually necessary.

## Frequently check your mix in mono

Although your mix will mainly be listened to on stereo headphones/earbuds or speaker systems, there will be situations where it's played back in mono, and it's important to make sure that it doesn't fall apart when it's summed.

In mono, all the panning that you used to create separation is removed from the mix, and everything is in the same vertical position. There's no way to avoid that, and periodic mono checks might help you decide to depend a little less on panning, and more on the front-to-back plane to achieve separation. It's a creative call that depends on the musical style and where you think your mix will be listened to the most. Checking in mono is also essential for detecting phase coherence issues.

## Conflicting Instruments

How do you keep instruments from conflicting with each other? A well written arrangement keeps instruments out of the way of each other.

## How To Fix Conflict

- Mute the offending instrument so they never play at the same time.
- Lower the level of the offending instrument.
- Tailor the EQ so that the offending instrument takes up a different frequency space.
- Pan the offending instrument to a different location
- Change the arrangement and re-record the track

# Arrangement Elements

(This is not the same as Mix Elements.)

| Element | Purpose | Typical Instruments |
|---|---|---|
| **Foundation** | The instruments that provide the groove and pulse of the song | Drums and bass |
| **Pad** | Long sustaining notes that glue the mix elements together | Organ, E.Piano, Strings, Synth, Guitar Power Chords |
| **Rhythm** | The instruments that provide motion to the song | Percussion, shakers, tambourine, rhythm guitar |
| **Lead** | The focal point of the song | Lead vocal, lead or solo instrument |
| **Fill** | The instruments that fill in the spaces between the lead phrases | Solo instrument, background vocal |

### Rules for Arrangements

- **Limit the number of arrangement elements.** Usually there should not be more than four arrangement elements playing at the same time.
- **Everything in its own frequency range.** The arrangement, and therefore the mix, will fit together better if all the instruments sit in their own frequency range. For instance, if a synthesizer and rhythm guitar play in the same octave, they may clash frequency-wise and fight each other for attention. The solution would be to change the sound of one of the instruments so they fill different frequency ranges. Have one play in a different octave or have them play at different times.

### Where To Build The Mix From

**Typical starting places:**

- From the Bass
- From the Kick Drum
- From the Snare Drum
- From the Toms
- From the Overheads
- From the Lead Vocal or Main Instrument
- When mixing a string section, from highest (violin) to lowest (bass)

## Level Setting Methods

Start low as each elements will add to the overall level. Overall, leave some headroom for the mastering.

## Panorama

The stereo picture. Placing the audio in the sound field.

## Phantom Center

A phenomenon that occurs with the output of two properly placed speakers combines to create the impression of a third center speaker.

## Big Mono

Big mono occurs when you have a track with a lot of pseudo-stereo sources that are all panned hard right and hard left. This robs the track of definition and depth.

## Panning Outside the Speakers

Using the phantom sound effect that can be created using effect processing, it's possible to create the impression of sound outside of the speakers rather than from them.

## There are two ways to do this:

1. On a stereo instrument, flip the phase of one channel. The panning will now seem to be beyond the speakers.
2. On a stereo instrument, feed some of the right channel signal to the left channel out of phase, and feed some of the left channel to the right channel out of phase. This can be done by copying both channels, flipping the phase on both, panning the opposite of the way they're already panned, and gradually increasing the level. As the level increases, the sound will seem to pan outside the speakers.

# Equalizing

### The Goals of Equalization

- To make an instrument sound clearer and more defined.
- To make the instrument or mix bigger.
- To make all of the elements of a mix fit together better by putting each instrument in it's own predominant frequency range.

## Frequency Bands

| | | |
|---|---|---|
| **Sub Bass** | **16Hz - 60Hz** | Sounds that are more felt than heard.<br>Gives songs a sense of power. Too much emphasis in this range makes the music sound muddy. Attenuating this range, especially below 40Hz, can clean up a mix considerably. |
| **Bass** | **60Hz - 250Hz** | Contains the fundamental notes of the rhythm section.<br>EQing this range can change the musical balance making it fat or thin. Too much boost in this range can make the music sound boomy. |
| **Low Mids** | **250Hz - 2kHz** | Contains the low order harmonics of the most musical instruments.<br>Can introduce a telephone-like quality to the music if boosted too much. Boosting the 500Hz to 1000Hz octave makes the instruments sound horn like. Boosting the 1kHz to 2kHz octave makes them sound tiny. Excess output in this range can cause listening fatigue. |
| **High Mids** | **2kHz - 4kHz** | Contains speech recognition sounds such as "m", "b" and "v".<br>Too much boost in this range, especially at 3kHz, can introduce a lisping quality to a voice. Too much boost in this range can cause listening fatigue. Dipping the 3kHz range on instrument backgrounds and slightly peaking 3kHz on vocals can make the vocals audible without having to decrease the instrument level in mixes where the vocals would otherwise seem buried. |
| **Presence** | **4kHz - 6kHz** | Responsible for clarity and definition of voices and instruments.<br>Boosting this range can make the music seem closer to the listener. |

| | | |
|---|---|---|
| | | Reducing the 5kHz content of a mix makes the sound more distant and transparent. |
| **Brilliance** | **6kHz - 16kHz** | Controls brilliance and clarity.<br>Too much emphasis in this range can produce sibilants on the vocals. |

# EQ Methods

## Method 1: Equalize for Definition

Even well recorded material can sound lifeless. This is due to certain frequencies being over emphasized and some others being severely attenuated. More often than not, the lack of definition of an instrument is because of too much lower midrange in approximately the **400Hz - 800Hz** area. This area adds a "boxy" quality to the sound. Sometimes it's because the sound is lacking in the **3kHz - 6kHz** range that makes it sound undefined. Subtractive equalization is a method that allows you to zero in on the frequencies that are masking the definition in a sound.

1. Set the Boost/Cut control to a moderate level of cut (8 to 10 db should work).
2. Sweep through the frequencies until you find the frequency where the sound has the least amount of boxiness and the most definition.
3. Adjust the amount of cut to taste. Be aware that too much cut makes the sound thinner.

There are two spots in the frequency spectrum where the subtractive equalization is particularly effective: between **200Hz** and **600Hz** and between **2kHz** and **4kHz**. This is because most directional microphones provide a natural boost at **200Hz** to **600Hz** because of the proximity effect brought about by close miking, and many mics (especially those known for being good vocal mics) have a presence boost between **2kHz** and **4kHz**. Dipping those frequencies a few dB (more or less as needed) can make the track sound much more natural than if you were to try and add frequencies instead.

If there was a limited number of microphones (or even just one) used to record all the instruments in a home studio, these two frequency bands, or any other where there's a peak in the response, will build up as more and more instruments are added. By dipping those two frequency bands a bit, you'll find that many of the instruments can sit better in the mix without having to add much EQ at all.

**Note:** Always try attenuating (cutting) the frequency first. This is preferable because all equalizers add phase shift as you boost, which results in an undesirable coloring of sound. Usually, the more EQ you add, the more phase shift is also added and the harder it may be to fit the instrument into the mix as a result.

## Alternate Method

1. Starting with your EQ flat, remove all the bottom end below 100Hz by turning the low-frequency control to full cut.
2. Using the rest of your EQ, tune the mid-upper midrange until the sound is thick yet distinct.
3. Round it out with a supporting lower-mid tone to give it some body.
4. Slowly bring up the mud-inducing bottom end enough to move air, but not so much as to make the sound muddy.
5. Add some high-frequency EQ for definition.

## Method 2: Equalize for Size

Making a sound bigger comes from the addition of bass and sub-bass frequencies in the 40Hz to 250Hz range, although most will come from an area just below 100Hz, a region just above 100Hz or both.

To use the method, the low-frequency band of your EQ must be sweepable.

1. Set the Boost/Cut control to a moderate level of Boost (8 or 10dB should work).
2. Sweep through the frequencies in the bass band until you find the frequency where the sound has the desired amount of fullness.
3. Adjust the amount of Boost to taste. Be aware that too much Boost will make the sound muddy.
4. Go to the frequency either half or twice the frequency that you used in step 2 and add an amount of that frequency as well. Example: If your frequency in Step 2 was 120Hz, go to 60Hz and add a dB or so as well. If your frequency was 50Hz, go to 100Hz and add a bit there.

## Method 3: Juggling Frequencies

Start with the rhythm section (bass and drums). The bass should be clear and distinct when played against the drums, especially the kick and snare. You should be able to hear each instrument distinctly. If not, do the following:

1. Make sure that no two equalizers are boosting at the same frequency. If they are, move one to a slightly higher or lower frequency.
2. If an instrument is cut at a certain frequency, boost the frequency of the other instrument at the same frequency. For example, if the kick is cut at 500Hz, boost the bass at 500Hz

Add the next most predominant element, usually the vocal, and proceed as above.
Add the rest of the elements into the mix one by one. As you add each instrument, check it against the previous elements above.

## Golden Rules of Equalization

- If it sounds muddy, cut some at 250Hz.
- If it sound honky, cut some at 500Hz.
- Cut if you're trying to make things sound better.
- Boost if you're trying to make things sound better.
- You can't boost something that's not there in the first place.

## Finding an Offending Frequency

Sometimes a sound has a frequency that sticks out of the mix. The method to find it and attenuate it is similar to EQ Method 1.

1. Set the Boost/Cut control to a moderate level of boost. Eight or 10 dB should work.
2. Sweep through the frequencies until the frequency that's giving you trouble leaps out.
3. Adjust the amount of cut to taste until the offending frequency is in balance with the rest of the sound. Be aware that too much cut can also decrease the definition of the sound.

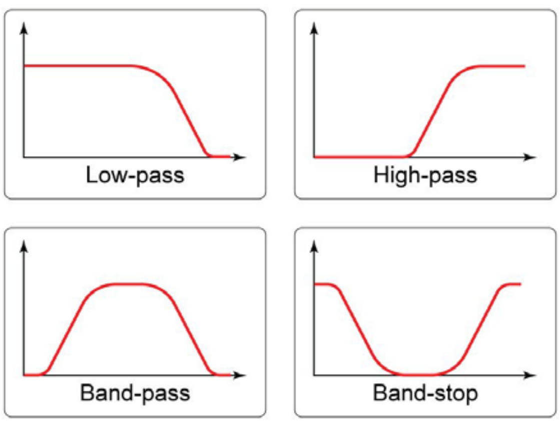
# The Magic Frequencies

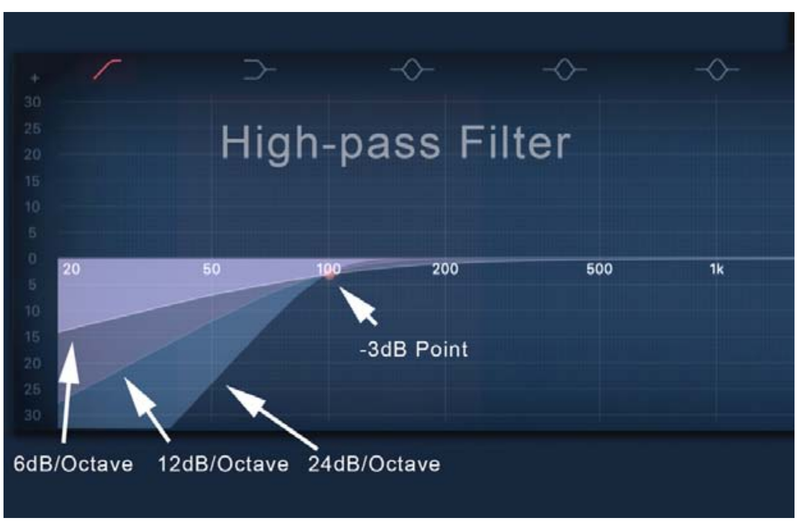
| | |
|---|---|
| **Bass Guitar** | Bottom at 50Hz to 80Hz, attack at 700Hz, snap at 2.5kHz |
| **Kick Drum** | Bottom at 80Hz to 100Hz, hollowness at 400Hz, point at 3kHz to 5kHz |
| **Snare** | Fatness at 120Hz to 240Hz, point at 900Hz, crispness at 5kHz, snap at 10kHz |
| **Toms** | Fullness at 240Hz to 500Hz, attack at 5kHz to 7kHz |
| **Floor Tom** | Fullness at 80Hz, attack at 5kHz |
| **Hi-Hat and Cymbals** | Clang at 200Hz, sparkle at 8kHz to 10kHz |
| **Electric Guitar** | Fullness at 240Hz to 500Hz, presence at 1.5kHz to 2.5kHz, attenuate at 1kHz for 4 x 12 cabinet sound |
| **Acoustic Guitar** | Fullness at 80Hz, body at 240Hz, presence at 2kHz to 5kHz |
| **Organ** | Fullness at 80Hz, body at 240Hz, presence at 2kHz to 5kHz |
| **Piano** | Fullness at 80Hz, presence at 3kHz to 5kHz, honky-tonk at 2.5kHz |
| **Horns** | Fullness at 120Hz, piercing at 5kHz |
| **Voice** | Fullness at 120Hz, boomy at 240Hz, presence at 5kHz, sibilance at 4kHz to 7kHz, air at 10kHz to 15kHz |
| **Strings** | Fullness at 240Hz, scratchy at 7kHz to 10kHz |
| **Conga** | Ring at 200Hz, slap at 5kHz |

## The Magic High-Pass Filter

Musical instruments, especially stringed instruments such as guitars produce frequencies beyond their intended notes. These frequencies are subtle, but they can build up with layering. Other instruments, like a piano for example can give off unwanted frequencies when they're being played by the sounds of the players fingers hitting the keys. The low frequencies of many instruments just get in the way of each other and don't add much to the sound. That's why, if you roll the low frequencies off below 100Hz (or even higher in some cases) on instruments other than the kick and bass, the mix usually cleans up significantly. Rolling off the low frequencies of a vocal mic can eliminate noises that are too low to distinguish, but will still muddy the mix. Even rolling off the bass and drums at between 40Hz and 60Hz can sometimes make the mix a lot louder and punchier without affecting the perceived low end.

**Note:** In the end if you solo an instrument in the mix, it may not sound right. The important thing is how it sounds in the mix. Cutting and boosting frequencies will alter the timbre of the instrument.



## Bass and Drums

One of the more difficult tasks of mixing is the balance of bass and drums, especially the bass and kick drum. To make a good mix you have to make space for both of these so they won't fight each other.

- **EQ the Kick between 60 and 120 Hz.** This will allow it to be heard on smaller speakers. For more attack and beater click, add between 1.0 kHz and 4.0 kHz. You may also want to dip out some of the boxiness that lives between 200 and 600 Hz. Eqing in the 30 to 60 Hz range will produce a kick that you can feel if your speakers are large enough, but that can also make it sound thin on smaller speakers and probably won't translate to a variety of speaker systems. Most 22-inch kick drums like to center somewhere around 80 Hz, for instance.
- **Bring up the bass with the kick.** The kick and bass should occupy slightly different frequency spaces. The kick will usually be in the 60 to 80 Hz range, whereas the bass will emphasize higher frequencies anywhere from 80 to 250 (although sometimes the two are reversed depending upon the song). Before you continue to EQ at other frequencies, try filtering out any unnecessary bass frequencies (below 30 Hz on kick and 50 Hz on the bass, although it varies according to style and taste) so the kick and bass are not boomy or muddy. There should be a driving, foundational quality to the combination of these two together. A common mistake is to emphasize the kick with either too much level or too much EQ and not enough on the bass guitar. This gives you the illusion that your mix is bottom-light, because what you're doing is effectively shortening the duration of the low-frequency envelope in your mix. Since the kick tends to be more transitory than the bass guitar, this gives you the idea that the low-frequency content of your mix is inconsistent. For pop music, it's best to have the kick provide the percussive nature of the bottom while the bass fills out the sustain and musical parts.
- **Make sure the snare is strong; otherwise, the song will lose its drive when everything else is added in.** This usually calls for at least some compression. You may need a boost at 1.0 kHz for attack, 120 to 240 Hz for fullness, and 10 kHz for snap. As you bring in the other drums and cymbals, you might want to dip a little of 1.0 kHz on these to make room for the snare. Also, make sure the toms aren't too boomy. If so, try rolling them off a bit below 60 Hz first before you begin to EQ elsewhere.
- **If you're having trouble with the mix because it's sounding cloudy and muddy on the bottom end, turn off the kick drum and bass to determine what else might be in the way in the low end.** You might not realize there are some frequencies in the mix that aren't musically

necessary. With piano or guitar, you’re mainly looking for mids and top end to cut through, while any low end might just be getting in the way of the kick and bass, so it’s best to clear some of that out with a high-pass filter. When soloed the instrument might sound too thin, but with the rest of the mix the bass will sound so much better, and you won’t really be missing that low end from the other instruments. Now the mix will sound louder, clearer, and fuller. Be careful not to cut too much low end from the other instruments, as you might lose the warmth of the mix.

- **For dance music be aware of kick drum to bass melody dissonance.** The bass line is very important and needs to work very well with the kick drum when it’s reproduced over the huge sound systems commonly found in today’s clubs. If your kick has a center frequency of an A note and the bass line is tuned to A#, they’re going to clash. Tune your kick samples to the bass lines (or vice versa) where needed.
- **If you feel that you don’t have enough bass or kick, boost the level, not the EQ. This is a mistake that everyone makes when they’re first getting their mixing chops together.** Most bass drums and bass guitars have plenty of low end and don’t need much more, so be sure that their level together and with the rest of the mix is correct before you go adding EQ. Even then, a little goes a long way.

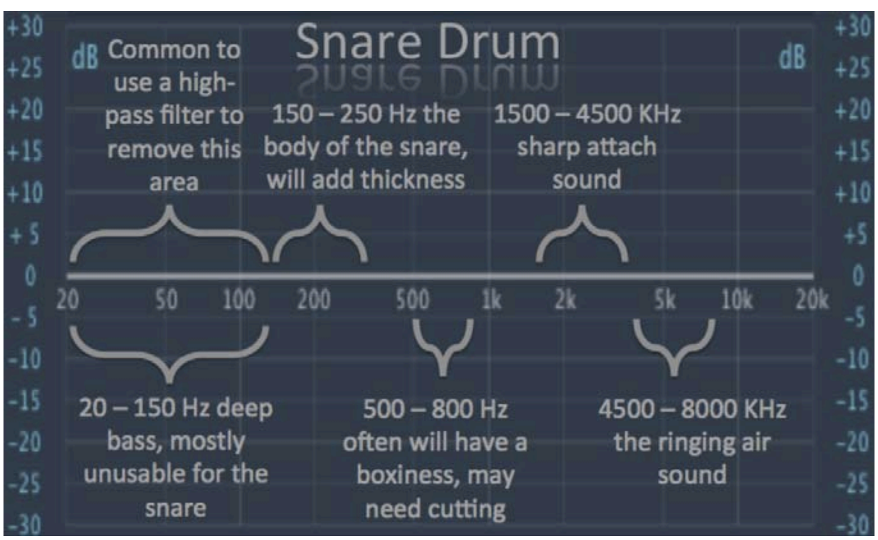
# EQ Techniques

## General Tips

- Use a narrow Q (bandwidth) when cutting and a wide Q when boosting.
- If you want a sound to stick out of the mix, roll off the bottom; if you want it to blend, roll off the top.
- The fewer instruments that are in the mix, the bigger each one should sound.
- Conversely, the more instruments in the mix, the smaller each one needs to be for everything to fit together.
- It’s usually better to add a small amount at two frequencies than a large amount at one.
- Be aware that making an instrument sound great while soloed may make it impossible to fit together with other instruments in the mix.

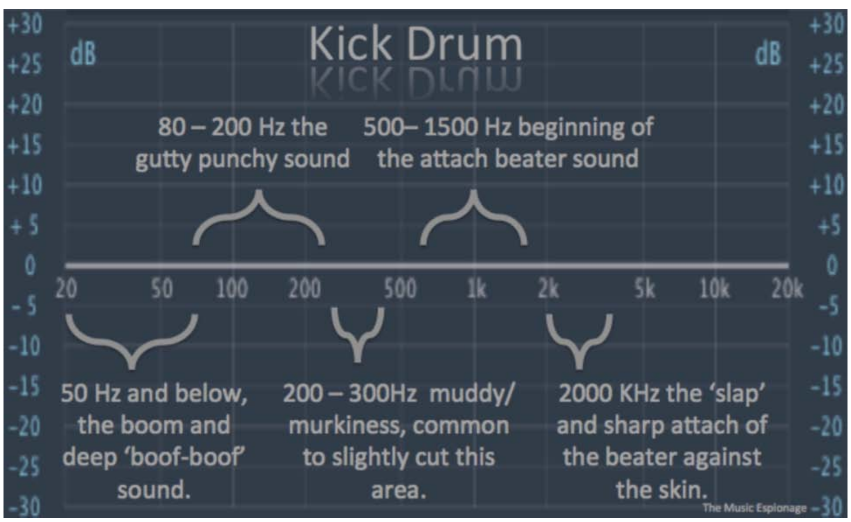
## For Snare

- To find the “point” on the snare, boost the upper midrange about 5 or 6 dB at 2 kHz or so. Open up the bandwidth (if that parameter is available) until you get the snare to jump out, then tighten the bandwidth until you get only the part of the snare sound that you want most. Then fine-tune the frequency until you need the least amount of boost to make it sit in the mix yet have some definition.



## For Kick

- Try boosting 4 kHz, cut 200 to 400 Hz, and then boost the low frequencies around 60 to 100 Hz to find the drum's resonance.
- One fairly common effect used in R&B is a trigger a 32 Hz tone with the kick, then gate it so it has the same volume envelope. Blend and compress both the original kick and the 32-Hz tone to taste.
- For a metal kick, add a bit of 3 kHz or so on the kick drum to function as the "nail in the paddle" sound.
- For a kick meant for clubs, emphasize the 200 to 300 Hz range while rolling off the extreme low end. The club system makes up the difference, so if you mix the bottom of it the way you think you'll hear it in a club, you're probably going to overload the house system.
- If your bass is a very pure sine wave-like sound and your kick is an 808, they may mask each other. If the kick is lower-sounding than the bass, add a sample with some mid or top punch. If the kick is higher than the bass, you can add some distortion or a plug-in like MaxxBass to add higher harmonics to the bass. Make sure you check both on small speakers.

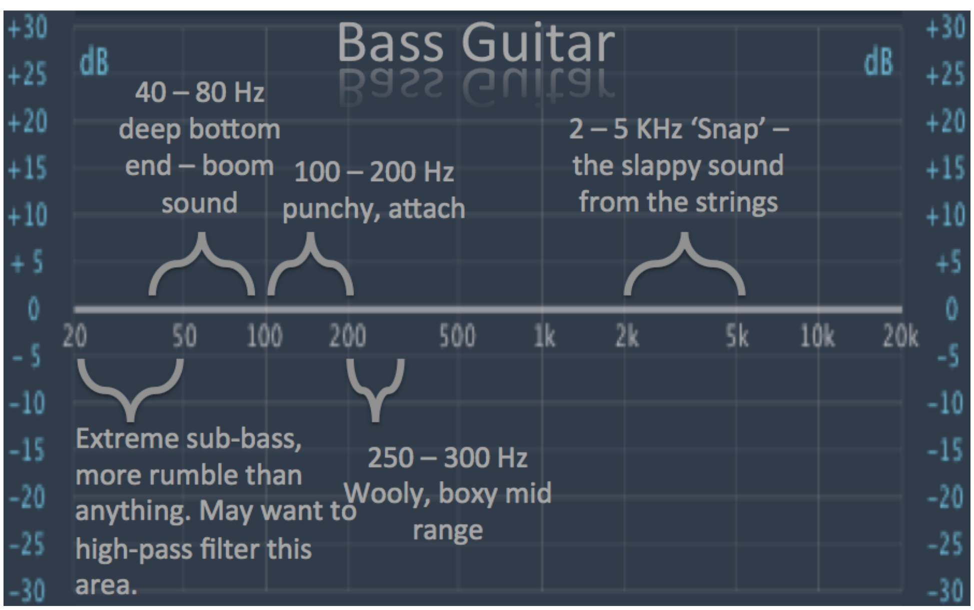


## For Bass

- The ratio between the low bass (80 to 100 Hz) and the mid bass (120 to 200 Hz) is important. Try using two fairly narrow-peaking bands, one at 100 Hz and another at 150 Hz, boosting one and cutting the other slightly. If the bass is too “warm”, sometimes reducing the lower band can make it more distant without removing the deeper fundamentals that live in the 100-Hz band.
- For clarity on the bass, try boosting some of the 800-Hz area, since this can provide definition without getting too snappy sounding.
- A four-band parametric will allow you to adjust several bands below 200 Hz. Try attenuating the low frequencies around 40 to 70 Hz, then slightly boosting the frequencies from 80 to 120 Hz where the fundamental lies, then boost the frequencies from 130 to 200 Hz where the overtones and cabinet/neck/body resonances live.
- Muddy bass can mean a lot of things, but at a minimum it usually involves a lack of presence of the higher harmonics. Most bass tracks have a sweet spot between 600 Hz and 1.2 kHz where the upper-order harmonics sing, and this is the place to boost for more presence in the mix.
- Take a low-cut filter and center it at 250 Hz so that all the lows of the bass are attenuated. Now take a bell shaped EQ and boost it 4 dB with a narrow band and sweep around the 80 to 180 Hz region to find where your bass frequencies fit best in the track. Once you find it, widen the bandwidth and boost more if necessary. If you want more density on the bottom, you may need to do this with another bell filter on the frequencies below the previous one. This should tighten up the low end, add space for a kick drum, and make your mix less boomy.
- High-pass filter the bass anywhere from the 40 to 80 Hz. It’s amazing how much that can help the bass tighten up sometimes.
- Any instrument with low frequency content (below 500 Hz) can effect the sound of the bass. This includes kick, keyboards, (male) vocals, double bass, cello, low-tuned guitars, and so on. Cut the low-end (anywhere below 80 to 120 Hz) from tracks where the low end isn’t needed. This will help those tracks to cut through while leaving more space in the mix for the bass and the kick.
- Removing the 250-Hz region information from instruments such as guitars, keyboards, and even vocals is often of more use than cutting it from the bass.
- To achieve more definition from a bass guitar, first make a duplicate of the bass track and then process the duplicate track in your DAW with Moogerfooger Low-Pass plug-in. Set the plugin's

Mix parameter to 10, the Resonance to 1, and the overall Amount to 1. Now turn the frequency control up until you get a well-defined sub tone. Group both bass tracks to a separate bus and create a new aux track. Be sure to assign the input to the same bus. Then, on the original bass track in your DAW, use an EQ plug-in to roll some of the low subs out. Next blend the original bass with the Moogerfooger track to create a fat, solid composite bass sound. The aux track will now become the bass master track. Finally, EQ and compress that aux track to fine-tune the bass sound to taste. The same tip can also be used on kick drums.

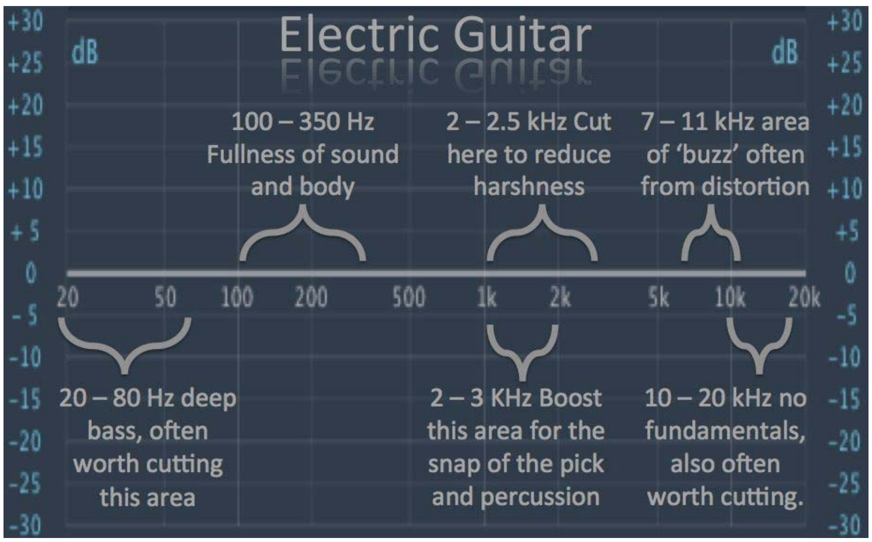
- With hip-hop and electronic music, the bass tends to contain a lot of information in the 30 to 60 Hz range that you can feel. Many hip-hop or EDM records will raise the low frequency target area slightly higher to the 70 to 100 Hz range and elongate the duration to create the illusion that there's a lot of bass information so that it can sound full on smaller monitors. Be careful not to over-EQ, though. Clubs and cars with huge bass drivers are already hyped in this frequency range.
- With rock bass, the idea is to create an aggressive in-your-face bass sound. For this the focus will be mainly on the amp sound. Boost anywhere between 50 and 100 Hz for the bottom end, dip between 400 to 800 Hz (this will allow the guitars and vocal to have more room to speak musically) and boost between 1.5 and 2.5 kHz for midrange.

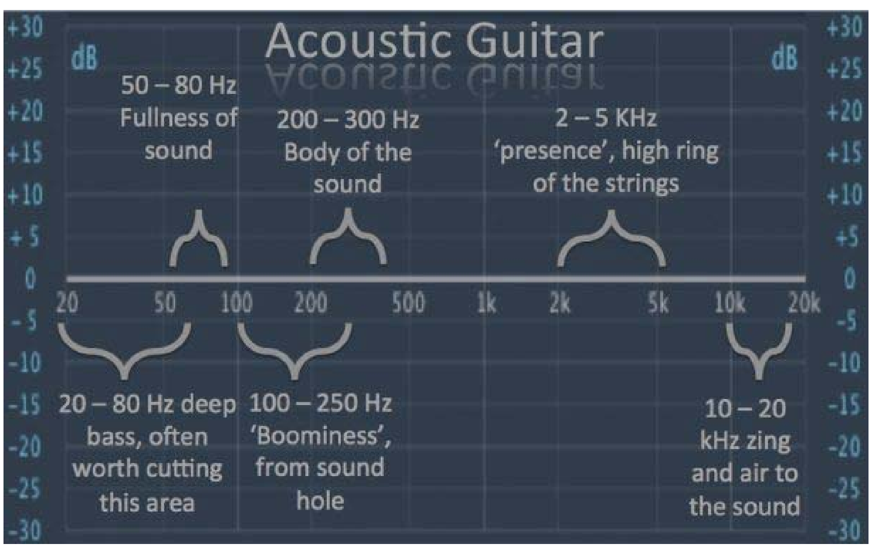


## For Guitars

For a fatter-sounding guitar, boost the midrange about 9 dB or so and sweep the frequencies until you hear the range where the guitar sounds thick but still bright enough to cut through the mix. Back the boost down to about the point where the guitar cuts through the mix without being too full-sounding.

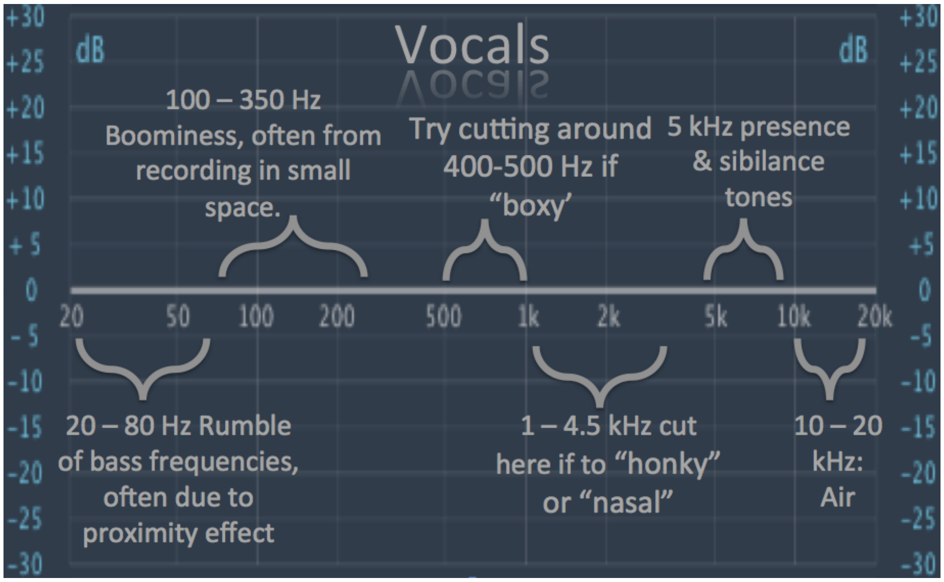
Boosting 10 kHz on guitars will accentuate finger noise and tiny string movements. Boosting 5 to 8 kHz will allow the guitar to better cut through the mix. Boosting 1 to 5 kHz will give the guitar more presence. Consider filtering the guitar with both high-pass and low-pass filters. If you leave too much low-end in a distorted guitar, it will compete with the rhythm section. Too much above 8 kHz can compete with the cymbals.



## For Vocals

Boost a little 125 to 250 Hz to accentuate the voice fundamental and make it more "chesty" sounding.
2 to 4 kHz accentuates the consonants and makes the vocal seem closer to the listener.




# The Dimension Element:
## Adding Effects

The fourth element of a mix is dimension, which is the ambient field that the track or tracks sit in. Dimension can be captured while recording, but it's usually created or enhanced when mixing by adding effects such as reverb, delay, or modulation. Dimension might mean re-creating an acoustic environment, but it could also mean adding width or depth to a track to try to spruce up a boring sound.

**There are four reasons why a mixer might add dimension to a track**

- To create an aural space
- To add excitement
- To make a track sound bigger, wider or deeper
- To move a track back in the mix (giving the impression that it's farther away)

## The Six Principals for Adding Effects

There are six principals that offer a general guideline on how effects can be used.

**Principal 1:** Picture the performer in an acoustic space and then recreate that space around him/her. This method usually saves time, as it's a lot faster than simply experimenting with different effects presets until something excites you (although that method can work too). Also, the artificially created acoustic space needn't be a natural one. In fact, as long as it fits the music, the more creative, the better.

**Principal 2:** Smaller reverbs or short delays make things sound bigger
Reverbs with decays under a second (and usually much shorter than that) and delays under 100 milliseconds (again, usually a lot shorter than that) tend to make the track sound bigger rather than push it back in the mix, especially if the reverb or delay is in stereo.

**Principal 3:** Long delays, long reverb pre-delay, and reverb decay settings push a sound further away if the level of the effect was enough.
Delays and pre-delays longer than 100 milliseconds are distinctly heard and begin to push the sound away from the listener. The trick between something sounding big or just distant is the level of the effect. When the decay or delay is short and the level is loud, the track sounds big. When the decay or delay is long and loud, the track just sounds far away.

**Principal 4:** If delays are timed to the tempo of the track, they add depth without being noticeable.
Most mixers set the delay pulse time to the tempo of the track. This makes the delay pulse with the music and adds a reverb type environment to the sound. It also makes the delay seem to disappear as a discrete repeat, but it still adds a smoothing quality to the mix element.
If you want to easily find the right delay time to the track and you have an iPhone, grab my Delay Genie App from the app store. It's free and will make timing your effects to the track easy.

**Principal 5:** If delays are not timed to the tempo of the track, they stick out.
Sometimes you want to distinctly hear a delay and the best way to do that is to make sure that the delay is not exactly timed to the track. Start by first putting the delay in time, then slowly alter the timing until the desired effect is achieved.

**Principal 6:** Reverbs sound smoother when timed to the tempo of the track.
Reverbs are timed to the track by triggering them off of a snare hit and adjusting the decay parameter so that the decay just dies before another snare hit. The idea is to make the decay "breath" with the track. One way to achieve this is by making everything as big as possible at the shortest setting first, then gradually making the settings longer until it's in time with the track.

Of course, the biggest part of adding effects to a mix is experience, but keeping these principles in mind will provide a perfect place to start.

# Using Delays

Delays are a secret weapon of many mixers who sometimes opt to use several of them instead of reverb to add depth to the mix. If set up correctly, a delay can both pulse with and blend into a track, making it seem both deeper and fatter without calling attention to itself.

## Types of Delays

**Haas Effect** This is a delay of 40 milliseconds or less that's not perceived as a separate repeat but can add a sense of spaciousness if panned opposite the source.

**Short:** A short delay is generally anywhere from 40 milliseconds to around 150. The idea is to add a double tracked effect. This can be heard on old Elvis records as well as others from that era.

**Medium** A medium delay is anywhere from 150 milliseconds to around 400. Even though we hear it as a distinct repeat, this length of delay is more for adding a sense of space around the source, even though it may be somewhat imperceptible sometimes if timed to the track.

**Long** A long delay is anywhere from 400 milliseconds up to about a second (1,000 milliseconds). You hear a long delay as a very distinct and specific repeat.

**Stereo** A stereo delay allows for a different delay time on each side of the stereo sound-field.

**Ping-Pong** A delay that bounces from one side of the stereo to the other.

**Tape** In the analog days, delay was accomplished by using an outboard tape machine. The delay occurred because the playback head was located after the record head, which created a time delay. As the speed of the tape machine changed, so would the delay. Today there are plugins that try to emulate the nuances of tape delay.

## Calculating Delay Time

Delays are measured tempo-wise using musical notes in relation to the tempo of the track. In other words, if the song has a tempo of 120 bpm, then the length of time it takes a quarter note to play would be 1/2 second (60 seconds / 120 = 0.5 second). Therefore a quarter-note delay would be 0.5 second or 500 milliseconds (0.5 x 1000 ms per second), which is how almost al delay devices are calibrated.

But 500 ms might be too long and set the source track too far back in the mix. To get smaller delay increments, do the following:

Divide a quarter-note delay in half to get an 1/8 note delay (500 ms / 2 = 250 ms)
For a 1/16 note delay (250 ms / 2 = 125 ms)
For a 1/32 note delay (125 ms / 2 = 62.5 ms, or rounded up to 63 ms)
For a 1/64 note delay (62.5 ms / 2 = 31.25 ms)
For a 1/128 note delay (31.25 ms / 2 = 15.625 ms)
For a 1/256 note delay (15.625 ms / 2 = 7.8125 ms)

Now such small increments like 8 ms and 16 ms might not seem like much, but they're used all the time to make a sound bigger and wider. Even a short delay like this will fit much more smoothly into the track it it's timed.

It's also possible (and sometimes even preferable) to use other note denominations, such as triplets or dotted eighths, sixteenths and so on. These odd denominations can be calculated as follows:

**Delay Time x 1.5 = Dotted Value**

Example: 500 ms (quarter-note 120 bpm delay) x 1.5 = 750 ms (dotted quarter note)

**Delay Time x 0.667 = Triplet Value**

Example: 500 ms (quarter-note 120 bpm delay) x 0.667 = 333.5 ms (quarter note triplet)

These can be further subdivided just like we did with the regular notes.

### Setting the Repeats

Repeats, sometimes called regeneration, sends some of the output of the delay back into the input. The amount is usually set in percentages where 0% is a single repeat and 100% is an infinite loop of repeats that eventually becomes feedback that gets louder and louder.

Although the number of repeats is dependent upon the song and its tempo, most of the time delays are set up for one to three repeats (less than 20% feedback). Too many repeats can sometimes result in the track sounding muddy.

The best way to set the repeats is to only have enough to either fill a hole in the arrangement or fill the space until the next event. In the case of a vocal, the repeats would occur from the end of a phrase only until the next phrase began. Many delay plugins have a feature called dynamic delay, where the delay only begins after the source has died away and then automatically ceases the repeats when the next phrase begins again. This automatically keeps the delay out of the way, but the benefits of having a delay on all the time are negated.

### Typical Delay Setups

Many mixers use standard delay setups of short and long; short, medium and long; or even full-note, triplet, and dotted note. Many times the delays will be set up during the session prep, but the timing will be determined during the mix.

### Sample of a two-delay setup:

A song that is 105 bpm.

**Short Delay:** 1/32 note = 71 ms

**Long Delay:** Dotted 16th note = 214 ms

### Sample of a three-delay setup:

A song that is 105 bpm.

**Short Delay:** 1/32 note triplet = 48 ms

**Medium Delay:** 1/16 note = 143 ms

**Long Delay:** 1/4 note triplet = 381 ms

## Delay Techniques for Vocals

- A stereo delay with a 1/4 or 1/8 note delay on one side and a 1/4 or 1/8 note triplet or dotted note on the other provides movement along with depth and is a favorite trick of EDM mixers.
- To simulate a vocal double, dial in a 1/16-note delay and then modulate it so it slowly raises and lowers in pitch. If the modulation can be set so it's random, it will sound more realistic.
- For a quick vocal effect to give it some space and depth during tracking and overdubs, set up a mono 220 ms delay with a couple of repeats.
- Paul McCartney reportedly used a 175 ms delay on his vocals almost all of the time.
- For getting a dry vocal to jump out, use two bandwidth-limited (at about 400 Hz to 2.5 kHz) delays in the neighborhood of 12 ms to the left and 14 ms to the right, each panned slightly off-center. Bring up the delays until you can hear them in the mix and then back it off to where you can't. Occasionally mute the returns to make sure it's still bringing the vocals out as they sit well into the rest of the balance. You can also time the delays to a 1/64 note on one side and a 1/128 note on the other.

## Delay Techniques for Guitar

- During the 80s when guitars were often recorded direct, many LA session guitarists used a short stereo delay of 25 ms on one side and 50 ms on the other to provide some space around the sound.
- To make the guitar sound large, set a delay at less than 100 ms (timed) and pan the guitar to one side and delay to the other.
- Use a mono delay on the guitar set to about 12 ms (or whatever the tempo dictates) and hard-pan both the guitar and the delay. This makes the guitar sound much bigger and almost like two people playing perfectly in sync, yet still keeps a nice hole open in the middle for the vocals.
- Pan the guitar track and the delay to the center (or put your monitors in mono); then slowly increase the delay time until it sounds bigger. Increase it a little more for good measure. You'll probably find the result is in the area of 25 to 30 ms.

**Tip:** Instead of always syncing your delay to the tempo of the song in your DAW, try tapping the tempo manually instead or set the delay slightly ahead of the beat or behind the beat for a more organic sounding groove.

## Delay Technique for Keyboards

A stereo delay setting of 211 ms on one side and 222 on the other provides a quick and easy room simulation and adds some life to a directly recorded keyboard.

# Using Reverb

There are five primary categories of reverb, all with a different sonic character; three of these are actual acoustic spaces, one is an analog way to reproduce one, and one is not found in nature but can easily sound cool. The reason why there's a difference is that just like everything else in music and audio, there are many paths to the same end result. You'll find that every digital reverb plugin or hardware-unit provides its own version of these sounds.

## Hall

A large space that has a long decay time and lots of reflections. Sometimes there's a subcategory of the hall reverb called "church" which is just a more reflective hall with a longer decay.

## Room

A much smaller space that can be dead or reflective, depending upon the material that the walls floor and ceiling are made of. It usually has a short decay time of about 1.5 seconds or less.

## Chamber

An acoustic chamber is a dedicated tiled room that many large studios used to build to create reverb.

## Plate

A plate is a 4 foot by 6 foot piece of sheet metal with transducers attached to it that many studios used for artificial reverb when they couldn't afford to build a chamber. The first plate reverb was the EMT 140 developed in the late 1950s.

## Non-Linear

The non-linear category is strictly a product of modern digital reverbs, as the sound isn't found anywhere in nature. While natural reverbs decay in a rather smooth manner, once a reverb is created digitally, it's possible to make the delay happen in unusual ways. The reverb tail can be reversed so that it builds instead of decays, or it can be made to decay abruptly, both of which make the decay non linear. This preset was a popular mixing effect used on drums during the 80s, when the feature first became available on the AMS RMX 16 Digital Reverb. As to which reverb category to use, it's up to the mixer. Many mixers might always use a room or chamber on drums, a plate on vocals and guitar, and a half on strings or keyboards, while others may do the opposite.

## Timing Reverbs to the Track

Just like delays, reverbs sound smoother if they're timed to the pulse of the track. Doing this adds depth without sticking out and makes the mix seem more polished. The two parameters that are adjusted for timing are the decay time and the predelay.

## Timing the Decay

In simple terms, the decay time is the time it takes for the reverb tail to die out. If the decay time is timed to the pulse or bpm of the song, the track seems tighter and cleaner while still retaining all of the depth.

To time the decay time to the track, trigger the reverb with the snare and adjust the decay parameter so that the decay just dies by either the next snare hit or a later one. The idea is to make the decay "breath" with the track. You can use this decay time for the other reverbs, but you'll probably have to adjust them slightly because the decay response of every reverb or reverb setting is different.

## Timing the Predelay

Predelay means delaying the reverb entrance slightly after you hear the source signal. The reason it's used is so the source signal doesn't sound washed out in ambiance. With a little bit of predelay, you'll hear the source's attack, then the reverb, so the source signal has more definition as a result.
Predelay is usually timed to the tempo of the track. Back in the days of real plates and chambers, predelay was achieved by using the slap delay from a tape machine, but today it's a standard parameter on every reverb plugin or hardware device.
The same way that you determined the delay for the track provides the timing for the predelay. The difference is that you usually need a smaller increment than you might've used for a delay, and it's less than 100 ms.
Of these two parameters, the predelay is probably the most important in that the reverb seems more a part of the track when that parameter is timed.

## Typical Reverb Setup

In many cases two reverbs are used, with one set to a short decay and used on drums and a second with a longer decay and used on the other mix elements.
A song with a 105 bpm:

1. Short Reverb: Room, 18 ms predelay, 1.2 second decay
2. Long Reverb: Plate, 36 ms predelay, 1.8 second decay

### Three Reverb Setup

In a three reverb setup, we may use different reverb categories, but the decay is relegated to short, medium and long.

A song with a 105 bpm:

1. Short Reverb: Plate, 0.8 second decay
2. Medium Reverb: Chamber set to a decay time of 1.4 seconds
3. Long Reverb: Hall, 2.2 second decay
4. Start with all three predelay s set to 18 ms and adjust them from there as the mix progresses

## Reverb Techniques

### Vocals

Automate the delay or reverb return so that in the sparse parts of the arrangement, particularly in the beginning of the song, the vocal is less wet and more up front and intimate, which also makes the effect less obvious.

Try mixing various reverbs. Set up three reverbs: short, medium and long (the specifics of what the actual lengths are varies with the song). On a non-ballad vocal, favor the short and medium over the long. The short (try a 0.3 to 0.6 second room or plate) one will thicken the sound. Blending in the medium (1.2 to 1.6 second plate or hall) will create a smooth transition that is quite dense but still decays fairly fast. Add a little of the longer one (2 to 3 second hall) for whatever degree of additional decay you want. The three combined will sound like one thick reverb that will stick to the vocal and not muddy it up with excess length and diffusion.

With a singer/acoustic guitar player, try to picture the performer in an acoustic space and then realistically re-create that space around him. This lends itself to a medium-sized room or a small plate, with perhaps a little more reverb on the voice than the guitar. If the vocal is wet and the guitar dry (forgetting about leakage for a moment), it's difficult to have them both appear to share a common acoustic space.

If a vocal effect is too prominent, bring up the reverb to where you can hear it, then back off the level 2 dB. Add a dB or two at 800 Hz to 1 kHz to either the send or return of the reverb to bring out the effect without it being too prominent.

For an interesting reverse reverb effect on a vocal where it's whooshing in before the vocal begins, set a reverb to a very long decay time (more than 4 seconds) and then record the reverb only onto a second track. Reverse it and move it forward on the timeline so it begins before the vocal.

## Drums

For the Tommy Lee “Thunder Drums” effect, set a reverb on the cathedral” or “large hall” setting and then add a little to each drum. Pan the reverb returns so the reverb sits behind each part of the kit. For this effect to work, the bass drum has to sound tight to begin with and have a decent amount of beater present, and all the drums should be gated with the gate timed to the track.

For an “exploding snare” type of effect, add a short slap from 50 to 125 ms with a touch of feedback to the bottom snare mic. Bring the slap back on a second channel. Using an aux, send signal from both top and bottom snare mics and the slap to a short reverb of a second or less (timed to the song). By adjusting the proportions, phase, and EQ, the effect will fit it into almost any situation.

## Percussion

For hand percussion, such as shakers and tambourines, use a medium (0.8 to 1.2 seconds) room or plate reverb with either zero or very short (20 ms) predelay.

## Guitars

To make guitars bigger, take a mono reverb and lower the decay time to as low as it will go (0.1 seconds if it will go that low). Pan the guitar to one side and the reverb to the other. Try different reverb types to see which works better in the song. Increase the decay time slightly to make the sound bigger or to eliminate any metallic-sounding artifacts from the reverb.

For that early Eddie Van Halen sound, use either a chamber or a plate reverb set to about 2 seconds decay time and around 120 ms predelay that's timed to the track. Pan the guitar to one side and the reverb more to the other.

## Keyboards

For a keyboard pad sound that melts into the track, use a hall reverb with a 2 to 2.5 second decay and a short (20-ms) predelay that's timed to the track. Set any EQ or filters so that the extreme high and low ends are rolled off to about 8 kHz and 150 Hz.

## Strings

Use a hall reverb set to between 2.2 and 2.6 seconds with a predelay of at least 20 ms timed to the track

# Using Modulation

Modulation is the third type of effect that adds dimension to a mix, although it accomplishes this more by movement than by ambience. Most musicians and engineers are very familiar with the types of modulation, but they're not clear on how they differ and when they're best used.

## Types of Modulation

There are three basic types of modulation effects: phase shift, chorus, and flange. The difference between them is that basically a chorus and flange effect comes as a result of a modulated delay that's mixed back into the original signal, with the flanger having shorter delay than a chorus. On the other hand, the phaser doesn't require a delay to achieve its effect.

## The Differences between Modulation Effects

| Effect | Delay | Description |
|---|---|---|
| **Phase Shift** | None | Cancels out frequencies by shifting their phase to create the effect. Frequency notches are spaced evenly across the frequency range. |
| **Flanging** | 0.1 ms to 5 ms | The deeper the frequency cancellations, the deeper the effect. Frequency notches are randomly and harmonically spaced across the frequency response. |
| **Chorus** | 5 ms to 25 ms | Used to thicken the sound and create a stereo image. Frequency notches are spaced harmonically across the frequency response. |
| **Tremolo** | None | Cyclically changes the volume. |
| **Vibrato** | None | Cyclically changes the pitch. |

All three produce a series of frequency notches that slowly sweep across the frequency bands of an instrument or vocal, leaving only a series of peaks, which is what you hear. Here's where the differences are more pronounced, as a phaser has only a small number of notches that are spaced evenly across the frequency range, while flangers and choruses have many more notches that are harmonically spaced, which provides a much more aggressive sound as a result.

Tremolo and vibrato are also popular modulation effects, although they operate differently because a delay isn't required for them to work. It's easy to get these confused, or even use their names interchangeably, but they are distinctly different; a tremolo varies the signal up and down in level, while vibrato varies the pitch up and down.

## Flangers and Phasers

The flanger is a dramatic effect that was first created in 1966 by Ken Townsend, the chief technician at EMI Studios in London (now known as Abbey Road Studios), in an attempt to come up with something called Automatic Double Tracking, or ADT. The effect was created because The Beatles' John Lennon loved the sound of his voice when it was doubled but hated the fact that he had to sing it a second time, so the EMI tech staff was asked to come up with a solution. The effect was accomplished by using two tape recorders at the same time, but it led to an almost accidental discovery. By slowing one of the tape machines down by placing a finger on the tape flange (the metal part of the tape reel), a sweeping harmonics effect resulted. Lennon then coined the effect “flanging."

The first known use of flanging came on the Beatles' song "Tomorrow Never Knows, but one of the first big Top 40 hits that used the effect came a year later on a song called “Itchycoo Park” by the British group the Faces, which featured a large dose of the effect at the end of the song. Soon every artist and producer wanted the effect on their song, but there was a problem in that to get the effect, two tape recorders needed to be set up, which was both expensive and very time consuming.

Even though technology was marching along, back in the '70s the only feasible electronic simulation was an analog effect called a phaser, but the sound had none of the intensity of tape-driven real flange. That's why phasing isn't used much even today; it's just not that dramatic of an effect.

It wasn't until digital delays came on the market in the '80s that it became possible to simulate true tape flanging, and now just about every modulation plug-in and stomp box can do at least a passable simulation of the effect if set up correctly.

## Chorus

What makes a chorus different from a flanger is that the delay is longer, going from about 5 to 25 ms, and the frequency notches needed to create the effect occur in a fixed cycle. The effect shines in stereo and can really widen the sound of a track quite a bit. Since chorusing first was introduced by Roland in 1980, many hits from that era used the effect over and over.

Today you'll find that most modulation plug-ins and hardware allow you to easily select between chorus, phasing, and flanging, since they're all related, but the ones that are used in stereo are the most dramatic.

## Tremolo and Vibrato

For years, guitar amps included tremolo as a standard feature, although in some cases (such as on Fender amps) it was mislabeled as vibrato. As stated previously, they're not the same, since tremolo is a cyclic variation in volume, while vibrato changes the pitch of the sound.

Guitars weren't the only instrument to use tremolo, as both the original Rhodes and Wurlitzer electronic pianos had the effect built in. Vibrato is rarely used because the variation in pitch can make the track or other tracks seem out of tune.

## Typical Modulation Setups

While flanging, tremolo, and vibrato are pretty dramatic effects and can cause a mix element to stand out, chorusing is often used to widen a track. That said, modulation effects aren't usually set up in advance on a dedicated set of sends and returns, since the effect tends to work better on an individual instrument or vocal.

That said, sometimes a flanger is required across the entire mix, so it's best inserted either across the stereo buss or across a separate subgroup so the level can be easily automated and adjusted.

## Flanging With A Subgroup

Tracks sent to a sub channel with a flanger insert along with unprocessed signal sent to the main buss.

```
[Individual Channels -  T1  T2  T3  T4 ]
  |  |
  |  Sent To Subgroup
  |  |
  |  With Flanger Insert
  |  |
To Master Buss
```

## Modulation Techniques

Here are a few techniques often used when adding modulation to a mix element. Don't be afraid to experiment, though, because any of these techniques may work well in other mix situations as well.

**For fatter lead or background vocals:**

Use some chorusing panned hard left and right to fatten up the sound. Ride the chorusing effect, adding and subtracting it according to what sounds best.

**For out-of-tune vocals:**
If you have something against Auto-Tune or just want to cover up an out-of-tune vocal, use a stereo chorus or flanger and pan it slightly left and right. The more out of tune the vocal, the more modulation you need to cover it up. This does an effective job of taking the listener's attention off any sour notes.

Pan a delayed signal behind the vocal and then send it to a chorus, detuning both sides a bit so the delay sounds wide and the modulation steals your attention from the tuning.

**Robot voice:**
Use a deep flange to make the voice sound metallic and then use it as an external key to gate SMPTE code in time with the voice. Mix in the gated code gently.

# EQing Effects

One of the things that many mixers struggle with is getting reverb and delay effects to blend well in the mix. This happens more with reverbs than delays, especially during those times when the reverb just never seems to sound quite right. Usually the way the problem is addressed is to audition different reverb presets until something is found that seems to work better, but that can take time, and it's easy to end up chasing your tail to the point where you're never sure which preset actually sounds the best. What many seem to forget is that most of those presets are the same basic reverb with different EQ settings, which you can add yourself to get there faster.

One thing that happened regularly back in the early days of analog reverb (especially with plates) is that one way to tune a reverb to the track was to insert an EQ on the send before the actual reverb itself. Usually the EQ was set more to cut than to boost (although you'd boost it if you wanted a bright-sounding plate that jumped out of the mix), but if done well, the reverb would suddenly fit a lot better in the track. In fact, back in the classic days of the big studios, this was done in the back room and not left up to the engineer at the console, and it became one of the reasons for clients wanting to work there; they loved the sound of their reverbs.

We can use those same techniques today using reverb plug-ins on our DAW, or any other effects for that matter. Just remember that it usually sounds best if the EQ is placed before the reverb, not after it, because it has a great effect on
the frequency response of the reverb.

**TIP:** Try adding tape-saturation plug-ins such as Avid's Heat or Universal Audio's Studer A800 Tape Recorder to the effect send or return. The extra harmonics sometimes give it more depth

## The following sections describe three EQ curves that are frequently used:

### On Vocals

One thing about reverb is that any low end from it just muddies up the track, and any high end may stick out too much, which is why it might be a good idea to roll each end of the frequency spectrum off a bit. In many cases this means somewhere around 200 Hz and 10 kHz (or even lower). When reverb is used on vocals, sometimes it fits
better if there's also a bit of an EQ scoop in the midrange around 2 kHz, where the consonants of the vocal live, so the effect stays out of the way frequency-wise. Once again, this is very effective on delays and modulators as well.

**Vocal Reverb EQ Curve**

High Pass Filter at 200 Hz, Attenuate 2 kHz, Low Pass Filter at 10 kHz

### On Instruments

For instruments, the Abbey Road curve, which is what the famous studio has used on their reverbs since the '60s, works very well. This means that the low end is rolled off at 600 Hz and the high end at 10 kHz. This curve makes any reverb sound a lot smoother and fit better with the track. You'll find that this setting just increases the depth without it sounding washed out when you add more reverb using this curve. Of course, too much of a good thing is no good either, so be judicious with the amount you add.

**The Abbey Road Reverb EQ Curve**

High Pass Filter at 600 Hz - Low Pass Filter at 10 kHz

### On Drums

Sometimes reverb on the drums is the toughest of all in that you want depth without calling attention to the ambience. A good way to do that is a variation of the Abbey Road curve where the high end is severely rolled off to ok, 4k, or even 2 kHz! You'll find that you'll have some depth without the ambience ever calling attention to itself.

**Drum Reverb EQ Curve**

High Pass Filter at 600 Hz - Low Pass Filter at 6 kHz

While these EQ curves work great with reverbs, don't be afraid to try them with delays or modulation effects, as the results are very similar. You'll get depth without the delay or effect getting in the way. Of course, if you want to really hear the reverb or delay, go the opposite way and increase the high end, and the effect will jump right out of the track.

## Equalization Tips for Reverbs and Delays

- To make an effect stick out, brighten it up.
- To make an effect blend in, darken it up (filter out the highs).
- If the part is busy (such as with drums), roll off the low end to keep it out of the way.
- If the part is open, add low end to the effect to fill in the space.
- If the source part is mono and panned hard to one side, make one side of the stereo effect brighter and the other darker and pan the brighter side opposite the source track.

## Layering Effects

Layering means that each instrument or element sits in its own ambient environment, and each environment is usually artificially created by effects. The idea here is that these sonic atmospheres don't clash with one another, just like in the case of frequency ranges.

The following sidebar features some suggestions so that the sonic environments don't clash.

**Layering Tips for Reverbs and Delays**

Layer reverbs by frequency, with the longest being the brightest and the shortest being the darkest.
Pan the reverbs any way other than hard left or right.
Return the reverb in mono and pan accordingly. All reverbs needn't be returned in stereo.
Get the bigness from reverbs and depth from delays, or vice versa.
Use a bit of the longest reverb on all major elements of the track to tie all the environments together.

"My personal taste is to use more layers, like using several reverbs to create one reverb sound, or using several short and long delays. My reverbs and effects usually end up coming from four to eight different sources. They'll be short, long, bright,
dull, and everything you need to make an environment." -Bob Bullock

**Reamping**

One of the ways to re-create a natural environment is by a process known as reamping. This is accomplished by actually sending a signal of an already-recorded track (say a guitar or keyboard) back out to an amplifier in the studio, then miking it from a distance to capture the ambience of the room. It's all the better if the ambience is captured in stereo.

**The Dynamics Element: Compression, Limiting, Gating, and De-Essing**

At one point in recording history, the control of the volume envelope of a sound (or its dynamics) would not have been included as a necessary element of a great mix. fact, dynamics control is still not a major component in some of the classical and jazz mixing world. Today's modern mixes have different demands, though, so the manipulation of dynamics plays a major role in the sound in most music, because the fact is that almost nothing else can affect your mix as much and in so many ways as compression.

“I think that the sound of modern records today is compression. Audio purists talk about how crunchy compression and EQ are, but if you listen to one of those jazz or blues records that are done by the audiophile labels, there's no way they could ever compete on modern radio even though they sound amazing. Every time I try to be a purist and go, 'You know, I'm not gonna compress that, the band comes in and goes, 'Why isn't that compressed?'"" - Jerry Finn

### Types of Dynamics Control

An audio source's dynamic range is controlled by the use of compression, limiting, and gating. For those of you new to mixing or for those who need a review or clarification, here's a brief description of each. See the glossary or any number of recording texts for more complete information.

# Compression

What we know as compression is more properly called dynamic range compression because it's the process of taking an audio source signal with a large dynamic range and making it smaller. This is done by lowering the loudest portions of the program and increasing the lowest ones so the volume level is more constant.

Compressors work on the principle of gain ratio, which is measured on the basis of input level to output level, and is set by using the Ratio control. This means that for every 4 dB that goes into the compressor, 1 dB will come out, for a ratio of 4 to 1 or 4:1. If the ratio is set at 8:1, then for every 8 dB that goes into the unit, only 1 dB will be seen at the output. A ratio of 1:1 results in no compression at all.

### Typical Compressor Parameter Controls:

```
+4 dB input        4:1 Compression ratio      +1 dB output
--------------------------------------------------------
+8 dB input        8:1 Compression ratio      +1 dB output
```

The Threshold control determines the point in the signal level where the unit will begin to compress. As a result, threshold and ratio are interrelated, and one will affect the way the other works. Some compressors (such as the Universal Audio LA-3A) have a fixed ratio, but on most units the parameter is variable.

Most compressors have Attack and Release parameter controls that determine how fast or slow the compressor reacts to the beginning (the attack) and end (the release) of the signal envelope. These controls are especially important because the setting is crucial to how the compression works and sounds. Because the settings on the Attack and Release controls can be tricky if you're not sure how to use them, some compressors have an Auto mode, which will set the attack and release according to the dynamics of the input signal. Although Auto mode can work relatively well, it doesn't allow for the precise settings that may be required to properly control certain source material. While most compressors have some control over the volume envelope, some compressors have a fixed attack and release (such as the dbx 160 series), which gives it a particular sound as a result.

Many compressors also have a Knee parameter, which sets how fast the compressor will begin to compress after the signal reaches the threshold level. A low value (sometimes it's measured in dB, so 0 dB would be the lowest) means that the compression will begin instantly at threshold. A higher setting will gradually ease in the compression, which may sound better on certain program material.

When a compressor operates, it actually decreases the gain of the signal, so there's another control that allows the signal to be boosted back up to its original level or beyond called Make-Up Gain or Output.

All compressors have a gain-reduction meter to show how much compression is occurring at any given moment. This can look like a normal VU or peak meter, but it reads backward. In other words, 0 dB means that the signal is below threshold and no compression is taking place, and any travel to the left or down into the minus range
shows the amount of compression that's occurring. For example, a meter that reads -6 dB indicates that there is 6 dB of compression taking place at that time.

Many compressors also have a feature known as a Sidechain (sometimes called a “key” input), which is a separate input into the compressor so that other signal processors can be connected to it. This sidechain can have a number of useful purposes, such as when an EQ is connected to it to make a makeshift de-esser. If only the frequencies in the upper range are boosted, the loud “S” sounds from a vocalist will be attenuated when they exceed the compressor's threshold (more on that in the section on “De-Essing”). You can also connect a delay, reverb, or any other signal processor to the sidechain to create some unusual, program level-dependent effects.
Because a sidechain isn't needed for most everyday compressor operations, many manufacturers elect not to include sidechain connections on hardware units, but may include one on the plug-in version.

A sidechain can also be used to "duck" or attenuate another instrument when a new one enters. For instance, if you connected a send of a vocal track into the sidechain of a compressor inserted on a loop track, the loop would lower in volume whenever the vocal entered and then return to its normal level when the vocal stopped. This is what happens at an airport when the music is automatically lowered for a gate announcement and then returns to its normal level when the announcement is finished.

```
Vocal | Connected  | When vocal    | Inserted | Loop
      | to         | enters loop   | in       |
      | compressor | is attenuated | loop     |
Track | sidechain  |               | channel  | Track
      | input      | [Compressor]  | signal   |
      |            | [with      ]  | chain    |
      |            | [sidechain ]  |          |
      |            | [input     ]  |          |
```

## Compressor Differences

In the days of analog hardware compressors, there were four different electronic building blocks that could be used to build a compressor. These were:

**Optical** A light bulb and a photocell were used as the main components of the compression circuit. The time lag between the bulb and the photocell gave it a distinctive attack and release time (like in an LA-2A).

**FET** A Field Effect Transistor was used to vary the gain, which had a much quicker response than the optical circuit. (A Universal Audio 1176 is a good example.)

**VCA** A voltage-controlled amplifier circuit was a product of the '80s and had both excellent response time and much more control over the various compression parameters. (The dbx 160 series is an example of a VCA-type compressor, although some models didn't have a lot of parameter controls.)

**Vari-Gain** The Vari-Gain compressors are sort of a catch-all category because there are other ways to achieve compression besides the first three (such as the Fairchild 670 and Manley Vari Mu).

As you would expect, each of the above has a different sound and different compression characteristics, which is the reason why the settings that worked well on one compressor type won't necessarily translate to another. The good thing about living in a digital world is that all of these different compressor types have been duplicated by software plug-ins, so it's a lot easier (not to mention cheaper) to make an instant comparison on a track and decide which works better in a particular situation.

### Multi-Band Compression

Multi-band compression splits the input audio signal into two or three frequency bands, each with its own compressor. The main advantage of a multi-band is that a loud event in one frequency band won't affect the gain reduction in the other bands. That means something like a loud kick drum will cause the low frequencies to be
compressed, but the mid and high frequencies are not affected. This allows you to get a more controlled, hotter signal with far less compression than with a typical single-band compressor.

## Limiting

Compression and limiting are closely related, the main difference being the setting of the ratio parameter and the application. Any time the compression ratio is set to 10:1 or greater, the result is considered limiting, although most true limiters have a very fast attack time as well. A limiter is essentially a brick wall level-wise, allowing the signal to get only to a certain point and little more beyond. Think of it doing the same thing as a governor that's sometimes used on 18-wheel trucks owned by a trucking company to make sure that they're not driven beyond the speed limit. Once you hit 65 mph (or whatever the speed limit in your state is), no matter how much more you depress the gas pedal, the truck won't go any faster. The same theoretically occurs with a limiter. Once you hit the predetermined level, no matter how much you try to go beyond it, the level pretty much stays the same.

Most modern digital limiters (either hardware or software) have an internal function known as “look ahead” that allows the signal detection circuitry to look at the signal a millisecond or two before it hits the limiter. This means that the limiter acts extremely fast and just about eliminates any overshoot of the predetermined level, which can be a problem with analog limiters because they react much more slowly to transients.

Limiting is used a lot in sound reinforcement for speaker protection (there are some limiters on powered studio monitors as well), and not as much in mixing with the following exception. Many engineers who feel that the bass guitar is the anchor for the song want the bass to have as little dynamic range as possible. In this case, limiting the bass by 3 to 6 dB (depending on the song) with a ratio of 10:1, 20:1, or even higher will achieve that.

## De-Essing

One of the major problems when tracking vocals is a predominance of S's that comes from a combination of mic technique, the mic used (usually a condenser with a presence peak in the sibilance range), the EQ added to make it cut through the mix, and the use of heavy compression. Sometimes this might not be too much of an issue until it's time to mix, but when a compressor is put on the vocal to even out the level, all of a sudden every S from the singer seems ear-piercing. This effect is what's known as sibilance and is totally undesirable.

The way to combat sibilance is to use a de-esser, which is a unit that compresses just the S frequencies that usually fall somewhere between 3 kHz and 10 kHz, depending upon the situation.

A de-esser can be created from a compressor with an equalizer plugged into the sidechain as stated above, or it can be a dedicated unit designed just for this purpose. While many hardware de-essers are limited to two parameter controls, Threshold and Frequency (Figure 9.7), software de-essers are much more sophisticated, allowing for pinpoint control that a hardware unit just can't duplicate. The Threshold control is similar to the control found on a compressor/limiter in that it sets the level when the de-essing process begins. The Frequency control allows you to fine tune the frequency to exactly where the S's occur.

# Gating

Although not used nearly as much in the studio now that so much processing and editing is possible in digital workstations, gates are still used in certain situations in the studio and a lot in sound reinforcement. A gate keeps a signal turned off until it reaches a predetermined threshold level, at which time it opens and lets the sound through. The gate can be set to mute the sound completely when it drops below threshold or to just lower the level to a predetermined amount. Depending on the situation, just turning the level down a bit many times sounds more natural than turning it completely off, although the total silence can sometimes be used as a great effect.

For a long time engineers would use a gate (sometimes called a noise gate or expander) to reduce or eliminate problems on a track such as noises, buzzes, coughs, or other low-level noises off-mic. On loud electric-guitar tracks, for instance, a gate could be used to effectively eliminate amplifier noise during the times when the guitar player is not playing. Because most of these situations can now more effectively be dealt with in the DAW, gates aren't used as much in the studio, although there's still a need for them in a live-performance situation.

Gates are still frequently used on drums to control leakage resulting from tom mics in a mix, or to tighten up the sound of a floppy kick drum by decreasing the ring after it's struck by the beater.

Gates can also have a sidechain, or just an additional input called a key or trigger input, which allows the gate to open when triggered from another instrument, channel, or processor. This can be very useful in a number of situations, as seen in the "Gating Techniques" section at the end of this chapter.

Just like compressors, modern hardware gates are very fast, but plug-ins have the advantage that they can be designed so that the sidechain senses the signal a millisecond or two before it arrives at the gate's main input, allowing it to begin opening just before the transient arrives. This look-ahead facility is only an advantage when dealing with sounds that have a very fast attack, so it's sometimes switchable or variable when it is provided.

# Using Compression

If there is one major difference between the sound of a demo or semi-pro recording and a finished professional mix, it's the use of compression. As a matter of fact, the difference between the sound of one engineer's mix from another is more often than not how he uses his compressors. There are two reasons to add compression to a track or mix: to control the dynamics or as an effect.

### Controlling Dynamics

Controlling dynamics means keeping the level of the sound even. In other words, lifting the level of the soft passages and lowering the level of the loud ones so that there's not too much of a difference between them. Here are a couple of instances where this might be useful:

### Bass Guitar

On a bass guitar. Most basses inherently have certain notes that are louder than others and some that are softer than others depending upon where on the neck they're played. Compression can even out these differences.

### Vocal

On a lead vocal. Most singers can't sing every word or line at the same level, so some words may get buried as a result. Compression can help every word to be heard.

### Kick or Snare

On a kick or snare drum. Sometimes the drummer doesn't hit every beat with the same intensity. Compression can make all hits sound somewhat the same.

**TIP:** When controlling dynamics, usually a very small amount of compression (2 to 4 dB) is used to limit the peaks of the signal.

"I like to compress everything just to keep it smooth and controlled, not to get rid of the dynamics. Usually I use around a 4:1 ratio on pretty much everything I do. Sometimes on guitars I go to 8:1. The kick and the snare I try not to hit too hard because the snare really darkens up. It's more for control, to keep it consistent. On the bass, I hit that a little harder, just to push it up front a little more. Everything else is for control more than sticking it right up in your face." – Benny Faccone

## Compression as an Effect

Compression can also radically change the sound of a track. A track compressed with the right compressor and with the correct settings can make it seem closer to the listener and have more aggression and excitement. The volume envelope of a sound can be modified to have more or less attack, which can make it sound punchy, or to have a longer decay so it sounds fatter.

"Compression is the only way that you can truly modify a sound because whatever the most predominant frequency is, the more predominant that frequency will be the more you compress it. Suppose the predominant frequencies are 1k to 3k. Put a compressor on it, and the bottom end goes away, the top end disappears, and you're left with "Ehhhhh" [a nasal sound)." - Andy Johns

"I usually spend the last two hours of my mix not doing much mixing but listening and then making little adjustments to the master buss compressor and hearing what the impact is to all the parts. Most of the effect is how it's putting the low-frequency information in check, which it does without the meter even moving at all. Even when you're hitting it very lightly, it still has a dramatic effect on the music." - Bob Brockman

## Placement in the Signal Chain

Where a compressor, limiter, de-esser, or gate is placed in the signal chain has a dramatic effect on what that device does to the sound. If a compressor or limiter is placed after an EQ, every time the EQ is changed it will affect the compressor settings. Plus, any frequency that's boosted will have less of an effect because it will be turned down by the compressor. Likewise, a de-esser placed after an EQ may become ineffective if the midrange or high frequencies are changed. Similarly, the threshold of a gate can change if the EQ is changed if it's placed before it in the signal chain.

That's why it's best to place any compressor, limiter, de-esser, or gate first in the signal chain, because the signal will receive all the benefits of the dynamics module and any other processor that comes after it.

```
Compressor    -> Equalizer -> Track
Expander/Gate -> Equalizer -> Track
```

## Setting the Compressor

In most modern music, compressors are used to make the sound “punchy” and in your face. The trick to getting the punch out of a compressor is to let the attacks through and play with the release to elongate the sound. Fast attack times can sometimes reduce the transients, and therefore the high end of a signal, while slow release times may make the compressor pump out of time with the music or even cause the signal to distort.

Since the timing of the attack and release is so important, here are a few steps to help set up a compressor. Assuming you have some kind of constant tempo to the song, you can use the snare drum to set up the attack and release parameters, which may also transfer to other tracks as well. Regardless of the instrument, vocal, or audio source, the setup is basically the same.

1. Start with the attack time set as slow as possible (probably around 500 ms), the release time set as fast as possible (probably around 0.1 or 0.2 seconds), and the threshold set as high as possible so no compression is triggered.
2. Decrease the threshold until the meter shows some compression, then turn the attack faster until the sound of the instrument begins to dull, which now means that you're compressing the transient portion of the sound envelope. Stop increasing the attack time at this point and even back it off a little so the sound stays crisp and maintains its clarity.
3. Adjust the release time so that after the initial attack, the volume goes back to at least 90 percent of the normal level by the next beat. If in doubt, it's better to have a shorter release than a longer one, although you may hear the compressor pumping if it's set too fast.
4. If you want the compression to sound smooth and controlled, select a low ratio of around 2:1. If you want to hear the compressor or you want the sound to be punchier, select a compression ratio above 4:1.
5. Bypass the compressor to see whether there's a level difference. If there is, increase the Gain or Output control until the volume is the same as when it's bypassed.
6. Add the track to the rest of the mix and listen. Make any slight adjustments to the attack and release times as needed.

**TIP:** The idea of setting the compressor's attack and release is to make it breathe with the pulse of the song

"I get the bass and drums so they just start to pump to where you can actually hear them breathing in and out with the tempo of the song. What I'll do is put the drums and bass in a limiter and just crush the hell out of it. Then I'll play with the release and the attack times until I can actually make that limiter pump in time with the music, so when the drummer hits the snare, it sucks down and you get a good crest on it.

When he lets go of the snare, the ambience of the bass and the drums suck and shoot back up again. You can actually hear a [breathing sound] going on that was never there before. It really was there; it's just that you're augmenting it by using that limiter."
-Lee DeCarlo

"I set the attack as slow as possible and the release as fast as possible so all the transients are getting through and the initial punch is still there, but it releases instantly when the signal drops below threshold. I think that's a lot of the sound of
my mixes. It keeps things kinda popping the whole time. Also, you can compress things a little bit more and not have it be as audible." - Jerry Finn

### What's the Right Amount of Compression?

The amount of compression added is usually to taste, but generally speaking, the more compression the greater the effect will be and the more likely it will change the sound of the instrument or vocal. Compression of 6 dB or less is meant more for controlling the dynamics of a track rather than for changing its sonic quality, and often just a dB or two is all that's needed.

That said, it's not uncommon for radical amounts of compression to be used sometimes. Fifteen or 20 dB is routinely used for electric guitars, room mics, drums, and even vocals. As with most everything else, it depends on the song, the arrangement, the player, the room, and the gear.

"There are times when there's singing when they're not in compression at all, but if my limiter hits 15 or 20 dB of compression and I don't hear it, I don't think about it for an instant more." - Nathaniel Kunkel

**TIP:** The higher the compressor ratio control is set, the more likely you'll hear the compressor work. The more compression that's added, the more likely it is that you'll hear it.

## Parallel Compression

Parallel compression is a trick that mixers have been using ever since the '70s to make a track sound punchy in a very natural way. The trick involves sending the signal to another channel either through a mult on a hardware console or via a buss or aux on a DAW mixer, then adding a compressor to the second channel only.

The compressed channel is then brought up in level so it's just under the original non-compressed channel. This gives the track a feeling of control without sounding too squashed. Keep in mind that parallel compression works well with just about any individual instrument or vocal, and if you're working in a DAW, you're limited only by the number of tracks that you want to deal with.

**Parallel Compression**

```
Both channels mixed together
---------------------------Bus-->
    ^              ^
    |              |
    |              |
    |          Compressor
    |              |
    |              |
  Bass         Copy of
  Track        Bass Track
```

“On this mix right now there's a parallel compressor on the kick and snare, then there's another just on the snare. There's a stereo one on the toms and overheads, a mono one on just the dirty bass (this song has three basses), a stereo one on the guitar and vocals, and then a couple of different ones just for the lead vocal, one that's sort of spitty and grainy and one that's sort of fat. That changes from mix to mix.” - Andrew Scheps

## The New York Compression Trick

One of the little tricks that seem to set New York mixers apart from everyone else is something I call the “New York Compression Trick.” It seems as if every mixer who's ever mixed in New York City comes away with this parallel compression maneuver. Even if you don't mix in NYC, once you try it you just might find yourself using this trick all the time, because it's indeed a useful method to make a rhythm section rock.

## Here's the trick:

1. Route the drums, and even the bass, via a send or buss to a stereo compressor or two DAW channels with the compressor inserted.
2. Adjust the threshold of the compressor so there's at least 10 dB of compression. (Use even more if it sounds good.)
3. Return the output of the compressor to a new pair of input channels. (No need to do this if you're using a DAW since you already have them set up.)
4. Add some high end (6 to 10 dB at 10 kHz) and low end (6 to 10 dB at 100 Hz) to the compressed signal. (This is optional, as you'll still get the sound without the EQ.)
5. Bring the fader levels of the compressor channels up until they're tucked just under the present rhythm section mix to where you can just hear it.

The rhythm section will now sound bigger and more controlled without sounding overly compressed.

“What I will do a lot is buss my drums to another stereo compressor and blend that in just under the uncompressed signal. Sometimes if everything sounds good but the bass and kick drum aren't locked together or big enough to glue the record together, I'll take the kick and bass and buss them together to a separate compressor, squish that a fair amount, and blend it back in. I'll add a little bottom end to that if the record still isn't big enough on the bottom. This helps fit the bass and kick lower on the record and gets it out of the way of the vocal.” - Joe Chiccarelli

## Compression on Individual Instruments

Back in the early days of recording, a studio was lucky to have more than a couple of compressors available, which meant that you had to be judicious in what you used them on. This all changed as recording became more and more sophisticated, even to the point where eventually large-format consoles came with dynamics on every channel. Now when it comes to adding dynamics in a DAW, the limiting factor is the processing power available in the computer. While it may not be necessary or even desirable to add compression on every track in a session, sometimes tracks can benefit from just a slight touch. Let's look at the effect compression can have on individual music tracks.

"To me, the key to compression is that it makes the instrument sound like it's being turned up, not being turned down. If you choose the wrong compressor or you use it the wrong way, then your stuff can sound like it's always going away from you. If you use the correct compressor for the job, you can make it sound like, “Man, these guys are coming at you.' It's very active and aggressive." - Ed Seay

## A Drum-Compression Primer

One thing that most modern mixes call for is a punchy drum sound, and this can be achieved with compression, although a great acoustic drum sound, a great drum, and a great recording are certainly all major components to the sound. It would be great if every drummer hit every beat on the kick and snare with somewhat the same intensity, but unfortunately that doesn't even always happen with the best session drummers you can find.

Some intensity changes are both proper and natural in music, but when the intensity noticeably changes from beat to beat, the pulse of the song can feel erratic and sometimes even a slight change in level can make the drums feel a lot less solid than they should be. Compression works wonders to even out those erratic hits and helps to push the kick and snare forward in the track to make them feel punchier. Let's take a look at how to do that with the drums.

## Compressing the Kick and Snare

The biggest question that arises when compressing either the kick or the snare is, “How much is enough?” This depends first and foremost on the sound of the drum itself and the skill of the drummer. A well-tuned drum kit that sounds great in the room should record well, and a reasonably good drummer with some studio experience usually means that less compression is needed because the hits are fairly even to begin with. Even a great drummer with a wonders for the sound in many situations. With only that amount, the setup of the compressor is a lot less crucial, are less understandable than others. If so, increase to about 6 dB and listen again. Don't be afraid to increase great-sounding kit can sometimes benefit from a bit of compression, though, and as little as a dB or two can work especially the attack and release.

Sometimes you need the kick or snare to cut through the mix and seem as if it's in your face, and that's when 3 to 6 dB or so does the job. It's here that the setup of the compressor is critical because you're imparting its sound on the drum. Make sure to tweak the Attack and Release controls, as outlined earlier, as well as the Ratio control, and even try a number of different compressors. You'll find they all react differently, even with the same settings, so it's worth the time to experiment.

**TIP:** If the attack is set too fast, the drum will sound less punchy, regardless of how much or how little compression you use.

## Compressing the Room Mics

Room ambient mics are meant to add the “glue” to the sound of a kit and can really benefit from a fair amount of compression, which means anywhere from 6 to 10 dB. In fact, many mixers prefer the room sound to be extremely compressed, with way more than 10 dB being the norm.

The problem is that the more compression you use, the more the ambience of the room is emphasized. That's okay if the room tracks were recorded in a great-sounding acoustic space, but if it has a lot of bad-sounding reflections and the ceiling is low, you may be emphasizing something that just doesn't add much to the track.

**TIP:** If the ambience on the room-mic tracks sounds bad, set the attack time so it's much shorter than usual to cut off the sound of the initial drum transient, and keep the release time short so the ambience isn't emphasized, then tuck the room tracks in just under the other drum tracks.

Note that regardless of how good the room mics sound, the more of them you use, the less space there will be for the other instruments in the track. The more instruments there are, the more you'll have to back them off. Sad but true, but unfortunately, there's only so much sonic space to any mix.

## Compressing Vocals

Most singers aren't able to sing every word or line at the same level, and some words get buried in the mix as a result. Compression helps to even out the level differences so it's easier to hear every word.

Depending on the vocal style, the song and arrangement, the mic technique, and any number of other factors, the amount of compression required on a vocal can vary a great deal. Some vocalists are so consistent that only a dB or two is needed, but it's not uncommon to use as much as 10 dB or more on some vocals either. Here's how to set up the compressor on a vocal.

1. Solo the vocal and insert a compressor on its channel. Begin with the ratio set to 4:1, but experiment with higher ratios if you're not getting the sound or control that you're looking for. You can also use a 2:1 ratio with more compression to make it sound smoother with less of the sound of the compressor.
2. Try setting the Threshold to where there's about 2 dB of compression and notice if there are any words that the more color it will add.
3. Set the Attack and Release controls as described previously to breathe with the track, but also experiment with extreme settings.
4. Bypass the compressor and listen to the level, then add makeup gain to make the before and after compression levels the same.

5. With the attack and release set, un-solo the lead vocal and listen to what it sounds like in the track, then tweak as necessary.

**TIP:** Compressing the vocal more may not alleviate the need for automating the track, because even though the vocal may stay at a steady level, the density of the arrangement may change around it.

## Compressing Loops

Much of modern music is derived from samples and loops, but a danger lurks when compressing anything from a loop library, because most of the components may be heavily compressed already. A potential byproduct of using additional compression on a loop is that it could change its groove, which won't be desirable in most cases. That said, sometimes just a few dB of compression can handle any peak that still might exist and allow it to sit better in the mix.

## Compression on the Mix Buss

Along with compressing individual tracks, many engineers place a stereo compressor across the mix buss to affect the entire mix as well. Originally the practice came about during the late '70s, when artists began asking why their mixes sounded different in the studio than what they heard on the radio or when their record came from the pressing plant. (This was still back in the vinyl days.) Indeed, both the radio and record did sound different because an additional round or two of compression was added in both mastering and broadcast. To simulate what this would sound like, mixing engineers began to add a touch of compression across the mix buss. The problem was, everybody liked it, so now it's not uncommon for mixes to have at least a few dB of compression added to the stereo mix, despite the fact that it will probably be re-compressed again at mastering and yet again if played on the radio or television.

"Compression is like this drug that you can't get enough of. You squish things and it feels great and it sounds exciting, but the next day you come back and you're saying, 'Oh God, it's too much,' so I've been trying to really back it off, especially with stereo buss compression." - Joe Chilcarelli

When it comes to compressing the mix buss, not all compressors are up to the task. Because only a few dB of compression may be all that's added (although it can be a lot more), the compressor itself actually adds an intangible sonic quality. Among the many favorites are the Fairchild 670 and the Neve 33609.

"Generally, the stereo buss itself will go through a Fairchild 670 (serial #7). Sometimes I'll use a Neve 33609 depending on the song I don't use much, only a dB or 2. There's no rule about it. I'll start with it just on with no threshold, just to hear it." - Don Smith

Generally you'll find that most renowned mixers use the buss compressor to add a sort of “glue” to the track so the instruments fit together better, but that doesn't necessarily mean that it requires a great deal of compression. In fact, sometimes only a dB or two of gain reduction at the most is added for the final mix. That being said, many mixers will also offer their clients (artists, band members, producers, and label execs) a more compressed version to simulate what it will sound like after it's mastered. This “client mix” is achieved by using a signal path across the mix buss
that's similar to what a mastering engineer would use—that is, a compressor that's fed into a limiter at the end of the chain to raise the level to sound like that of a mastered release.

## Client Mix Signal Chain

```
[Compressor]
   |
   |
[Limiter]
   |
   |
Mix Bus
Master Fader
```

Because the clients get used to hearing the “client mix,” it's easy for the heavy buss compression to get out of hand. One of the problems with compressing too much is that it leaves the mastering engineer with a lot less room to work, and in the case of a track that's “hyper-compressed,” it virtually eliminates the ability for the mastering engineer to be of much help at all. That's why care should be exercised anytime compression is added to the mix buss.

That said, there are two times during the mix that you can insert your buss compression: when you first start the mix or toward completion. While the two choices might not seem all that different, you will get a slightly different result from each.

## At the End of the Mix

If you wait to insert the buss compressor until later in the mix, the compressor settings can be less aggressive, because a fair amount of compression may have been already inserted on the individual tracks. If you choose to wait until later in the mix, usually the best time to insert it is at the point where most of your elements and effects have already been added and it's now time to concentrate on final balances.

One advantage of inserting buss compression toward the end is that if you don't like the sound, you can easily substitute a different compressor or even eliminate it all together.

**TIP:** If you're using the right compressor, it may take only a dB or two of buss compression to make a sonic difference to where the mix is bigger sounding.

## At the Beginning of the Mix

The other approach is to insert the buss compressor right at the start of the mix and build your mix into it. Because this affects the dynamics of the mix right from the beginning, mixing this way might take a little getting used to, but it has some advantages.

First, the mix comes together a little quicker since it has that "glue" almost right away as a result. Second, you may find yourself using a little less compression on the individual tracks. This has the secondary benefit of giving you greater control of the overall compression of the mix. If you feel like there's too much, it's easy to back off on the buss compressor to the point where you or your client feel better about the amount, whereas if you added it toward the end, sometimes the only way to dial the total compression of the mix back is to tweak the individual instrument compression, which can take quite a bit of time and rebalancing.

The third thing is that the buss compressor tends to even out the levels of the individual instruments a lot, so you might not need to automate the fader levels as much. The downside of doing it this way is that if you decide you don't like the sound of the compressor, the overall sound and balance of the mix can change a lot when you insert a different one.

Finally, when starting with the buss compressor in the signal path from the beginning of the mix, you may find that you'll be using somewhat more buss compression than if it was introduced toward the end of the mix.

## The SSL Mix Buss Compressor

The sound of a great many records from the '80s and '90s comes from the sound of the built-in mix buss compressor on an SSL console. This is an aggressive compressor with a very distinct sonic signature. Some have even gone so far as to call the compressor In button (meaning it's present in the signal path) the “Good” button, because it makes everything sound better.

If you happen to get a chance to work on an SSL console (any vintage—they all have a mix buss compressor), the outboard rack-mount version, the version by Alan Smart, or any of the various plug-in emulators, here's the time-honored setting to use as a starting point.

**TIP:** The typical SSL buss compressor settings are:

**Attack:** All the way slow.

**Release:** All the way fast.

**Ratio:** 4:1.

**Threshold:** To taste.

## Compression Techniques

Here are a number of techniques often used when adding compression to particular mix elements. Don't limit yourself to just these examples, as the settings can just as easily work for other instruments in other situations as well.

### For Snare:

- To compress the snare drum to get more sustain so it sounds bigger, go to a part of the song where the drums are playing straight time and adjust the release time so that the sound doesn't begin to attenuate until the next snare hit.
- To remove snare leakage from the overheads without making the hi-hat sound too ambient, compress the overhead mics and key them from the snare by sending the snare signal into the sidechain input of the over-heads' compressor.
- Instead of adding more high-end EQ to the snare, try compressing it instead, but be sure that the attack is set fairly long so that the initial transient does not get compressed. This will allow you to elongate the snare drum's duration and create the illusion that it is brighter.

"What I do a lot is take a snare drum and go through an LA-2, just totally compress it, and then crank up the output so it's totally distorted and edge it in a little bit behind the actual drum. You don't notice the distortion on the track, but it adds a lot of tone in the snare." - Ed Stasium

If a sample is being added to the snare (see Chapter 11 for more on how that's done), compress the original snare and not the sample, because the sample usually has fewer dynamics. This works when replacing other drums as well.

### For Kick:

- For a punchy kick, set the attack time slow enough to let the initial attack through, and set the release time so that it just begins to die at the next kick beat. Increase the ratio to make it punchier-sounding.
- For a floppy-sounding kick (either with or without a front head), set the release time shorter than normal to deemphasize the end portion of the kick signal.
- Synthesized kicks like from the famous 808 inherently have a lot of low end on them that sounds better when it's controlled. Try some parallel compression by sending it to another channel that has a compressor inserted, compressing it with a very fast attack and release, and mixing the compressed signal back in with the original to help punch it up.

## For Room Mics:

Compress the room mics by 10 to 20 dB to increase the room sound with a slow attack time and the release time fairly long or timed to the track. If the acoustics of the room sound good to begin with, it will sound tighter and better than an outboard reverb.

## For Bass:

- Set the threshold of a compressor to its highest ratio (even if it's infinity:1) with 3 or 4 dB of gain reduction. This will keep the bass solid with the level constant in the mix.
- With a bass track that has no definition to the notes, try an 1176 with the attack set to around the middle and the release set to around 3 or 4 o'clock with an 8:1 ratio and a fair amount of gain reduction. This helps put a front end on the notes that may not have been articulated properly when the part was recorded.
- With a bass track where the notes don't sustain long enough, increase the release to where the longest notes don't die until the next one sounds.
- If a bass player is using a pick, it may create high midrange transients that might need to be limited even before compression is added. Insert a limiter first and set the attack and release times fairly fast, then insert a compressor set from medium to slow on both attack and release and a ratio of 4:1.
- On some rock bass parts played with a pick, a multiband compressor across the bass can achieve a more even level without sounding compressed.

## For Vocal:

A good starting point for a lead vocal is a 4:1 ratio, medium to fast attack and medium release, and the threshold set for about 4 to 6 dB of gain reduction.

When a single compressor or limiter just isn't enough for a troublesome vocal, try an 1176 (hardware or plug-in) set to fast attack (all the way clockwise, which is backward from other compressors), the release set to medium (5), and the ratio set to either 8 or 12:1 to clip the peaks by 4 to 5 dB. Feed the output of the 1176 into an LA-2 (again, hardware or software) set for gentle gain riding of 2 to 3 dB.

As an option to above, if you have access to another 1176 or have enough DSP to insert a second, set the first one on fastest attack and an 8:1 ratio, and the second on a slower attack and a 4:1 ratio. With both set for about 4 to 6 dB of compression, the vocal will be silky and smooth, yet in your face.

## For Piano:

If you liked the early Elton John piano sound, put the piano into two LA-2As or similar compressors (or stereo versions of the plug-in) and compress the signal a large amount (at least 10 dB), then connect the output into two pultecs or similar equalizers. Add 14 kHz and 100 Hz to taste. The effect should be a shimmering combination of sound where the chords hold and seem to chorus.

## For Guitar:

Higher ratios of compression around 8 or 10:1 sometimes work well, with the threshold set so that the guitar cuts through the track. Attack and release should be timed to the pulse of the song.

"I may go 20:1 on a [UREI) 1176 with 20 dB of compression on a guitar part as an effect. In general, if it's well recorded, I'll do it just lightly for peaks here and there." - Don Smith

When a guitar is recorded with both close and distant mic tracks, you can use this trick to get guitars to sound big, yet stay out of the way of the vocal. Pan the distance mic in the same direction of the close mic and then attach a compressor across the distant track keyed off the lead vocal. When the lead vocal is present, the ambience is decreased. When the lead vocal stops, the ambience returns to full level. As a variation, try panning the distance-mic track to the opposite side of the close-mic track.

With doubled guitars, pan them medium left and right, then send the same amount of both to a compressor on a separate track, either via an aux send or a buss. Pan the output of the compressor track to the center and bring it up just underneath the original tracks. Now the guitars don't have to be as loud as before to still have presence.

If you notice a guitar getting lost in places, try compressing with a ratio of 2:1 to 4:1 with medium attack and release times. This will allow the rhythmic transients to get through somewhat uncompressed while boosting the sustain of the guitar sound.

With rock guitars, the idea is to have them big and "in your face." This is accomplished by first limiting the transients from the signal with a ratio of 10:1 or higher. Be careful to set the release so that any sustaining parts of the signal return to unity gain before the next transient.

# Using a De-Esser

Sibilance is a short burst of high-frequency energy where the S's are overemphasized, which comes from a combination of mike technique by the vocalist, the type of mic used, and heavy compression on the vocal track. Sibilance is generally felt to be highly undesirable, so a special type of compressor called a de-esser is used to suppress it.

Most de-essers have two main controls, Threshold and Frequency, which are used to compress only a very narrow band of frequencies anywhere between 3 kHz and 10 kHz to eliminate sibilance. Modern software de-essers are much more sophisticated, but the bulk of the setup still revolves around those two parameters. One frequently used additional feature is a Listen button that allows you to solo only the frequencies that are being compressed, which can be helpful in finding the exact band of offending frequencies.

### To use a de-esser, do the following:

- Insert the de-esser on the vocal channel and solo it.
- Lower the Threshold control until the sibilance is decreased, but you can still hear the S's. If you can't hear them, then you've raised the Threshold too far.
- Scan the available frequencies with the Frequency control until you find the exact spot where it's most offensive, then adjust the Threshold control until the S's sound more natural.
- Un-solo the vocal and listen in context with the track. Be careful that you don't completely eliminate all the S's, because that makes it sound unnatural.

**TIP:** When using the Listen feature, remember that the audio you're hearing isn't in the signal path, just the sidechain. Don't forget to disengage Listen when you've found the correct frequencies.

# Using a Gate

Like the de-esser, a gate can sometimes consist of just a few controls, principally the Threshold, Range, and sometimes a Hold or Release control (see Figure 9.16). Range sets the amount of attenuation after the threshold is reached and the gate turns on. Sometimes when gating drums, the Range control is set to attenuate the signal only about 10 or 20 dB. This lets some of the natural ambience remain and prevents the drums from sounding choked. The Hold control keeps the gate open for a defined amount of time, and the Release control sets how quickly the gate closes again.

We'll use the snare drum as an example of how to set up a gate, but the same technique is used to set up a gate on any drum, instrument, or vocal.

1. Insert the gate on the snare channel and solo it.
2. Raise the Threshold control until you can hear the snare drum hit, but there's no sound in between the hits.
3. If the snare sounds unnatural or cut off, raise the Threshold a bit to see whether that improves the sound.
4. If the snare still sounds unnatural or cut off, try increasing either the Hold or the Release control.
5. Try adjusting the Range control so the snare is attenuated by 10 dB between hits to hear whether it sounds more natural.
6. If the gate chatters, try fine-tuning the settings of the Threshold and Release controls (if the gate has one or both). This may require some experimenting, so be patient.
7. Try timing the Release control so that the gate breathes with the pulse of the song.

**TIP:** Gates are finicky because the dynamics of the signal are usually constantly changing. Inserting a compressor in the signal chain before the gate improves the performance greatly.

## Gating Techniques

Here are some techniques often used when gating a mix element. Gates usually take time to set up properly, so take your time and don't be afraid to experiment.

### For Snare:

- This simple technique allows a different effect to be placed on the snare during harder hits and prevents leakage to the effect during tom hits, hi-hat, or stray kick-drum beats. Duplicate the snare to another channel and insert a gate on this new channel. This gated channel is generally not sent to the main mix (although it can be), but it is used primarily as an effects send. By adjusting the threshold, you can have more control over how the signal is sent to the effects unit.
- Another way to make the snare feel as if it's breathing with the track is to copy the snare onto a second track and insert a gate on only that channel. Time the release so it cuts off right before the next snare hit during the main part of the song when the snare is just playing time. Bring the gated snare channel up just underneath the main snare track.
- To make the snare drum sound bigger, gate either room ambience or reverb and trigger it from the snare by sending the snare signal to the trigger/key input of the gate.

### For Drums:

- When gating toms, set the Range control so it attenuates the signal only about 10 or 20 dB. This lets some of the natural ambience remain and prevents the drums from sounding choked or unnatural.
- To make the rhythm section feel tighter, feed the bass through a gate with only 2 or 3 dB of attenuation when the gate is closed. Trigger the gate from the kick drum so that the bass is slightly louder on each kick-drum beat.
- For times when the groove doesn't quite lock or the bass player is playing on top of the beat and the drummer is laying back, insert a gate on the bass channel and key it from the kick drum. The bass will only play when the kick drum is played, so the rhythm section will now sound tight. Set the sustain and release so that it sounds more natural and less staccato, since the kick is a transient instrument and the bass is not, or leave it staccato if that works well.
- The above technique can also be used to get more space and rhythm in a big chorus by gating the rhythm-guitar track keyed from the hi-hat.
- Key a gate placed across a synth or keyboard pad from the hi-hat to make the pad more rhythmic.
- For tighter background vocals, patch the best-phrased harmony line into the key insert of a gate across a stereo submix of the harmonies. This will ensure that all of the vocal entrances and exits will remain tight.

### Advanced Techniques

Whereas a decade ago a mix was deemed complete as soon as it felt good to everyone involved, mixes these days require more precision than ever before. This is because the mindset of a mixer has changed thanks to the ability to now perform many more mix moves in a repeatable manner inside a DAW. Here we'll look at a few of the more advanced techniques used to make a mix competitive today. Keep in mind that cleanup, timing adjustment, and pitch correction are considered more a part of production and should be completed before the track is sent to be mixed, but mixers are sometimes expected to perform these tasks anyway.

# Cleanup

As stated in before, a big part of today's mixing can be track cleanup. Although it's always preferable for this to take place prior to mixing, sometimes the previous engineers don't have time, are careless, or maybe even don't know how to properly clean up the tracks before mixing begins.

What's meant by cleanup? For the tracks to be as clean as possible with the least amount of aural distraction, it's important that any clicks or pops are removed, the bad fades are fixed, and any noise is eliminated. Here's how to do it.

## Removing Noise

Noise on a track can mean anything from the guitar-amp buzz before the guitarist plays a part, to the shuffling of feet or the clearing of the throat captured by the vocal mic before the singer begins to sing. As a matter of course, the length of each audio clip should be trimmed to just before and after the part plays to eliminate the noise. Be sure to add a short fade-in or fade-out to eliminate any potential clicks.

- **Noise at the beginning of a clip -** Clip shortened to eliminate the noise
- **Fade at the beginning of a clip.** While shortening the clip doesn't usually pertain to the drum track because doing so can make it sound choppy and unnatural, there are times when it's appropriate, such as in the case where there's a measure or more of silence in the song. Be sure to use a fade that both sounds natural and doesn't cut off the cymbal decay or the attack of the next downbeat after the silence.
- **Adding fades to a drum track.** Fade to silence
- **Removing Clicks and Pops.** Clicks and pops can come from a number of situations, including butt-cut edits (edits without fades that are butted up against one another) or transient overloads, such as from a thumb-slapped bass with active pickups. Most clicks occur during cut and paste, timing adjustment, or noise elimination operations where a fade isn't used so only a butt-cut remains. Although sometimes the click is very apparent, many times it's quiet enough that it will get lost in the track when the other instruments are introduced. Sometimes an edit can be completely silent if it happens to be made in the right place in the audio waveform.

**TIP:** While you may not hear the clicks when the track is played back over monitors, you may find that they jump right out when you're listening with headphones. That's why it's best to always listen to a pass with phones to see whether any clicks are audible.

The way to fix any potential trouble spots is by adding fades to the edits. This is something that you can do by eye, although it's time consuming. While you may hear clicks on butt-cuts at the beginning and ending of clips, it's more likely that the ones that will stick out are edits that lack crossfades. Go through the track and add crossfades where appropriate.

### Cross-fade across both clips

Remember that even though these clicks might not be readily apparent, any kind of noise accumulates and plays a very subtle role in how the listener perceives the song. Sometimes he can't explain the difference, but he can hear it.

## Removing Count-Offs

Leaving count-offs (some musicians call them “count-ins”), such as drum-stick clicks, is a sure sign of a demo recording and is something that no one wants to listen to. While you can set any DAW to begin precisely on the downbeat of the song after the count-off, it's best to eliminate the count-off altogether, especially if there's another instrument that's beginning the song against what's supposed to be silence.

There are two ways to eliminate a count-off; either shorten the clip length as we did previously for noise elimination, or make a cut so there's a separate clip containing the count and then mute it. Muting the clip leaves the count intact in the event that the count is needed later for additional overdubs (although this shouldn't be needed again once it's time to mix). Create a separate count clip and muting it.

### Fixing Bad Fades

Sometimes adding a default fade just doesn't sound natural. Either the fade is too tight and cuts off the attack or release of the part, or the fade itself just isn't smooth-sounding enough. Now is the time to fix any fades that don't work in the track by adjusting the fade timings.

### Exponential Power Fade

In the case of a fade that seems unnatural (especially a fade-out), try an exponential power fade instead of the default fade.

### Eliminating Unwanted Distortion

Distortion can be anything from a breath blast to a note on a direct bass that's been clipped. In both cases it's the transient of the attack that causes the problem, but thankfully it's extremely short, and there are a number of ways to either fix it or mask it.

### Replacement

Sometimes the most natural-sounding operation is to replace the area of distortion with a similar area from another part of the song that's clean. That means finding a similar piece, copying it, and then pasting it over the section of the clip that has the distortion or noise.

**Transient causes pop:**

1. Part from another section of the song
2. Copy and paste to eliminate distortion.

## Clip Level Adjustment

If a clean section of the song isn't available to copy and paste, the next thing to try is to adjust the level of just the transient. Make a cut around the transient and lower the level until it sounds natural, provided your DAW has a clip-based level feature.

## Automation

If a clip level adjustment feature isn't available on your DAW, you can accomplish the same thing using track automation. Draw the automation curve so that the level of the transient decreases to a point where it seems natural.

## Elimination

Sometimes a transient passes so quickly that you can easily delete it without it being noticed. Be sure to add fades to the new edits so as to not create a new click.

## Deleting Extra MIDI Notes

Delete any extra “split” notes that were mistakenly played when the part was originally recorded. You might not hear all the notes play distinctly when all the instruments of the track are playing together, but just like the noise at the beginning of tracks, they have a tendency to come to the forefront after things get compressed (see Figure 11.13) and are cumulative in nature, just like noise.

## Adjust the Timing

No matter how great the players on the session are, there's always some portion of a recording that doesn't feel quite right. Usually, the timing of the basic tracks will be tweaked right after your tracking session so you have a solid rhythm section to overdub against, but if you're just now discovering some sections of an overdub that don't feel right (which happens more than you might think), prepare for the joys of slipping and sliding time.

Here's a list of some of the dos and don'ts for tweaking the track timing:

- Don't edit by eye. In most music (electronic music being the exception), you can't successfully edit by just trying to line everything up to the kick and snare or the grid and still have it sound natural and human. Often, tracks that look perfectly lined up don't sound or feel right, which is why listening is more important than looking. Turn your head away from the monitor or close your eyes and just listen before and after
- move anything.
- Every beat doesn't have to be perfect. In fact, if it's too perfect, you'll suck the life out of the performance. Unless something really jumps out as being out of time, you might get away with leaving it as is. Another way is to just line up downbeats and any major accents, which gives you the best of both worlds: a loose feel that still sounds tight.
- Copy and paste another section. If you have to make too many edits to a particular section, chances are it won't sound as good when you're finished as just finding a similar section in another part of the song and pasting it in over the area that's suspect. It's a lot faster and easier to do and will probably sound cleaner and groove better as well.

**TIP:** Many times the bass will speak better if it's a few milliseconds behind the kick drum rather than right with it. It still sounds tight, but both the kick and bass will be more distinct, and the sound may even be fuller.

- Listen against the drums. If you listen to the track that you're editing all by itself, you can be fooled into thinking that the timing is correct when it's not, especially if you're editing to a grid. The real proof is when you listen against the drums. If the instrument sounds great by itself and great with the drums, you're home free.
- Trim the releases. This is one of the best things you can do to tighten up a track. Everyone is hip to tightening up the attacks, but it's the releases that really make the difference. Regardless of whether it's an accent played by the full band, the song ending, or a vocal or guitar phrase, make sure that the releases are pretty
- much the same length. If one is longer than the rest, trim it back and fade it so it sounds natural. If one is a lot shorter than the rest, use a time-correction plug-in to lengthen it a bit.

Of course, if you're using loops or MIDI instruments, you've probably quantized things to the track by now. If you haven't, remember that if it's too perfect to the grid, it may not sound natural.

## Pitch Correction

Depending upon how much of a purist you are, pitch correction is either the worst thing to ever happen or a godsend. Regardless of how you come down on the issue, it's at the very least a necessary evil in today's music.

Tuning vocals has been done since way back in the early '70s, starting with the first Eventide H910 Harmonizer. Primitive as it was, it did allow for slight pitch changes, although the digital artifacts that it imposed on the sound were quite substantial. With every new model of pitch-shift hardware that was subsequently introduced, the technology improved to the point where today there are some excellent plug-ins commonly available that would simply astound any engineer transported in time from back then.

There are three popular track-tuning programs commonly in use: Antares Auto-Tune, Waves Waves Tune, and Celemony Melodyne, as well as less often used variations, such as Avid's Elastic Audio or Cubase VariAudio. Be aware that all tuning programs impart their own sound on the audio you're tuning, and it might not always be pleasing, which is why many engineers own several different ones so they can compare which one sounds better in a particular situation.

Although the way pitch-correction plug-ins are used is somewhat the same, there are a number of guidelines that are worth following:

- **Use the performance itself first**. Before you apply pitch correction, exhaust all other remedies to keep the performance as natural-sounding as possible. These include vocal comping and copy and pasting phrases, words, or syllables from other parts of the song.
- **A little goes a long way**. The fewer notes you correct, the more natural the performance will sound. You're much better off just correcting a few notes than attempting to correct the entire performance.

**TIP:** Generally, background vocals can get away with much more pitch correction than lead vocals without being heard.

- **Use the most precise mode**. Most engineers avoid using Auto mode because it's not precise enough for most applications, and as a result, it causes audible fluctuations that make the vocal sound like it's tuned, which is usually not what you're after (Cher and T-Pain aside). Use the graphical mode if the plug-in has one in order to achieve the most precise tuning with the least amount of audible artifacts.

- **Don't perfectly tune the vocals**. Even the best vocalists are never precisely on pitch, so if you tune it that way, it may sound unnatural. Getting the pitch within a few cents will sound more like the real thing, since it's the variations and inaccuracies that make a human voice sound human.
- **Print the pitch correction**. Instead of leaving the pitch correction patched in as a plug-in, you're better off to print a corrected pass and use that vocal instead. This not only saves precious system resources, but it also eliminates any problems that might occur if the session is moved to a DAW with a different software version.

## Pitch Correction Techniques

Here are a few techniques often used when correcting the pitch of a mix element. As always, don't be afraid to experiment, since slight variations might fit better on a particular mix.

- Sometimes raising the formants (the harmonic resonance) of a voice can make the vocal sound a bit more exciting or breathier. It doesn't actually change the pitch, just the placement of the voice's harmonics.
- If the vocal is off enough that the pitch correction sounds robotic, try this trick from the old Harmonizer days. Copy the vocal to two additional tracks and spread them left and right. Tune one of the vocals up by 2 to 8 cents; tune the other down by 2 to 8 cents. This will smooth out an out-of-tune vocal and make it sound a lot thicker at the same time.
- Tuned vocals almost always sound better with a least a touch of reverb or delay on them.

## Sound Replacement

Although you may be great at recording drums and have a great-sounding studio with an excellent signal chain, the two chief variables in the recording are the drummer and his drums. No amount of technique or gear can overcome a bad-sounding kit or a drummer that hits inconsistently, hence the importance of sound-replacement software and a good sample library.

Drum replacement is a technique pioneered by the late great engineer Roger Nichols during the '70s while working on Steely Dan records. In the days before DAWs, Roger developed his own hardware device called the Wendel that was an ultra-fast set of triggers that could replace real drum sounds with more consistent and better-sounding ones from his library.

Today there are numerous drum replacement plug-ins that do the job equally as well, but it seems like every mixer has his own particular preference, be it Drumagog, Steven Slate's Trigger, the Massey DRT, or any of the others that may be included as part of a DAW software package. The drum-sound libraries can range from those that come with the software to personally made libraries consisting of the engineer's own samples.

While the original idea was to totally replace a subpar drum track, most mixers now only do that as a last resort, opting instead to double the original sound with a sample to help keep the human feel and the basic drum sound intact. The new sampled drum acts as an acoustic EQ, changing the sound by augmentation instead of electronically changing the frequency bandwidth of the original recorded drum. Although some mixers try to delete the leakage between all of the drums of the kit, that's what gives the kit its particular sound, and doubling the original kit with a sample helps to maintain some of that relationship.

## Keeping the Sound Natural

It's very easy to make a drum track seem more like a drum machine than a human if you're not careful. For drum replacement to sound natural, the following must occur:

- The sample should sound better than the drum it's replacing.
- The sample should blend seamlessly with the other drums in the kit.
- No one should be able to tell that you've replaced the sound.

To replace drums, it's important to understand that any drum sound is composed of two different parts: the initial transient hit, and the resonant sustain and decay that make up the body of the sound. It's much easier to trigger a replacement drum if the initial transient hit is strong, since it will be clear to the software exactly when to trigger the sample.

**TIP:** Sometimes it's easier and more natural sounding to replace a sound manually by copying and pasting a hit or fill from another place in the song, especially if there are only a few hits that need replacing.

It's usually best to record the replacement samples each to a new track rather than to keep the drum-replacement plug-in inserted as a plug-in during the mix. This saves system resources (some sound-replacement plugs are resource hogs) and makes your mix more transportable because the sounds are printed.

## Sound-Replacement Techniques

There are many techniques when it comes to sound replacement or enhancement. While most refer to drums, it's possible to use some of the same techniques in other situations as well if you're willing to experiment.

- If the original drum sounds okay soloed but disappears in the track, select a sample with a high initial transient. If the original tone of the drum doesn't fit the song, select a sample based on the character of the body of the sound after the initial hit.

- The simpler the drum pattern, the easier it is for the sound-replacement plug-in to recognize the parts.
- Most drums become brighter the harder they're hit; therefore, choose a sample that has these characteristics in order for it to sound natural.
- If the drummer plays the snare using mostly a sidestick, be sure that the sample has mostly sidestick as well.
- Timing is extremely important when adding a sample to the original sound, so take care to ensure that both transients line up together. Many sound-replacement plug-ins have some latency, so be sure to check this closely and move the sample as necessary.
- The easiest way to find the transient is to use the Tab to Transient feature in Pro Tools, Hit Detection in Nuendo, or any other transient-detection feature that your DAW has to offer. As an alternative, you can use any available Strip Silence-like feature on a copy of the original track to make the transients more distinct.
- Well-played drums with a few ghost notes will beat robotic-sounding, replaced drums any day.
- It's critical that you check the phase alignment of the sample against the original drum. Even though it may look to be in phase, your ears may tell you something completely different when the Phase button is selected.
- Snare drum is the hardest to augment/replace because of the nuances in how hard and how frequently the drummer hits it. Be sure to use a multi-sampled snare drum with multiple level variations to keep it sounding like it was played by a real drummer.
- If you don't have a multi-sampled snare or the sound doesn't have enough sample levels, you can make your own by duplicating the sample a few times and raising and lowering each one by 2 cents. Slightly changing the pitch gives the impression that the snare is hit a bit louder or softer.
- If the drummer plays a lot of ghost notes on the snare, sometimes it's best to make a copy of them and concentrate on replacing only those. This allows you to fine-tune the sound-replacement software only for ghost notes.
- Many times the best way to trigger the snare drum is by using the sound from the under-snare mic if one has been recorded. It's usually a highly transient signal that can be easily recognized by the sound-replacement plug-in.
- Sometimes using a gate on the original drum track can maintain just the transient hit, making the sample easier to trigger. This works especially well if you're trying to replace the body sound of the drum rather than the hit part.
- Sometimes compression of the drum track will provide a more consistent drum trigger.
- Many engineers always have the drummer give them solo hits of each drum while recording so they can use them as samples later.

**TIP:** You can make your own samples especially for the song by taking the best-sounding hits from the recording of the kit you're mixing and triggering them as needed.

## Automation

Automation is the process of recording parameter adjustments so that your DAW software can automatically execute them during playback. Before automation, any mixing moves during a complex song had to be made by the engineer, the assistant, the band members, and anyone else who got their arms near the console. While this might have been a very "organic" way of mixing, it was not repeatable in the slightest, so engineers everywhere longed for a way that at least their fader moves could be memorized and played back later.

The first primitive automation on consoles came during the 1970s and was known as VCA automation, which revolved around a component called a voltage-controlled amplifier. It used two tracks of the tape machine to store the automation data and any updates, and while it remembered your fader moves, the faders themselves didn't actually move. This was obviously confusing and became really unwieldy as consoles became larger and larger, plus having the VCAs in the signal path degraded the sound as well.

The next generation of console automation system actually provided moving faders (the first one was introduced by Neve and called NECAM), which were extremely expensive, as was the corresponding computer that memorized their movements. Don't forget, this was in the pioneering days of computer science, when even 500 kilobytes of RAM (yes, kilobytes, not the far larger gigabytes that we're used to today) cost thousands of dollars.

Modern mixing has always been a dynamic operation where a long list of console parameters are set during a mix, so it's important to remember exactly each parameter's setting position in order to make a remix of a song easier at a later date. The first generation of console automation to make that possible was SSL's Total Recall in 1981, which took a snapshot of all the switches and parameter controls on the desk. The problem was that the reset had to be done manually, usually by an assistant, which could take two or three hours to complete on a console with 48 or more channels.

For a brief period, resettable consoles became all the rage, where most of the console parameters would reset themselves automatically without the help of a human. Although this evolution came about in the analog days, it became much easier and cheaper to implement as digital came of age and is pretty much standard in most digital consoles today, even those costing as little as $1,000 or so.

But now we're in the age of the DAW, where dynamic automation of virtually every parameter is the norm. This allows mixes to be more intricate than ever and take more time than ever as a result, but the pinpoint accuracy every parameter movement during every millisecond of a mix is assured.

## Fader Automation

Most automation systems, regardless of whether they're in hardware on a console or in a DAW, have the following modes:

**Write** Used the first time the automation data of a channel is recorded or when writing over the existing automation data.

**Touch** Writes automation data only while a fader is touched. The fader then returns to its previously automated position after it's been released.

**Latch** Starts writing automation data when a fader is touched and stays in that position after it's released.

**Read** The written automation moves are replayed.

**Off** The automation is temporarily disabled.

Besides the fader moves, all of the above modes include the fader and the mute button at the very least, but they extend to other parameters, such as panning, aux send levels, insert in/out, and other parameters, as well on most DAWs.

Generally speaking, most mixes are begun with the automation off until the basic balance is complete and any compression or effects are inserted. The first automated pass will be in the Write mode to initially record the static balance and parameter settings. From that point, most automation is written in Touch mode, where the changes are written as long as the parameter is being changed but then return to their initial state after the changes are completed. Occasionally a track may be written in Latch mode, where a brief change is required but it must then carry on until the end of the track.

It's not uncommon for a track to have many automation moves but require that the level of the entire track, including all the moves, be either raised or lowered in level. Most automation systems have what's known as a Trim control that allows the level of the entire track, or sometimes just a portion of it, to be changed by a selected amount.

## Drawing the Automation

many cases, it's much easier to draw the automation in than to use a fader if you need a high level of precision. This can work if you know exactly the curve you want the automation to look like. Sometimes it's easier to adjust the curve of what was previously recorded as fader automation, perhaps to trim the move shorter or longer.

## Using Automation to Add Dynamics

For the most part, a mix where the faders remain more or less static can be boring and unexciting. Even before automation, mixers were constantly riding instrument and vocal faders during a mix to make sure they stood out in certain places or added an extra intensity to the mix. The best part about automation is that those moves can be
exactly replicated on every playback.

**TIP:** The key to understanding how to use automation to add dynamics is by observing a performance by a great band. This will help you be able to hear all the nuances that the dynamics of the mix need for it to be exciting.

## Among the ways to add dynamics to a mix are:

- Slightly boost the rhythm section during fills, turnarounds, and even choruses (usually only a couple of dB is all that's necessary, but it depends upon the track).
- Boost the snare and toms during fills.
- Boost the kick, snare, or cymbals on accents or the downbeat of a new section.
- Duck the rhythm instruments during an instrument solo to help clear out space in the mix.
- Boost the hi-hat in parts where it's being struck and decrease it where it's not.
- Add additional reverb or delay to an instrument when it gets masked as other instruments are added to the mix.
- Pump a strumming rhythm guitar in time with the music, pushing it especially on 2 and 4, or push it on the upbeats (one AND two AND three AND...).
- Gently boost the fills or other instruments between vocal phrases.
- Pull back the downbeat of a chorus if the drummer hits it too hard.
- Pump the “4 AND” on a percussion track.

## Automation Techniques

Each technique may apply to applications other than the one specified, so keep an open mind, especially when boosting the dynamics.

- Many old-school engineers use fader automation to control the dynamics of a vocal rather than use a compressor. This is accomplished by both riding the level of each phrase (or word or syllable) and riding the end of vocal phrases.
- Ride the bass to even out the energy differences of certain notes.
- Ride an instrument's sustain at the end of solos.
- Ride the reverb returns during parts where it gets lost.
- Use the Trim tool to adjust the automation of a track up or down slightly if necessary.
- No need to redo the track if the vibe is okay but the level isn't.
- Another way to adjust the automated track is to adjust the output volume of the last inserted plug-in up or down as needed. If the track doesn't have any plug-ins, insert any one that has the ability to control its output without inserting the effect.

## Gain Staging

Gain staging is the proper setting of the gain of each stage of the signal path so that no one section overloads. On an analog console, that would mean making sure that the input gain doesn't overload the preamp, so it wouldn't overload the equalizer section, so it wouldn't overload the panning amplifier, which wouldn't overload the fader buffer, which wouldn't overload the master buss. This is the reason why a pre-fader and after-fader listen (PFL and AFL) exist: to monitor different sections of the signal path to make sure there's no distortion.

## Subgroups

Many engineers like to group certain instruments together because it's easier to control the level of one fader than many at the same time. Some engineers prefer groups (where the faders are all linked together), while others prefer subgroups (where the output of the certain individual channels are fed into a dedicated subgroup channel) or even combinations of both.

While all of these stages were slightly tweakable, one rule exists in the analog world that aptly applies to the digital as well.

The level of the channel faders should always stay below the subgroup or master fader.

This means that the level of the master fader should always be placed higher than each of the channel faders. While it might be okay if one or two channels are slightly above the master (it's almost inevitable in every mix), just a single channel with big chunks of EQ (like +10 of a frequency band) or an insert with an effects plug-in with a level that's maxed can destroy any semblance of a good-sounding mix.

Mixing Board (Levels 1-10 For Clarity)

```
T1 | T2 | T3 | T4 | T5 | T6 | SG |
   |    |    |    |    |    |    |
09 | 09 |    |    |    |    |    |
   |    |    |    |    | 08 |    |
   |    | 07 | 07 |    |    |    |
   |    |    |    | 06 |    |    |
   |    |    |    |    |    | 05 |
   |    |    |    |    |    |    |
   |    |    |    |    |    |    |
```

Channel faders too high, subgroup fader too low.

```
SG1 | SG2 | SG3 | SG4 | SG5 | SG6 | MST |
    |     |     |     |     |     |     |
09  | 09  |     |     |     |     |     |
    |     |     |     |     | 08  |     |
    |     | 07  | 07  |     |     |     |
    |     |     |     | 06  |     |     |
    |     |     |     |     |     | 05  |
    |     |     |     |     |     |     |
    |     |     |     |     |     |     |
```

Subgroup too high, master fader too low.

```
SG1 | SG2 | SG3 | SG4 | SG5 | SG6 | MST |
    |     |     |     |     |     |     |
    |     |     |     |     |     | 09  |
08  | 08  |     |     |     |     |     |
    |     | 07  | 07  |     | 07  |     |
    |     |     |     | 06  |     |     |
    |     |     |     |     |     |     |
    |     |     |     |     |     |     |
    |     |     |     |     |     |     |
```

Channel and subgroup faders at correct levels.

Because many large analog consoles have sufficient headroom these days, this rule hasn't always been religiously followed, but it has been a golden rule since day one of modern consoles. Not following it is the main reason why many mixes, especially those done in the box, lose fidelity. The master buss is overloaded!

## Headroom

Speaking of headroom, that brings us to a second rule:
Leave lots of headroom.

Recording engineers in the digital world have been taught to increase the level of anything recorded or mixed to as close to 0 dB full scale as possible in an effort to “use up all the bits.” While this might have been useful back in the days of 16-bit recording, it doesn't apply as much today in the 24-bit and beyond world. The short reason for this is that if a piece wasn't recorded hot enough in the 16-bit days, it would be noisy as a result. That's not so much true in our 24-bit world. Today we can record at a level a lot less than 0 dBFS and not have to worry about noise, and there's a great advantage in doing so: It gives us lots of headroom and therefore less distortion.

Headroom is our friend. We need it to preserve the super-fast transients that make up the first part of the sound of just about any instrument (but especially instruments such as drums and percussion). These transients can typically range as high as 20 dB above what a VU meter might be telling you (peak meters are much closer to the true level), and recording too hot means cutting them off by going into overload if only for a millisecond or two. This results in not only a slightly dull recording but one that sounds less realistic as well. The solution: more headroom.

Headroom means that our average level might be -10 dB or less on the meter, leaving plenty of room for transients above that. This concept is actually a holdover once again from the analog days, where a really good console might have a clipping point of +28 dB at the master mix buss. Since 0 dB VU = +4 dB (trust me, it does), that means that you had a full 24 dB of headroom before you ran into distortion as long as you kept your mix hovering around 0 VU.

While leaving 24 dB for headroom might be excessive in the digital world, leaving 10 or 15 dB may not be. Since it's easy enough to make up the gain later, you won't increase the noise, and your mix will be cleaner, so why not try it?

**TIP:** When using large amounts of EQ or a plug-in with a lot of gain, lower the channel fader or the plug-in output rather than bringing up the others around it.

# MASTERING

## Eight Indicators That Your Mix Is Finished

One of the tougher things to decide when you're mixing is when the mix is finished.

**Just when is a mix considered finished? Here are some guidelines:**

1. The groove of the song is solid. The pulse of the song is strong and undeniable.

2. You can distinctly hear every mix element. Although some mix elements, such as pads, are sometimes meant to blend seamlessly into the track, most mix elements should be clearly heard.

3. Every lyric and every note of every line or solo can be heard. You don't want a single note buried. It all has to be crystal clear. Use your automation. That's what it was made for.

4. The mix has punch. The relationship between the bass and drums is in the right proportion and works well together to give the song a solid foundation.

5. The mix has a focal point. What's the most important element of the song? Make sure it's obvious to the listener.

6. The mix has contrast. If you have too much of the same effect on everything, the mix can sound washed out. Likewise, if your mix has the same intensity throughout, it can be boring to the listener. You need to have contrast between different elements, from dry to wet, from intense to less intense, to give the mix depth.

7. All noises and glitches are eliminated. This includes any count-offs, singer's breaths that seem out of place or predominant because of vocal compression, amp noise on guitar tracks before and after the guitar is playing, bad-sounding edits, and anything else that might take the listener's attention away from the track.

8. You can play your mix against songs that you love, and it holds up. This is perhaps the ultimate test. If you can get your mix in the same ballpark as many of your favorites (either things you've mixed or mixes from other artists) after you've passed the previous seven items, then you're probably home free.

## Mastering

Technically speaking, mastering is the intermediate step between taking a mix from the studio and preparing it for replication, but it's really much more. Mastering is the process of turning a collection of songs into an album by making them sound like they belong together in tone, volume, and timing (spacing between songs in physical mediums such as CD and vinyl).

In the early days of vinyl, mastering was a black art practiced by technical curmudgeons who mysteriously transferred music from the electronic medium of tape to the physical medium of vinyl. Just as mixing became more sophisticated, so did mastering. Mastering engineers soon found ways to make the vinyl records louder by applying equalization and compression. Producers and artists began to take notice that certain records would actually sound louder than others on the radio, and if it played louder then it sounded better and maybe even sold better. With that, a new breed of mastering engineer was born, this one with some creative control and ability to influence the final sound of a record rather than being just a transfer jock from medium to medium.

Today's mastering engineer doesn't practice the black art of disc cutting much, but he's no less the wizard as he continues to shape and mold a project.

## Mixing for Internet Distribution

With CDs beginning their slow decline into history, it's now important to have a handle on the online distribution options and how to mix for them. Downloads may soon follow CDs into delivery oblivion as streaming becomes the delivery method of choice, but the increased Internet bandwidth now makes it possible to deliver higher-quality audio than ever before, a trend that happily will continue. In the meantime, here are some tips on making master files for any Internet delivery method.

## MP3 Encoding

Encoding an MP3 of your mix requires a bit of thought, some knowledge, and some experimentation. In the beginning days of online music, the idea was to encode the smallest file with the highest quality, which is, of course, the tricky part. That's changed a bit since broadband DSL and cable modems have become ubiquitous, but routinely 24-bit master files still need to be data compressed.

Here are some tips to get you started in the right direction so you won't have to try every possible parameter combination of the encoder. Remember, though, that the settings that might work great for one particular song or type of music might not work on another.

## The Source File

Encoding for MP3 makes the quality of the master mix more of an issue because high-quality audio will be damaged much less by lossy encoding than low-quality audio will. Therefore, it's vitally important that you start with the best audio quality (meaning the highest sample rate and most bits) possible.

## The Encode

It's also important to listen to your encode and perhaps even try a number of different parameter settings before settling on the final product. Compare it to the original source mix and make any additional changes you feel necessary. Sometimes a big, thick wall of sound encodes terribly, and you need to ease back on the compression of

the source track. Other times, heavy compression can encode better than with a mix with more dynamics. There are a few predictions one can make after doing it for a while, but you can never be certain, so listening and adjusting is the only sure way.

Here are some things to consider if your mix is intended for MP3 distribution:

- Start with the highest quality audio file possible.
- Filter out the top end at whatever frequency works best (judge by ear). MP3 has the most difficulty with high frequencies, so filtering them out liberates lots of bits for encoding the lower and mid frequencies. You trade some top end for better quality in the rest of the spectrum.
- A really busy mix can lose punch after encoding. Sparse mixes, such as acoustic jazz trios, seem to retain more of the original audio oomph.
- Don't squash everything. Leave some dynamic range so the encoding algorithm has something to look at.
- Use multi-band compression or other dynamic spectral effects sparingly. They tend to confuse many encoding algorithms.
- Set your encoder for maximum quality, which allows it to process for best results. It doesn't take any longer, but the results can be worth it.

**TIP:** MP3 encoding almost always results in the post-encoded material being hotter than the original material. Limit the output of the mix intended for MP3 to -1 dB, instead of the commonly used -0.1 or -0.2 dB.

# Bounce for Mastering In Logic Pro X

Once your mix is ready, you need to bounce it to a single stereo track for mastering.

## Bounce The Current Project To An Audio File

**1.** In the Logic Pro Tracks area or the Mixer, make sure that the tracks you want to include in the bounce are routed to the main output (Output 1-2) and are not muted.

If your project has multiple output channel strips, you can bounce only the tracks routed to a specific output channel strip using the Bnce button on that channel strip. See Use output channel strips in Logic Pro.

**2.** Choose **File > Bounce > Project or Section**.

**3.** In the Bounce dialog, select one or more destination formats in the Destination area.

**4.** When you select a destination format, bounce options for that format appear to the right of the Destination area. For each selected destination format, choose bounce options.

**5.** To limit the bounce to only part of the project, adjust the Start and End value sliders. You can click the up and down arrows, or click one of the numerals and enter a new value.

If Cycle mode is on when you choose File > Bounce, only the part of the project enclosed by the cycle area is bounced. You can adjust the Start and End value sliders to change what part of the project is bounced. See Set the bounce range in Logic Pro.

**Tip:** To avoid having reverb and other effect tails cut off at the end of the project, set the end position of the bounce a little bit past the end of the last region.

**6.** Set the bounce mode by selecting one of the two Mode buttons:

**Realtime:** Performs the bounce in real time. Use this setting when you want to bounce audio and software instrument tracks, or external MIDI sound sources that are routed to the Mixer via aux channels.

**Offline:** Bouncing offline can be faster than real time for more complex projects, and can perform bounces not possible in real time (because they might exceed the processing power of your computer).

**Note:** Only internal sources (audio or software instrument tracks) can be bounced offline; not MIDI tracks or audio channel inputs. Offline bouncing is available only for output channels of devices using native (Core Audio) audio drivers; not for DSP-based audio hardware (which can be bounced only in real time.)

7. For additional control over effect tails, select either of the following Mode options:

   **Bounce 2nd Cycle Pass:** The bounce process takes two repetitions of the cycle range into account, with the creation of the bounce file starting at the second repetition. This is useful if you want effect tails (from the first cycle pass) to be added to the start of the bounce file.

   **Include Audio Tail:** The bounce file is extended as far as necessary to include any instrument release and effect tail.

   **Note:** Some plug-ins, including plug-ins used for mastering and the test oscillator plug-in, can add noise to the signal. Include Audio Tail should not be selected when using these plug-ins, as the resulting bounce file would be too long.

8. Choose a normalization setting from the Normalize pop-up menu:

   **Off:** No normalization is applied.

   **Overload Protection Only:** Downward normalization takes place only for overloads (levels above 0 dB, which would lead to clipping), but no normalization takes place for lower levels.

   **On:** The project (incoming audio) is scanned for the highest amplitude peak, then the level is increased so that the peak is at the maximum possible level (without clipping).

9. To add PCM, MP3, or M4A bounce files to the Project Audio Browser, select Add to Project.

10. Click Bounce.

    Depending on the length and complexity of the project, the bounce process may take a few moments to complete.

# Quick Mastering In Logic Pro X

**Mastering Bit Depth and Sample Rate**

Master in the same bit depth and sample rate as the mix regardless of what the intended end product specs are. If the end product is of a lower resolution, convert it down after the mastering.

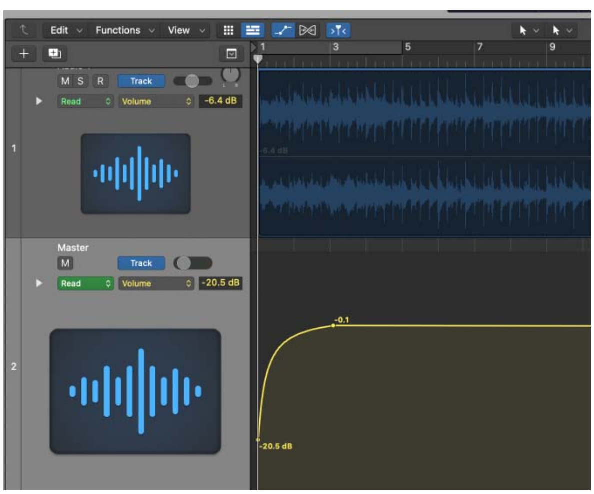
**Remove DC Offset**

DC Offset is when the waveform is not aligned or centered on the DC Line. This needs to be done for many reasons. If you apply a limiter and there is DC Offset, one side may end up being cut off.

Select the audio track, then hit Cmd-6 which will open up the audio editor. The audio should be selected, but if not click Cmd-A to select the audio. In the menu of the audio editor,
Click **functions > Remove DC Offset**.
A window will open up and show the amount of offset. Click OK to remove.

**Add Fades**

Put a master fader in the arrange area. In the Mixer, on the Master track, right-click to open the menu and select Create Track. The purpose of this is so fades can be done via Master last. In arrange view, select expand the size of the Master Track. Hit the letter "**A**" to open up the automation mode and select "**Volume**" as the parameter to automate. Using the **Automation Curve Tool**, put a Logarithmic Fade-In on the start of the track and an Exponential Fade-Out on the back of the track.





**Check Mono Compatibility**

**Plugin > Imaging > Direction Mixer**

We can temporarily convert the mix to mono by using a Directional Mixer. Make the stereo spread zero to convert the track to mono. Anything that is out of phase will show up in mono.

**Multimeter Plugin**

**Plugin > Metering > Multimeter**

Use this plugin to monitor how much the mix changes from input to output from mastering.


**Exciter Plugin**

**Plugin > Specialized > Exciter**

An aural exciter can boost the harmonics above a certain fundamental. This plugin works well with guitars in mixing. Boosting harmonics can give the track presence. Start at around 2k and adjust from there.

**Multipressor**

**Plugins > Dynamics > Multipressor**

This plugin splits the frequency into four bands that you can adjust. It allows you to boost or cut a frequency range, but also allows you to apply compression to the bands.


**Stereo Spread**

**Plugins > Imaging > Stereo Spread**

Only apply to the higher frequencies. Pull down the intensity in the lower range a lot. Pull down the intensity in the higher range a little. Increase the order to create a higher number of bands (denser)

**Adaptive Limiter**

**Plugins > Dynamics > Adaptive Limiter**

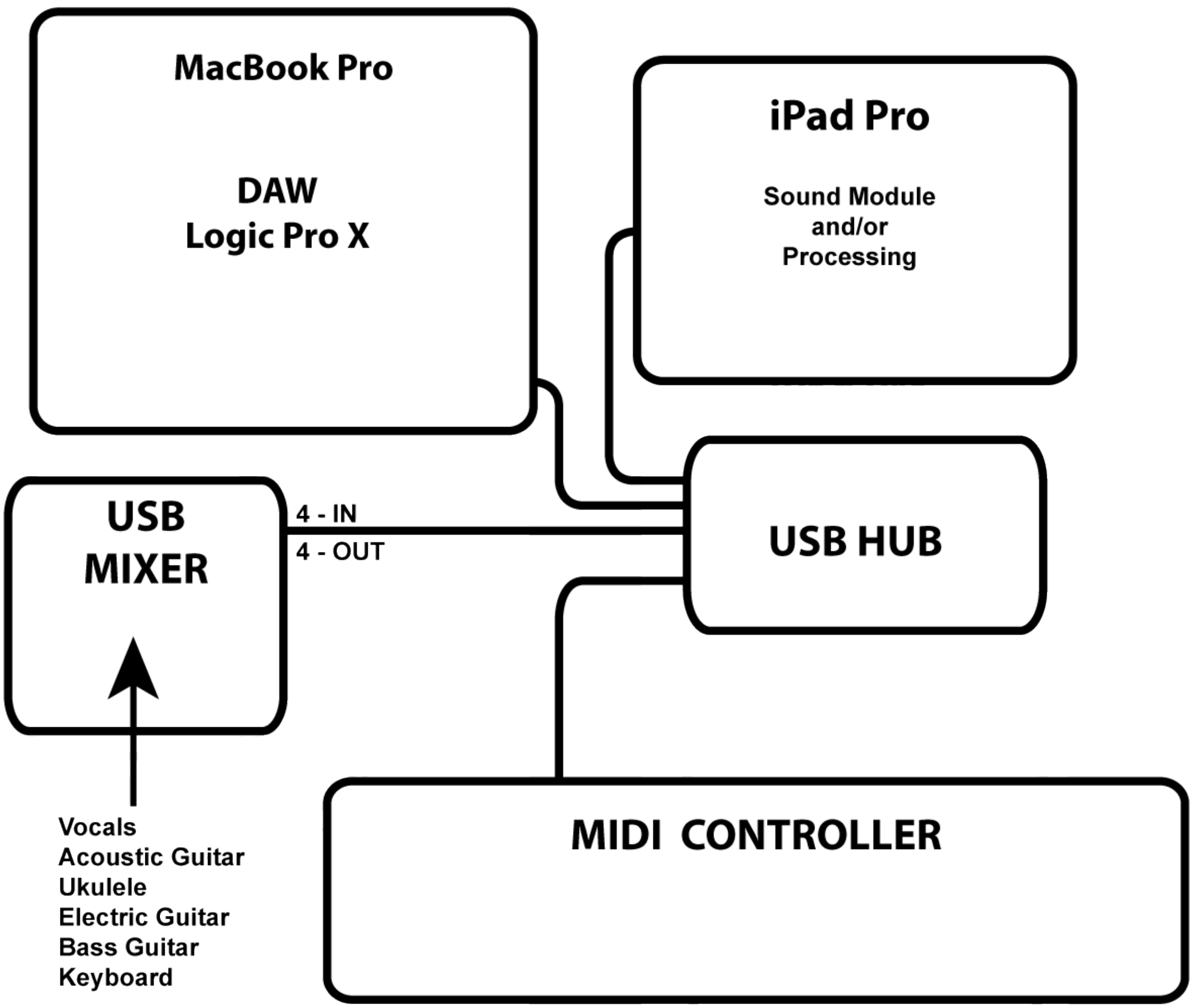
A master limiter. Start with an Output Ceiling of -0.1 dB and adjust the Gain. Pay attention to the Output Meter.


**Bounce Mastered Mix**

The final step is to bounce the mastered mix to an uncompressed file format such as PCM or AIFF.

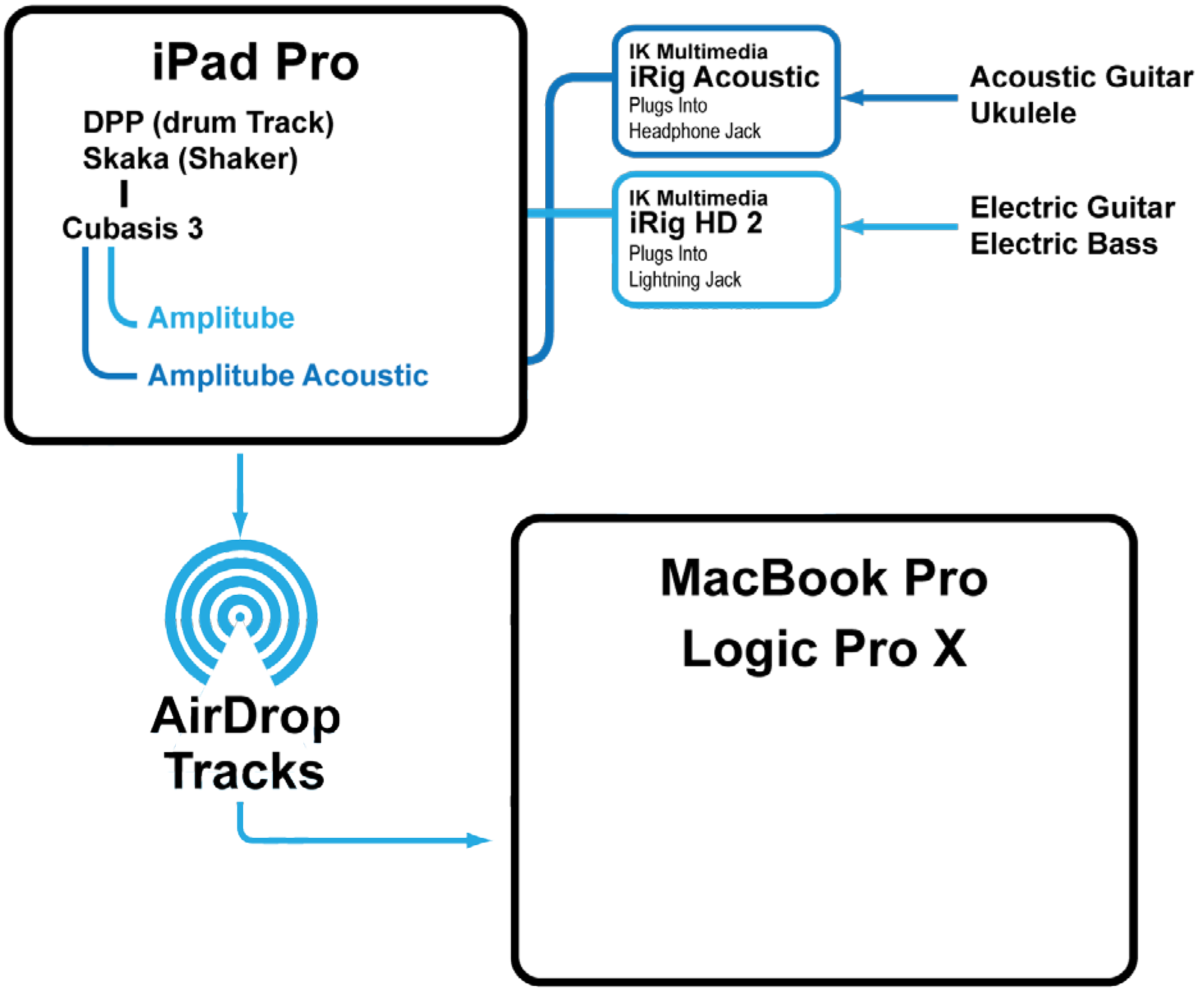
# Apple Recording Studio

Years ago with my first iPad, I purchased an app that was a sound module for Pianos. The only way to use it was to connect the iPad to a MIDI keyboard via the lightning port on the iPad to a USB out plug. Then to record it, I had to connect the headphones out jack to an adapter, to an audio interface and into a computer DAW. In essence I was going digital-analog-digital. Now, with IDAM (Inter Device Audio + MIDI), I can connect my iPad to my Macbook and have access to the sounds on the iPad and the transfer is digital to digital. With a MIDI controller hooked up, I have access to all of the sounds in LPX and on my iPad.




The great thing about this studio is that, besides the MIDI controller, the entire thing would fit in a backpack.

# Using the iPad for Development



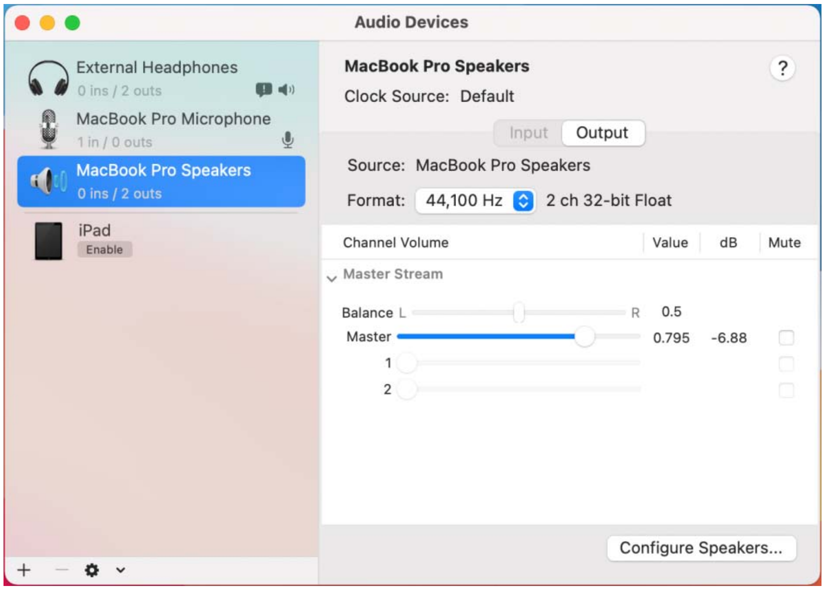

In this configuration much of the initial work is done on the iPad, then the tracks are ported over to Logic Pro X on the MacBook Pro to be mixed, have vocals added, add more tracks, and to polish it up.

Using an app like AUM and Cubasis combined with other apps, you can create some unique combinations that would be much more difficult to produce on the MacBook in LPX.

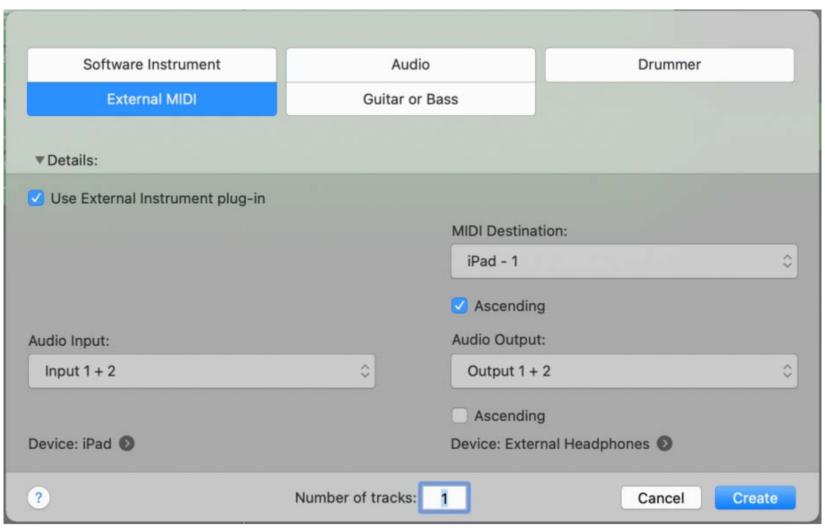
# Using iPad Sound Modules in LPX

Connect iPad to Macbook Pro using the correct adapter. Current iPads have Lightning ports and MacBook Pros have USB C ports.

After they have been connected, in **Audio MIDI Settings**, click on **Enable** button for **iPad**.




In Logic Pro X, Create a new **External MIDI** track




Check off the **Use External Instrument plug-in**

Set **Audio Input** to **Input 1+2**

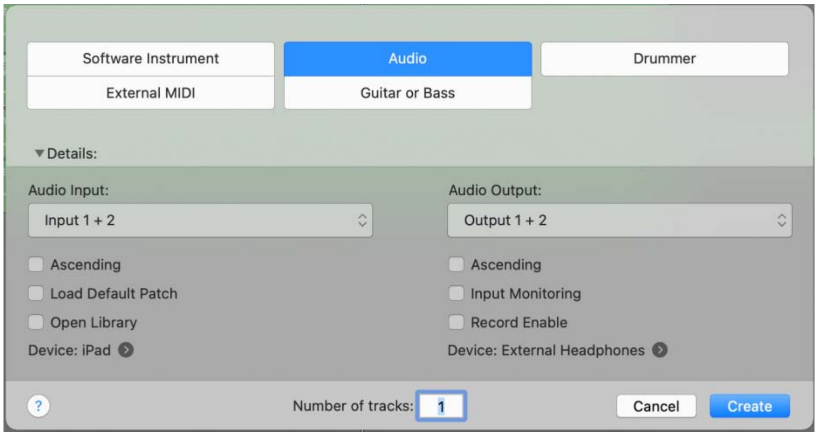
Set **MIDI Destination** to **iPad - 1**

Set **Audio Output** to **Output 1+2**

At this point it will readily record MIDI and the MIDI recorded will play back the sound module on the iPad, but it will not yet be able to record the sound.

## To Record The Sound Module Audio

Create an **Audio** Track

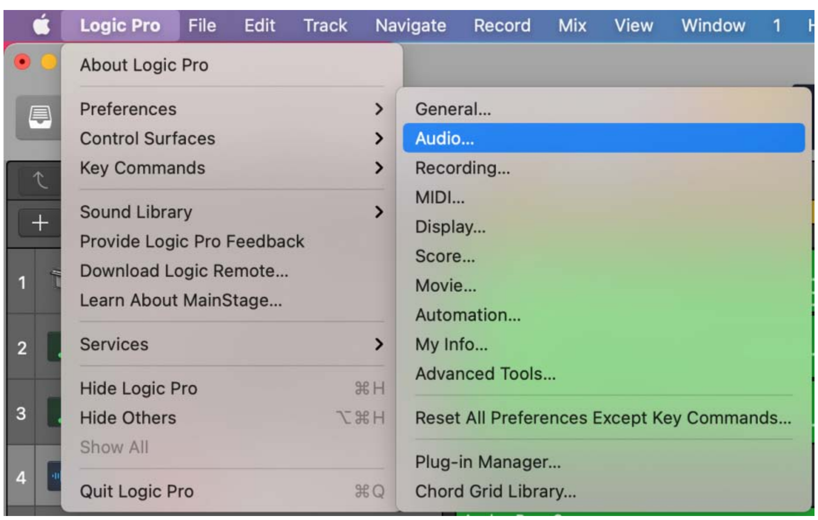



Select **Audio Input** of **Input 1+2**

Select **Audio Output** of **Output 1+2**

Open **Audio Preferences**

Set the **input device** to **iPad**.




Arm the **Record** on both the **Audio** and **MIDI** tracks

Select the **Audio Track** and click on the upper **Record** button

This will record the audio generated by the iPad onto the Audio track in LPX.

You can also use pre-recorded MIDI as the source to create an Audio track.

# Work Process on iPad

## Creating a Foundation

I like to start out creating a basic rhythm track. To do this I use the app Drum Perfect Pro. DPP allows the user to choose from a library of rhythms or they can create their own. The different rhythms can then be sequenced together into a song.

## Add Rhythm

On top of the drum track, I might add a rhythm track of a shaker to give the song some drive.

There are several options for pulling the drum song in DPP into Cubasis.

**1** Add an audio track to Cubasis and set its input as Inter-App Audio and choose DPP (all instruments), then arm and record the track. This will record the audio of the drum song onto the track in Cubasis.

**2** Use the export feature in DPP to export the track to Files, AudioShare, DropBox, Google Drive, Email and more. The exported files can then be imported into Cubasis.

Note that you'll most likely, at some point, create separate tracks of the drum parts to be mixed in the main DAW. If you're just using the tracks to keep time in Cubasis and plan on elaborating on the basic beat, option one might be fine to get things started.

Once a foundation is created it makes it easier to create the supporting tracks.

# Music Repositories

A music repository is just a cloud based file repository or drive. There are many options for this. The most popular being DropBox and Google Drive.

The great feature of these online repositories besides protecting your data from the crash of a system is that they make cross platform development much easier. Another great feature is that you can reuse data or use it as a basis to build upon.