Thank you for purchasing

Pro Techniques for Home Recording

This book will serve as a guide to making your home or project studio recordings sound as good as they can be.

At this point, you will be able to download a small batch of audio examples demonstrating some of the microphones, microphone techniques, preamps and converters, processing hints and mixing/mastering tips. *These files will be updated and expanded – so keep checking the site!* **You may download all the files in four batches of 20MB .zip files, or you may save them as you listen to them for further reference**. Just right click on the link and "save target as" in a folder on your hard drive for future use and reference. Click here: BATCH1 BATCH2 BATCH3 BATCH4 There are many freeware unzip utilities available on the web.

Always keep in mind that some of the greatest recordings of all time were made using equipment that had a fraction of the capabilities of today's recording gear. It is entirely possible to make a recording that sounds as good as (or better than!) a chart topping hit on a small digital recorder, a laptop, or a personal computer. Even a four-track cassette recorder can create excellent quality recordings if used properly.

It will take a little time and practice to develop your critical ear. This is the same when working in any media – noticing light and shading differences in photography, brush strokes in oil paintings, spices when cooking. I have provided some audio examples where the differences are obvious, others where they are very subtle. The important thing is to listen carefully and critically; then you will develop a sense of hearing which will serve your creative goals.

We will cover recording techniques from the barest of basics, through mic choice and placement, and all the way to mastering your masterpiece. Audio examples will be provided along the way so that you may hear what these ideas can do for you. Here is a list of the topics covered:

By clicking on any of the topics in this list, you will jump to that topic. Additionally, for those of you online with the computer connected to your studio monitors, the examples ***(EX#)*** are hyper-linked - just click on them and the example will open up from the web and start playing! Save them by right clicking and "save target as" to a folder on your hard drive.

After reading and listening to this material you will be ready to get your gear fired up and get to the good part – making and recording music. Take your time to get familiar with these techniques and spend some time trying them out. Get a musician friend to be your guinea pig so you can twist knobs and move microphones while someone plays.

There is no single technique that makes a recording sound fantastic; a great recording is the culmination of work on many small details, which add up to a great final result. Keep this in mind and you will create high quality recordings.

An old saying in the industry is "garbage in, garbage out", meaning that even the best audio engineering cannot rescue a lousy performance! It also means that the engineer must make the best possible recording with the available gear, because a great performance that is poorly recorded is also a disaster!

So, let's get to it.

Important terms, good gear, and bad myths

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Important terms...

The very first thing we must do is to get you familiar with the *lingua franca* of the recording world. You don't need to read straight through this chapter, but here are the basics you can refer to while reading. The amount of unfamiliar vocabulary can be daunting at first, but you will get used to it as you go:

¼" cable: The diameter of the connector of very commonly used audio cable, such as a guitar cable.

A/D: analog to digital. Refers to the process or device that converts an analog audio signal into digital information.

AES/EBU: A format of digital audio interface developed by the **A**udio **E**ngineering **S**ociety and the **E**uropean **B**roadcasting **U**nion. This format is transmitted on an XLR (3 pin) connector and is a standard for professional audio equipment.

amp: short for amplifier.

amplify: to increase the level of. Example: A guitar amp *increases the level of* an electric guitar's signal, making it strong enough to drive the speakers.

analog: from the word 'analogous', meaning 'the same as'. An analog signal is the same as it's source, only different in level and its resolution is theoretically very high, much more detailed than a digital representation of the signal.

aux send or auxiliary send: a signal path through which a copy of a channel's signal can be sent without affecting the output of that channel. An aux send is typically used to feed a copy of the signal to a reverb unit, and then the reverb unit's outputs are returned to the mixer via a stereo channel (or two mono channels). Some mixers have a dedicated 'aux return' – simply an input channel with no frills like EQ or compression.

bit: In digital audio we often hear of 16 bit or 24 bit audio. These are the number of possible steps of resolution. 16 bit audio has 2 to the 16th (or 2x2x2x2x2x2x2x2x2x2x2x2x2x2x2x2 = 65,536) steps, 24 bit audio has 16777216 – quite an increase in resolution!

bus: a path for a signal in a mixer. The main outputs can be called the 'main bus'. Some mixers have multiple busses that can then be sent to the main output. For instance, you can send 8 different cymbal and tom tracks to a stereo bus, so that you only need to adjust the bus fader to change the level of the whole mix of toms and cymbals. Busses also offer insert points for adding compression or EQ to bussed signals. This is very useful when mixing or processing tracks.

brighten: to increase the high frequency content of a signal with EQ.

click: studio slang for the sound of a metronome. The 'click track' serves as a reference for tempo to all musicians.

Class A: A design for audio amplifiers in which both halves of the signal (both positive and negative voltage values) are amplified simultaneously. That is to say that 100% of the input signal is used in creating the output signal. The drawback to Class A amplifiers is their relative inefficiency. Approximately half of the power drawn to operate the circuit is wasted – dissipated as heat. This makes the use of class A circuit design practical only for low-level amplification, such as a microphone preamplifier. Heat build-up can, however, still be a consideration and class A amplifiers should have ample airflow around them for proper cooling.

comp: Short for 'compile' *not* to be used as short for 'compression', which some people get confused. For example to 'comp a vocal track' means to compile the best takes of a singers performances into one track by editing.

compress: to reduce the level of an audio signal when it exceeds a defined threshold, thereby reducing the dynamic range of the signal. This can be done manually by pulling down the volume fader as the volume peaks, or by using a **compressor;** a device built to do this automatically.

D/A: digital to analog. The *yang* to the A/D *yin*! In order to listen to all your digitally stored sound.

DAW: Digital Audio Workstation. Usually a computer with a sound card (or interface like an Mbox) paired with software for editing and processing audio. Some companies, such as Fairlight, manufacture their own systems and run on their own host software platform (not Mac or Windows).

dBFS: without getting unnecessarily technical, this is how a digital meter functions. We can see the signal level as well as short peaks in the audio signal, which is very important for recording digital audio. Digital audio devices 'clip' (distort harshly with a pop, click, or harsh scratchiness) when the signal goes over the maximum allowable level (0dBFS). An analog device (like a tape machine) can handle mild overloads without harsh distortion, so monitoring level with dBVU meters is preferred in that case.

dBVU: the dBVU meter is the cool looking 'swinging needle' meter. Unlike dBFS meters, dBVU meters display more of an average level of the signal. The 'swinging needle' requires time to move, so the reaction to signals is very smooth, This smooth motion is very practical for audio metering because it reacts as our ears do. We perceive sounds with longer duration as louder. A gunshot in an acoustically 'dead' room sounds much quieter than one in an echo chamber. The peak volume is the same in both cases, but the duration of the sound is different; so to our ears, peak volume is relatively less important.

decay: the 'hold' of a sound. For instance, the echoing sound of a drum hit in a concert hall has a longer decay than a drum hit in a small room.

decibel: a unit of measurement for sound level, meaning 'one tenth of a Bel' (named after Alexander Graham Bell). Since dB is a logarithmic scaled measurement, 6dB higher is double the value.

digital audio clock: digital audio consists of multiple samples (see 'sample' below) and these samples must be taken at a steady rate. Any device dealing with digital audio needs a stable way to take these samples. Different devices also need to be synchronized, usually by allowing one device to be the *master* and other devices to be *slaves* (to follow the master clock). See also 'jitter'.

distortion: not just what guitarists think of, such as the pleasing sound of an overdriven amp. Distortion in audio often refers to any sort of change in the sound quality from that which is desired. In this respect, a nice sounding distorted guitar track could be distorted in a not nice-sounding way by, for instance, recording it at a too-high level and causing clicks and buzzes in the digital file!

dither: a random signal (noise) added to 20 bit, 24 bit or longer signals before reducing the resolution to 16 bits. Dithering aids in maintaining signal detail in low-level signals.

dynamic range: the difference between the loudest and softest volume level of an audio source. The piano is a great example; did you know that the piano is really named the ***piano forte*** ? Translated from Italian that makes it the "soft loud"! Imagine the quietest notes you can play on a piano, barely pressing the keys. Now imagine Jerry Lee Lewis hammering those keys so hard that he can be heard over a whole Rock band. This shows the piano's wide range of dynamics.

EQ: short for equalization.

fader: a type of potentiometer set up as a slider for controlling volume. The terms 'fade in' and 'fade out' refer to the volume control this device provides.

Firewire: Also known as IEEE 1394, Firewire provides a method of connecting devices to a computer with a data transfer of up to 800Mbps (megabits per second). It allows connecting or disconnecting devices without the need for a computer restart and is a popular interface for hard drives for digital audio storage. The term "Firewire" was coined after the protocol was invented, because it disconnects so easily while you are busy working, that you will pull it out of your gear like a garden weed and throw it into a fire.

ground loop: a complex topic, but for our purposes: when two pieces of audio equipment are plugged into two different outlets, especially when separated by some distance, a different electrical potential relative to ground can exist. This difference in potential (voltage) causes a signal to flow, generating hum or buzz at the frequency of the supply electricity (60Hz USA, 50 Hz in Europe). Some people remove the ground connection from one of the pieces of equipment, but do NOT do this. **Removing (or 'lifting') the ground is against electrical safety regulations in**

most civilized societies and is potentially very dangerous. See the section on 'Setting up your studio' for safe solutions.

impedance: a circuit's opposition to an alternating current signal, such as an audio signal. When interfacing audio devices, it is important to properly match impedances to avoid loss or distortion of the signal. A transformer D.I. offers a practical way to match impedances of, for instance, the pickups of an electric guitar and the input of a mixing console circuit.

jack: the socket into which an audio cable is inserted.

jitter: undesirable variations in the digital audio clock. These variations can lead to unpleasant sounding distortion and loss of definition and clarity in the audio signal. Sound cards built into a computer are often plagued by jitter problems, with detrimental effects on the sound quality.

latency: The time lag between the input signal and the output signal incurred by computer processing. Latency occurs in many links in the audio chain. Usually measured in *samples or milliseconds*.

line level: the operating voltage level (usually about .5 – 1.5 V) for signals in most professional audio devices.

mic or mike: short for microphone. Mic is the preferred spelling, for obvious reasons.

mic level: the relatively low level of signal, as low as 1mVolt (.001 Volts) which needs to be increased to *line level* by a preamplifier to be useful by other audio devices.

meter: a device that represents sound levels visually.

MIDI: Musical Instrument Digital Interface.

mix: to bring two or more audio signals together into one. Often used to refer to the process of balancing all the tracks in volume and tone, adding effects, etc when preparing the song for mastering.

monitor: to listen to a signal with speakers or headphones. It is important to understand *where* in the signal chain you are monitoring the signal. *Input monitoring* lets you hear the signal going to tape (or disk), *output monitoring* is listening to the signal coming from tape (disc). Output signals may sound different because they are directed through further effects devices. When recording on a computer, monitoring the output can cause increased latency.

MTC: Midi Time Code. A reference signal which carries bar and beat information and allows MIDI capable devices to synchronize.

muddy: A common description of an audio signal meaning that it sounds unclear or dull. There are many possible causes for this including poor room acoustics, overuse of reverb, and too much bass or low-midrange frequency content.

normalize: a digital audio process which raises the level of the audio file so that the loudest portion is at 0 dBFS. Not something you should do until mastering.

overdub: adding more recorded tracks after the main recording is finished. First developed by Les Paul (of Gibson guitar fame), he called it 'sound on sound', and it made possible the addition of vocals or instruments to a pre-recorded song.

It is fair to say that far more time is spent overdubbing in the modern recording studio than anything else.

For example, you record a drummer, guitar player, bassist, and a singer playing a song. The singer is unhappy with his performance, the guitar player played just one wrong chord and the others feel that the take was fine.

So the singer overdubs a new vocal and some harmonies. The guitar player 'punches in' the right chord, then overdubs a second track on acoustic, a solo and some more electric guitar fills. The drummer overdubs a tambourine and shakers. A keyboard player is brought in to overdubs some string sounds... and on and on.

patch: 1) As a verb, patch means to 'connect', referring to the patch bay – a collection of connectors which brings all your audio connections to one location to make it easier to interconnect gear.

2) A program on a synthesizer. Example: "That Yamaha keyboard has a great string patch."

phantom power: 48v power supplied to a microphone through the mic cable. Necessary only for condenser microphones. **Caution: Phantom power must NOT be used with ribbon microphones** as this can destroy the ribbon!

phones: short for headphones.

pickup pattern: see 'polar response'

polar response: a way of describing the way a microphone responds to sound in terms of directional sensitivity. The four major patterns are cardioid, hypercardioid, omnidirectional, and figure eight. See the section on "Microphone choice and placement for graphic representations of these characteristics.

A cardioid polar response means that the microphone picks up sound in front of it in with highest sensitivity, less at 90 degrees to it's side and very little from behind.

Hypercardioid response is more dramatic than cardioid; the response is already very low at 90 degrees from the front and is negligible from behind.

Omnidirectional microphones are designed to pick up sound evenly regardless of what direction it arrives from.

Figure eight patterns pick up sounds from the front and back, but little from the sides.

pot: Short for 'potentiometer'. A potentiometer is a variable resistor which is commonly known as the 'volume know' on a guitar, and finds use in many pieces of studio gear.

preamp or just **'pre':** the device that raises a signal up to the proper level for recording to tape or disk. The way in which a preamplifier amplified the signal has a major influence on the sound of your recordings. Class A amplifiers use both the positive and negative voltages of the signal at once to create the output signal, remaining perfectly true to the input signal. All other designs impart some change to the signal. Class A amplifiers are quite inefficient in their use of power, but are preferred for low level amplification due to their accuracy.

punch in: to record over a portion of an already recorded track. Example: "I made a mistake in the chorus, could you punch me in there?" So, as an engineer, you should wait until just before the chorus, then, on the first downbeat, slug the singer in the solar plexus. The drummer will now be your best friend.

When *destructively recording*, you permanently overwrite the portion recorded over, as with analog tape. *Non-destructive recording* is a nice development in digital recording, making it possible to get back the portion of the track recorded over after punching in! The only drawback to non-destructive recording is the additional hard drive space required to store both the new recording as well as the old portion recorded over. It can also be more CPU intensive on your computer.

resistor: An electronic device which resists or 'stands in the way' of electricity.

reverb: short for 'reverberation'. Reverb is often produced digitally and simulates the acoustic properties of rooms, halls or churches, for example. A *plate reverb* is a large metal plate (or a digital simulation thereof) to which a copy of the audio signal is applied, causing long lasting vibrations in the plate. These pleasant sounding after-vibrations are picked up by a microphone and mixed back with the original signal, adding decay to the sound.

sample: A small portion of information stored digitally. There are several ways this is used. 1) You may hear reference to 'a drum sample', in this case it is often a single drum hit recorded clearly and used to create a drum kit artificially (in a sequencer or other program). 2) When referring to **sample rate**, such as 44.1k or 96k, the sample is the smallest unit of measurement of the digitized audio. 44.1k means that the incoming audio signal was measured and recorded 44,100 times per second!

scratch: studio slang for a rough take of a part, helpful in remembering ideas or representing how a final, polished take will fit into the song. Example: "Hey, turn up that scratch vocal track, I forgot the words and need to hear what he sang there... Oh, O.K. I hear it now: he wasn't singing, just taking another hit on his crack pipe."

sequence: in the studio, this most commonly refers to the recording of a MIDI signal. A 'sequencer' is a way of referring to a MIDI recorder.

SP/DIF: Sony-Philips Digital Interface. A digital format transmitted on a 2-conductor RCA cable. It was designed as more of a consumer interface, but is found on many professional units as well.

sync: short for 'synchronize'. Can refer to a *signal* designed to synchronize the operation or functions of devices, such as MTC.

squash: studio slang for 'to compress heavily'.

take: studio slang for one recording of a performance. Example: "That vocal was out of tune, let me try another take." Mmm Hmm, that'll work.

timbre: pronounced "tamber". The classical music world word for 'sound quality' or 'sound texture'.

track: one separately recordable channel for audio. Example: "The guitars are on tracks one and two, the vocal on track three, and percussion on track four."

transient: Often used in describing the time or portion of an audio signal where the signal changes rapidly. For instance "the 'transient edge' of a percussion hit" – this would be referring to the detailed sound of the impact of that sound. Transients contain much fine detail, such as the pluck of a string or the hammers of a piano. The clarity of this portion of the signal has a major influence on our subjective evaluation of the quality of a sound.

TRS: Tip Ring Sleeve. A style of ¼" cable connector that allows for balanced mono signal or unbalanced stereo signal transmission.

truncate: to cut off. To send a 24 bit signal to a device that can only handle 16 bits will cause the receiving 16 bit device to simple ignore (truncate) 8 bits of resolution.

USB: Stands for Universal Serial Bus and allows for connecting/disconnecting devices from the computer without the need for a restart. It is a popular method for connecting audio interfaces to computers. USB 1.1 offers data transfer rates of up to 12Mbps, and USB 2.0 up to 480Mbps (in theory)

VCA: Voltage Controlled Amplifier. A device that can be used to compress a signal.

word clock: The signal for synchronizing a device to an incoming digital signal is usually carried within the incoming signal, but the receiving device may also be able to receive a sync signal to a 'clock input'. This allows you to connect one 'master clock' to all digital devices in the audio chain, creating a far more stable synchronization than several devices running at their own individual clock rates. Word clock is this master signal.

word length: 16 *bit*, 24 *bit.* How many *bits* in your signal is the word length. In processing digital audio (which is simply mathematical calculations), the word length can change, becoming quite long. The internal word length capabilities of a digital mixer, processor, etc. should be able to pass along these longer calculations without cutting off numbers and ruining your sound. Ever read of "32 bit internal architecture"? This is what it is referring to. A digital mixer that only outputs a 16 bit signal can seriously degrade your sound!

Good gear...

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No matter how modest your budget for recording equipment, you can get a decent sound happening. The important thing to know is what you ***need***, not just what you want ($$$!!!).

Some of the most common mistakes made when purchasing gear (and I, unfortunately, have learned by doing) are:

- **Buying much more than you need:** If you are only planning on making songwriting demos, you probably don't need a 24 track fully automated digital recorder or a Pro Tools HD rig. You will be much better off with a more portable and easily operated 8 track digital recorder, which you can pack in a suitcase and take on the road. If you just want to catalogue some ideas on an acoustic guitar while singing, you may just want a portable MP3 recorder which will run for hours on a battery while sitting in the park. Seriously consider your needs, and be realistic.

- **Buying the wrong piece of gear for the job** - Example: A Shure SM-57 ($79) can sound better on a guitar amp than a Neumann TLM 103 ($800). If you're doing a lot of electric guitar work, the former is a better choice.
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  Another example: It is much better to invest in a single high quality reverb that does the job simply and well than to get the latest all-in-one pitch shifter and flatulating flanger-from-hell "just so you have the options".

  First of all, you will very rarely use the more far-out effects, and the reverb quality in an all-in-one box is not going to have the quality of a dedicated unit.

  Second, we are going to look at how you can cook up some unique effects by using the more basic building blocks in the "effect processing and design" chapter; there you will learn how to cook up effects which are unique. A unique effect (as opposed to one which everyone and their cousin has in that all-in-one box) can really make your song stand out.

- **Misunderstanding the need for equipment:** My favorite example is the noise gate. If you record your tracks cleanly, a noise gate will be unnecessary. We will learn how you can work to make your studio quiet enough so that there will be little or no noise to gate out.

Additionally, you are also better off investing in a pair of excellent microphone preamplifiers than buying (for the same price) a set of eight lower quality preamps. In the long run, your recordings will benefit most from the high audio quality you get while overdubbing with the two.

Let us start looking for gear that will be useful with this concept in mind: *Sound quality is both objective and subjective*.

Objectively, a recording of a voice or instrument that sounds just like the original source when played back can be called a 'good' recording.

Subjectively, a strange, altered or distorted sound which is no longer very similar to it's original sound source can be 'good' – if it fits into context and if you like it!
Furthermore, gear which affects sound in a way that you have heard before and you have liked, sounds 'good' to you.

With these concepts (a bit of musical psychology) in mind, let's look at a list of some gear that can be found in any pro studio worth it's salt. I will obviously skip the $100,000.00 consoles and $25,000.00 digital workstations. We want to know ***what is the standard gear*** that a home/ project studio can fit also into its budget. **This will give our recordings a familiar air and a solid tone that even the non-musician/engineer will find to be 'good' sounding; and that is key to good recording technique.**

Microphones

On the front line of every recording is the microphone. It is the first step in capturing a sound, and the proper choice of microphone is crucial to a good recording.

There are four major types of microphone; dynamic, condenser, ribbon and crystal.

These types all exhibit unique characteristics and have typical uses. We will get into mic choice and placement in a later chapter, but here is a list of standard mics for you to consider when shopping around. It is probably a good idea to have, at the very least, one dynamic and one large diaphragm condenser microphone.

Dynamic microphones

Dynamic microphones function like a miniature speaker – in reverse. A coil of wire is fixed to a small cone which, when sound vibrates the cone, moves in a magnetic field and causes a signal to flow in the wire leads.

Dynamic mics are rugged and dependable and require no external power (phantom power) to function. This makes them a great choice for loud sound sources such as drums, percussion, electric guitars, organs, and bass amps. The only downside is that they can lack high frequency detail on more delicate sources like acoustic guitars, strings or piano. For these sources you should consider using a condenser microphone.

- The ***Shure SM-57*** ($80) is a required microphone for every studio. This should be your first purchase. It has been said that you will never get a fantastic sound with this mic, but the track will always sound good. Most pro studios have 3 or 4 lying about, wired and ready to go.

- The ***Shure SM-58*** ($90) is another standard mic, found in every studio. It is geared towards vocals, and will always give you a good sound. It may lack some of the detail of high priced condenser mics, but many engineers agree that it has a sound which just 'fits nicely into a track'. Here's a snippet of male vocal. Compare it to EX 4a and 4b, which are done on pricier condenser mics! ***(EX1)***

- The ***Sennheiser MD-421*** ($350) is a slightly more expensive mic, which is excellent for drums as well as loud amplifiers. Listen to the difference in sound between percussion and a guitar amp mic'ed with a Shure 57 (first) and the Sennheiser MD-421 (second) ***(.wav file EX2)***. Some people describe the sound of the 421 as 'bigger and fatter' than the '57. It makes a nice alternative to endless Shure 57 tracks.

- The ***Electrovoice RE-20*** ($400) is a well loved dynamic microphone for bass drum and bass amplifiers. Its sound reaches down a little lower into those nice bass frequencies.

- The ***AKG D-12*** ($200) provides a slightly more affordable alternative to the RE-20 for use in bass drums. It also has a little more high end 'click' to it, which is desirable in some styles of music.

Condenser microphones

Condenser microphones contain a very thin metallic membrane fixed to a metal back plate. These elements are electrically charges, and as sound pressure changes cause motion in the thin, sensitive membrane, a signal flows to the microphone output. This requires the application of 48V ‘phantom power’ to operate.

Because of the high sensitivity of the small metallic membrane, condenser mics pick up far more detail in a signal than most dynamic mics, making them ideal for sources with detailed high frequency response such as vocals, acoustic guitar, piano, high pitched percussion, and cymbals. They can, however, overload easily in high volume situations, causing unpleasant cracking and popping noises.

The most common pickup pattern (also called ‘polar response’, see definitions) in microphones is cardioid, but many condenser microphones are available omnidirectional or even switchable.

Condenser mikes are generally available with ½” or 1” diaphragms. The ½” diaphragm mikes look somewhat like a large pencil (sometimes referred to as ‘pencil mikes’). The very small, light diaphragm is highly responsive to sound sources with ample high frequency content, such as high hat and cymbals on a drum kit. The 1” diaphragm is a bit slower on the draw, but handles lower frequencies better, making it the standard choice for vocal recording.

Small (1/2”) diaphragm

- The ***Shure SM-81*** ($350) is an excellent deal in terms of price/performance and is used very often on acoustic guitar and drum cymbals. A studio standard.
- The ***Neumann KM-84*** ($700) is not quite such a bargain at double the price (you could be working in stereo with two Shure 81’s...), but it has that Neumann sound – very detailed and somehow larger than life.
- **AKG** provides a range of small diaphragm condensers from the **C1000s** ($150) to the **C451** ($500). The 451 holds it’s own with the Shure and Neumann above. The C1000 is probably better suited to live sound reinforcement, it just doesn’t have the smoothness and detail of big brother 451!

Large (1”) diaphragm

It is possible to spend many thousands of dollars on a large diaphragm condenser microphone, and many pro studios keep heavyweight classic mics such as the Neumann U47, U67, M49, the newer M149, the AKG C12, or boutique mikes by Brauner, Manley, Lawson and other small production/high quality microphone makers.

The truth is, most home studio enthusiasts don’t have the extra greenbacks for big $$$ mikes. So what are we to do for a great vocal sound? Here are a few under $1000.00 (still a lot of dough, but better than $6000.00!):

- A favorite of mine is the ***Audio Technica AT 4030*** ($300). Is a great cardioid-only mic that they don’t seem to be making anymore, so keep an eye out on eBay. They sound great on rock vocals - like a Shure 58 with more detail. You can get an AT4050 for about $500, have the option of selecting polar patterns, and they sound quite good as well.
- The ***AKG C3000*** ($150-200) is often offered to newbie recordists, touted as a great vocal mic, etc., but I have yet to hear a great sound from it, and they just ain’t makin’ it into the pro studios.
- The ***AKG C-414*** ($800) is another story. It sounds excellent for the price and is very flexible; a studio standard for voice, strings, percussion and piano.
- The ***Neumann TLM 103*** ($950) offers the small studio a taste of the big Neumann sound. It is cardioid only (which saves you quite a few bucks) and has a very detailed, clear sound on vocals, acoustic guitars, and as a drum room mic. Listen to how the AKG 414 (first) compares to the Neumann 103 (second) on acoustic guitar ***(.wav file EX3)***.
- Listen now to an ***MXL V69*** ($299) ***(.wav file EX4a)***. on the same exact vocal track as a Neumann TLM 103 ***(.wav file EX4b)***. It was mic’ed with both the Neumann 103 and the MXL V69 simultaneously. MXL has been manufacturing relatively inexpensive mics of good quality. Though not a studio standard by any means, the MXL mics are a low cost way to reinforce your studio mic cabinet.

Ribbon Microphones

Ribbon microphones are very delicately constructed – a few micron's thickness of metal ribbon is suspended between the poles of a microphone, and the motion of the air moves the ribbon in the field, creating a very low level signal.

The sound produced by a ribbon microphone is warm, rich, and 'classic' sounding. Ribbon mikes are well known for their fine sound on vocals, room ambience, acoustic instruments and, surprisingly – electric guitar amps! They are the secret weapons behind some of the greatest guitar tones recorded! We'll hear some examples later.

The drawbacks to a ribbon mic are its susceptibility to damage by wind and it's low output. Wind or bursts of air (as in the 'P' or 'F' sound sung by a vocalist) can tear the ribbon, ruining the mic. A wind screen or pop filter is a good tool to have around when using these delicate mikes.

You must also have a preamplifier that can provide enough gain to amplify the low signal produced by the ribbon mic to an adequate level without adding noise. For instance, the preamps on my Pro Tools MBOX 2 don't cut the mustard on my ribbon mic. But if you're far enough along in your gear hounding to add a specialty mic to your arsenal, then you should already have a fine mic preamp.

As with all microphones, prices vary and you get what you pay for. Lower cost ribbon mics tend to have less detail and clarity than the big boys. But at 1/5th of the price, can you really go wrong?

If you want to go cheap, try an ***Apex 210*** or a ***Nady RSM2*** (both around $250) there is also a German maker called 'T-Bone' making cheap ribbon mics. They will give you a taste, and may whet your appetite for more. I have heard tale that you may remove some internal windscreens (which the manufacturers put in place to prevent damage by newbie recordists) and replace the internal transformers with better models, but I still have no first hand experience doing this.

Oktava makes the ***ML-52/53*** and ***Electro-Harmonix*** licenses the same mic as the ***EH-R1*** (both about $350). Listen to the ML-53 (first) in action on acoustic guitar as compared to the AKG C-414 (second) ***(.wav file EX5).*** Notice the 'classic' sound of the ribbon mic and fullness to the low notes. Now listen to the ML-53 (first) on a guitar amp as compared to a Shure SM-57 (second) ***(.wav file EX6a)*** and ***.wav file EX6b***). Getting to know these differences is what will make you a good recordist!

On the upper end of the scale, The ***AEA R92*** ($850), the ***Coles 4038*** ($1300), and the coveted ***Royer*** line of ribbon mikes ($1200 and way on up) provide you with top notch electronics and craftsmanship which you can bank on to deliver world class tone. Listen to this recording of acoustic guitar done with Royer ribbon mikes ***(.wav file EX7)***. Wow!

Crystal Microphones

Crystal microphones function according to the ***piezoelectric effect,*** which is the same effect as in the pickups placed into the bridge of acoustic guitars.

When certain crystal materials have pressure applied to them, they give off a small current. By attaching a diaphragm to one of these crystals, a microphone is created.

The ***Crown PZM*** mic is probably the most famous example of a crystal mic that is commonly used in recording studios. It is popular for catching drum ambience and has a very particular, clear sound with a strong transient attack.

Microphone Preamplifiers

After the signal leaves the microphone, it is the job of the microphone preamplifier to bring the signal up to a recordable level. Mic pres are everywhere nowadays, and they are all boasting about low noise, low distortion, high gain and 'professional quality' – so who is telling the truth?

Well, they all are, in a way. You can't argue with technical specifications, so what we need to know is - what are the pros using? Well, no one with the responsibility for recording an album to be released by a record label other than 'Watercloset Records' is using Mackie or Behringer mic pres to cut tracks. Don't get me wrong, there are some pro studios using these mixers - to create monitor mixes or handle multiple sound modules. But the path to tape is always something special.

One of the greatest tips for improving your sound is **"Invest in a good preamp!"** You may only need a single preamp if most of your recording work is done one track at a time, and the difference this will make is astounding as the tracks pile up.

The most popular preamplifier in pro studios may be the ***Neve 1073*** ($3500.00 each). Ouch! Well, the good news is that companies such as Vintech, Great River, Dan Alexander Electronics, and Brent Averill Enterprises offer low cost alternatives to the pricey classic Neve units.

The veritable Neve 1073 includes some very high quality EQ circuits which, although they are great sounding and nice to have, are also very expensive. The companies which offer lower cost alternatives often focus just on the preamplifier stage of the 1073 modules, reducing the cost sometimes as low as $1000 a pair.

$1000 a pair still sound like a lot of money? Well, for many of us it is. The good news is that this type of gear, when well cared for, does not lose value. In fact, I sold a set of four ***API 312*** preamplifiers for $400.00 more than I bought them for - just 4 years later! The API 312's are a also a studio classic, high quality Class A preamplifiers with input and output transformers.

There are sort of two basic styles of high quality mic preamplifiers; those that 'color' the signal, and those that aim for a 'pure' sound. One of the most important determining factors is whether or not the design includes transformers on the inputs and/or outputs, and which types of transformers are used.

Transformers act to isolate circuits, match impedances, and even amplify a signal slightly. Transformers can be manufactured to be sonically very transparent, but are sometimes intentionally used to color the tone of the circuit. A transformer can make a sound seem 'bigger' and 'warmer', but generally at the sacrifice of transient detail. A balanced trade-off of a little high frequency detail can provide the coveted 'fat and rich' bass response so often desired for percussion, drums, bass and electric guitars – the so called 'classic sound'.

The pleasant sound and helpful functionality of good audio transformers can drastically increase the price of manufacturing an audio circuit. In a two-channel microphone preamplifier, $300-$400 in parts can be spent on transformers alone! So before you think that expensive preamps are a rip-off, or just some money making scheme, think again – you get what you pay for.

This is a good point to dispel a popular myth: That tubes sound 'warm'. The truth is that there are audio circuits available that use tubes, but still sound very crisp and clear, as well as non-tube designs that sound warm and fuzzy! In fact, driving a vacuum tube in the audio circuit (like many cheap audio toys) can sound downright distorted, which may sound more harsh than warm and fuzzy, and at a great sacrifice to the clarity and detail of a sound. Don't let the tube hype spread by cheap equipment manufacturers infect you! Use your ears first and foremost when evaluating a piece of gear.

On the 'clear and neutral' side of the fence, preamplifiers from ***Benchmark***, ***DACS***, ***Grace Designs***, ***John Hardy***, and ***Millennia Media*** provide the clarity and detail often preferred over the subjective color and 'vibe' of the classic style preamps for recording acoustic and classical music. If you are most often going to be recording instruments such as acoustic guitar, mandolin, dulcimer, piano, violin, cello or choir vocals, you may want to consider a preamplifier which will very clearly and naturally represent the abundant high frequency detail these instruments provide. Natural, acoustic music does not benefit as much from the coloration provided by the 'classic' school of audio preamps as much as rock, blues and pop music. Jazz engineers need to make a choice between a warmer, funkier sound and a natural, uncolored representation of the music and let this choice influence their decision to invest in gear.

My personal choice of microphone preamplifier tends to go in the 'classic' direction. I like the warmth and 'size' provided by a big hunk of transformer iron in the signal path. I find that in the digital age, any way to take the edge off a clinical, digital sound is welcome. And as noted above, just because there is a little vacuum tube glowing away in your audio circuit doesn't mean that your sound is going to be warm and full. The tried and true way to get this sound is by bringing your mic signal up to level with a preamplifier like the ***Vintech 1272***, A ***Brent Averill Neve 1272*** module, an ***API 312, Universal Audio 610, Great River MP1 (or 2) NV, AMI TAB-Funkenwerk V-78***, or one of many offerings from ***Chandler LTD***. There are many other units available, and some great deals on used gear. You may even consider contacting the manufacturer before buying a used piece to ask if they can perform some light maintenance on the unit after you purchase it. Considering that the parts are usually of such high quality, there is little chance that something is seriously wrong, and a little techie TLC will bring the unit up to 'good as new' for introduction into your studio!

For those of us who don't have the immediate re$ource$ to purchase high-end preamplifiers, there are a few options which will provide you with a clear, clean sound which will be very useable at a mild price.

Focusrite, traditionally a manufacturer of high end recording and mastering gear, has in recent years made more offerings in the modest-budget end of the industry. The OctoPre 8-channel preamp ($800) will get you enough preamps to handle a small project without breaking the bank.

The ***FMR Audio RNP*** (really nice preamp) is a popular unit for about half the price of a Neve 1272 style preamp. The designer's idea was to cutback the spending on components he found to be unimportant in the final audio quality to provide a high quality preamp for a fair price. The reviews out there are good. Check out www.harmony-central.com and look in the 'effects database'. You can search for gear reviews by manufacturer. This is a great place to research before you buy because of the averaging effect of many independent reviews. You can also, most importantly, see if any complaints crop up again and again. This can help steer you away from bum gear.

There are other companies offering low cost pre's that I may as well warn you about – ***ART*** and ***Presonus***. I have owned ART mic pres in the past and couldn't get rid of them fast enough. I have no personal experience with Presonus, but have not heard great things from engineer friends, they have spoken of that 'cheap sound' and of harshness… Buyer beware!

Here are a few audio examples for your discerning ear to contemplate:

(EX8 and 9 are coming soon!)

.wav file EX8: Acoustic guitar through the pristine sounding John Hardy preamp (first) versus the 'funkier' Universal Audio 610 (second).

.wav file EX9: Electric guitar recorded through MBox2 preamp (first) and through a Telefunken V676 (second).

.wav file EX10a and b: Bass into MBox2 preamp ***(EX10a)*** and into Demeter HM-1 tube preamp ***(EX10b).***

The final step into the digital realm

Now that your signal has been happily amplified to a proper level to be transferred to tape, you need to convert it into digital ones and zeros to be stored on hard disc.

Putting it that way sounds like this is an easy task that any computer would do in the same way, right? Nope.

A/D (analog to digital) converters, as well as their D/A counterparts exist in varying quality. Luckily, advancements in technology have improved conversion quality across the board. Converters today are of pretty good quality in general, with a major exception – built in computer sound cards.

Often called 'consumer cards', the built in A/D and D/A of most home computers are of poor quality, are noisy and are very susceptible to *jitter* (see vocabulary section). If you plan on doing anything more than the roughest demos, you will need to get yourself an interface which will do some decent conversion on the way to the computer.

The current generation of USB and Firewire boxes do a fair job of converting your signal to digital. ***Focusrite (Saffire), Digidesign (Mbox2), Maudio (410, 1814 etc.)*** and ***Lexicon (Omega)*** have made careful compromises between cost and performance to make their units price competitive and still fully functional. The difference is, again, that you get what you pay for. As the tracks pile up during recording, the sound quality of the inexpensive converters becomes apparent. Listen to these two recordings, the **first** is a melange of acoustic guitars and percussion recorded through the Digidesign MBox2's converters, the **second** is the same thing through an Apogee MINI ME's converters. *(.wav files **EX11a** and **EX11b**).* Granted, the difference is subtle, but do you notice the way the second recording has a more lifelike dynamic and a clearer midrange and high end? This is one of the finer points, and worth some serious listening time.

The ***Apogee MINI ME*** is a 2 channel A/D unit with a street price of about $1000 ($800 used, like mine!) The difference in the quality of the converters is quite obvious, especially on multiple tracks. Apogee offers a variety of units with up to eight channels of world class conversion. Other great converters are available from ***Crane Song, Lavry, Benchmark,*** and ***Prism,*** but prices run into the thousands of dollars!

RME Audio makes the 8 channel '***Fireface***' firewire converter for about $1500 – an excellent unit that will get a lot of audio in and out of your DAW with very good sounding conversion.

Converters are another important step in the audio chain on its way to recorded immortality (remember to make regular disc backups to insure immortality!) and shouldn't be underestimated. When getting started in your recording hobby/career, you will start off with the converters you have at hand. The point here is not to forget that conversion is part of the chain, and that the pros know that there is another level of quality to be had in that link in the chain. When you've bolstered your studio with some good mics, a few good preamps and some nice plug-ins or hardware effects devices, you may want to seriously consider upgrading your converters to give your sound that professional edge.

...and Bad Myths

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One of the major problems with teaching any subject is overcoming preconceived notions, misunderstandings and superstitions. All of the following topics (and more) will be touched on throughout this manual, but here are a few key points which come to mind:

Myth #1) **Tubes sound warm.** We've touched on this already – but it is worth repeating. Always do an eyes-closed listening test to decide what you like best, don't get fooled by the hype, advertisements, and my suggestions (!??). Use YOUR ears.

Myth #2) **You can fix it in the mix.** This is a load of baloney usually served by lazy engineers who are in a hurry. If you are not happy with a basic sound, all the tweaking in the world will not make it great. It may turn out O.K., or even pretty good, but for a fantastic sound, you need to cut it right from the beginning.

Myth #3) **Cables don't matter.** If you want to say that cables don't make a HUGE difference, then I'll agree. But if you want that extra edge, the investment in a few high quality microphone and instrument cables will be another top-quality ingredient helping to make your mix all it can be. **TIP:** Keep your cable runs as short as possible – longer cables reduce clarity! Remember, these details add up!

Myth #4) **Once it has been recorded, the audio is safe.** Nothing is ever safe. Hard disks contain a very small sensor which can pick up signals from your brain and determine what is the MOST unfortunate time to crash. Be sure to have double-secret backups, so your day/song/mix/life will not be ruined.

Myth #5) **You can soundproof with egg foam and carpet**. Not entirely true. We'll get to this in a moment, but I wanted to bring this up front, though, so you don't run out and spend fistfulls of money for no good purpose.

Myth #6) **You can fix it in the mix.** I'm repeating that this is a myth, because you will hear it repeated in the studio. Don't listen!

Myth #7) **16 bit is CD quality, so it is good enough.** This is an important issue when it comes to digital gear such as mixers. The ***word length*** that the internal processing carries through is crucial to the sound quality of the gear. 24- or (even better) 32-bit internal processing is necessary for the integrity of your signal. If you think you are saving money by going with a 16-bit internal architecture device, you are saving only at the expense of your sound. Buyer beware – ask salespersons and dealers about this specification before jumping in.

Myth #8) **Louder is better.** We will see (and hear!) why this is NOT the case. Not for guitar amplifiers, not for drummers, not for mixes, not for masters. It is great and liberating to blast away, but there are secrets at every turn. Everything is relative, especially when squeezed onto a CD or into an MP3.

Myth #9) **More expensive is better**. Whether it is the studio's rate, the cost of a compressor, or the rate for a studio musician, the bottom line is sound, not cost. A musician who plays the right part to your ears is the right musician, even if she's not from the London Philharmonic. The sound that your Radio Shack preamplifier gets on that kick drum – big and crunchy – is the right sound if it fits the track. And so on. Learn to be critical, but ***use your ears at all times***.

Now we'll start implementing all these ideas and some more to make your recordings sound better.

Setting up your studio

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Now that the research and shopping are done, it's time to connect all that good stuff and cut some tracks. The first thing that goes into the equation is the room(s) you are going to work in.

Acoustics

Room acoustics is a complex subject, and the way in which professional studios are constructed (floating floors, triple walls, double doors, etc.) is far beyond consideration for the home or semi-pro studio. We need to make the best of whichever room we have available in which to make our music.

There are a few basic things that can be done to improve the acoustics of a room without major cost or renovations. First, let's define a few of the common problems that occur in the acoustics of a small room.

1) **Standing reflections**: This is that 'ping-pong' or 'fluttering' echo effect heard when you clap your hands in a hallway or in a room with hardwood floors. Sound reflects from the floor and ceiling, echoing to the opposite surface and back, repeating until the sound dies away, absorbed by the surfaces as a small trace of thermal (heat) energy. This will sound pretty awful on a recording, making vocals unintelligible and instruments and percussion sound cluttered and odd.

2) **Resonant frequencies**: In the same way that blowing across the neck of a bottle creates a resonating tone depending on the size of the space in the bottle, rooms will resonate at a frequency dependent upon their dimensions and volume. The formula for this resonant frequency for each room dimension is:

f = V/2d

Where f is the resonant frequency, V is the speed of sound, and d the room dimension:

f = (1120 feet per second)/ 2 x (the room dimension in feet)

So, if you have a 15' x 20' room with an 8 foot ceiling, your problem frequencies will be:

37Hz (15' dimension), 28Hz (20' dimension), and 70Hz (8' ceiling)

It is not only these fundamental frequencies that are resonant, but also their harmonics (frequencies which are a multiple of the fundamental), those in the 200 – 400Hz range are especially audible and problematic in a small room.

3) **Outside noise**: To contain the sound of your music-making and prevent angry neighbors /roommates/ parents/ spouses, as well as keeping out the unwanted sounds of traffic, lawnmowers, screeching kids, etc. you will need a certain amount of sound insulation from the outside world.

4) **Inside noise**: While not a problem of the actual room *acoustics*, every studio has some inherent noise produced by fans, monitors, hard drives and mouth-breathing long-haired guitarists.

Some of us are lucky enough to have a separate control room and tracking room. The reality of living in a small house or apartment is often that the musician and recording gear are required to be in the same room. With a little careful planning, this problem will affect your recordings in a minimal way.

Dealing with fluttering, ping-ponging echoes is fairly easy. The way to do this is to break up the sound reflections. The best method is to create an irregular surface. There are commercially available wood units built with small squares of varying depths, but they can be quite expensive and are basically non-functional for other uses.

The home studio equivalent is a book shelf! Find the point where the echoes are worst and place a bookshelf at one of the walls. Use this shelf to store a collection of books (which are nice, dense, sound barrier material) as well as your microphone cases, equipment boxes, cables, etc. You will organize and conquer echoes in one shot.

The echoes occurring between floor and ceiling require a little more effort, as we can’t lay a bookshelf on the floor. If you are able to glue or screw things into the ceiling without losing your lease, you may consider some acoustic foam treatment from Auralex. Their lightweight foam can be glued to the ceiling with little trouble.

The room resonances: Unfortunately, just because you can’t hear the ‘ping-ponging' anymore doesn’t mean your problem is solved. A little foam will make the high frequency flutter disappear, but **lower frequencies are still resonanting**, though not as obviously. You can improve the absorption with a thick throw rug on several layers of carpet foam/padding in the middle of the room.

At lower frequencies, it gets harder to effectively absorb these higher energy standing waves. A large couch placed along a back wall or, even better, a love seat that can fit into a corner (corners are the spots where bass frequencies are most heavily reinforced) makes a good low frequency absorber that does double duty as a place to sit.

Luckily, for those of you who are handy with a ruler and a screw gun (or have a brother-in-law who is…) there is an inexpensive way to build your own ***Helmholtz resonators*** and ***Bass traps*** to capture those lower resonant frequencies that are out to harm your recordings and mixes.

Resonators are large, panel-like devices, which can be mounted to walls or ceilings. They consist of two panels with an airspace filled with acoustic fiberglass in between. The panels resonate sympathetically with the problem frequencies turning sound energy into mechanical energy, which is dispersed as a very small amount of thermal energy. The fiberglass inside also acts to dampen sound.

The great advantage of resonators is their ability to absorb lower 'muddy' sounding frequencies without making a room entirely 'dead' sounding. Dead rooms sound very unnatural. One of the worst experiences I ever had as a young engineer occurred in a band room-tuned-studio in the Bronx, NY. The studio manager wanted to make the room 'soundproof', so he hired carpet layers and had the floors carpeted – and the walls and ceiling. The recordings made there sounded like, well, carpet. The carpeting absorbed all high frequencies, making the recordings dull sounding, but provided no low frequency absorption, resulting in a big mud-bath of tracks. I had to leave the studio after three projects, there was no way to get a good sound there and the bands wanted to kill someone!

Since this is a manual on studio *technique*, not studio *construction*, blueprints and long instructions are a bit outside the scope of this book. if you wish to build your own Helmholtz resonator try this link: **http://www.saecollege.de/reference_material/pages/Low%20Mid%20Frequencies.htm**

At **http://www.ethanwiner.com/basstrap.html** there are simple designs and parts lists (under $40 for a bass trap!) for some great sound absorbers.

For some further understanding of the basics of improving the acoustics of your studio without much more than a little elbow grease, use this link: **http://arts.ucsc.edu/ems/music/tech_background/TE-14/teces_14.html**

General studio setup and ergonomics

One of the few exceptions to the rule that you should take the shortest cabling path for an audio signal is the ***patch bay***. A patch bay serves the wonderful organizational purpose of bringing all those audio connections around from the back of your gear and up to a single interface where you may easily interconnect audio as you please.

The patch bay consists of two rows of cable jacks in vertical pairs. Cables can be connected to both the front and the back. Generally, you use short cables to bring the inputs and outputs of the gear to the back side of the jack rows, and you can then make connections from the front with short ***patch cables***.

I have always used balanced patch bays from **Neutrik** ($99) and have had no trouble or sound degradation. The convenience of not having to climb behind my gear was worth every penny.

The nice thing about a patch bay is that you can set them up so that connections on the back panel are '***normalled***' together. This simply means that the top row of the back is normally connected to the bottom row on the back , unless you insert a cable.

An example for a normalled connection is that you may always want the outputs of your favorite mic preamplifier (1&2) connected to the inputs of your recorder (1&2). So you connect the outputs of the preamp to the back top row; jacks1 & 2. Connect the inputs of your recorder to the back bottom row, jacks 1 & 2. Now your mic pre outs are normalled to your recorder ins.

Now, imagine that you would like to put a compressor in between mic pre #1 and input #1 of your recorder to do some vocal recording. All you have to do is plug a patch cable into the *front* top cable jack #1 to get the output of the mic preamp. This ***disconnects*** it from the recorder input. Now plug this cable into the compressor input. Take the compressor output and patch it into the front bottom row jack #1 – it is now reconnected to your recorder input #1. If you think about the gymnastics this would require without a patch bay, you'll understand why a busy studio couldn't function with a patch bay.

A ***normalled patch bay*** is likely the most common wiring, although ***half-normalled*** and ***denormalled*** have their uses as well.

Denormalled is easier to define. There is simply no normal connection from top to bottom. Only front to back. I have a denormalled patch bay for effects devices and compressors, so that I can keep their inputs and outputs over each other without causing a feedback loop (output connected to input – yeaoOOOOWW!). This is useful just for organizing things.

Half-normalled is just like normalled, except that plugging in a cable does NOT disconnect the top back row from the bottom back row. This can be useful when you need a copy of a signal for something like a compressor side chain to create a de-esser (more on that later).

Abnormalled what you will become if you actually try to memorize all this patchbay shit in one sitting.

Patch bays are the height of convenience, allowing fast connections and improving spontaneity in the studio. ***Additionally***, I recommend keeping the following handy as your budget allows:

- an accurate digital tuner with variable reference tuning.
- a string winder and a few sets of spare guitar strings.
- a mic stand with a Shure '57 and a short cable within arm's reach
- blank CD's.
- extra instrument- and patch cables – there is nothing worse than stopping a session because there is no working instrument cable. This WILL happen to you, so be ready.
- a 'popper-stopper' or similar microphone pop filter (in a pinch you can stretch a nylon sock or panty hose over a coat hanger bent into a loop).
- an XLR to ¼" balanced adaptor, and the other way around.
- A D.I. box. Whirlwind makes decent ones, the 'Countryman' D.I. is even better. This can cure buzzing and noise in a hurry and save a session.

When your wiring is all smoothed out and easy to deal with, it is time to be sure that you are comfortable in your workspace.

Proper lighting prevents eyestrain and headaches. Beg, borrow or steal some lamps and place them where they will give some indirect lighting, soft on the eyes and conducive to relaxed working. And sweet love.

Place mixers and computer monitors at a comfortable level to prevent back and neck strain. A stiff neck will distract you from your work, and your mixes will suffer.

If you can afford the extra space, put a couch, some bean bag chairs, or some kind of comfortable furniture in the studio. Aside from a good work chair for the mix position, you need an alternate place to sit or lay and listen to your work. The relaxed body allows relaxed and focused ears. And sweet love.

Aqua de la Viva: Electricity

No matter where you build your studio, your water of life is the electricity to run your gear. Unfortunately, AC (*Alternating Current*) electricity has a frequency of it's own – 60 cycles per second; 60Hz. This frequency and it's harmonics (120Hz, 240Hz, etc.) can sneak into your audio signals on occasion, and you need to know how to deal with this.

First of all, try to get clean power to your gear. At the very least, you will want to keep a surge protector between the wall AC and your precious gear. Power companies are not too concerned with sending you pristine power; they have enough trouble just getting it out to you. So spikes in the voltage, surges in power and general noise can be a problem.

The best protection you can have is a ***power conditioner*** that also provides you with a small reserve of power in case of power failure. Since power dropouts are not uncommon, especially in very hot weather when everyone is running air conditioners, you may wish to have a ***UPS*** (**Uninterruptible Power Supply**) covering your back so that all your gear doesn't just flash off during that magic final vocal take!

Next, we need to use a little common sense. Some household devices, such as refrigerators, air conditioners and lamps with dimmers can introduce noise into or create spikes in your power. Try to get all of your studio gear onto a separate circuit breaker than other household toys. Vacuum cleaners and blenders are able to transmit their whirring and humming right into your recorder!

Very important when wiring your studio is to use balanced (three conductor) wiring whenever possible. Balanced wiring can help cancel out noise from stray currents that may find their way into your gear. Additionally, keep cables as short as possible for every application. A long cable is essentially an antenna and will eagerly pick up stray noise from computer monitors, hard drives, radio stations and T.V. broadcasts.

All AC power chords in your studio are also broadcasting noise, so never run an audio cable alongside a power cable. In fact, keep them as far away from each other as you can. Obviously they will come close at the gear, but avoid contact until then. If you absolutely must cross an audio cable over a power cable, cross them at a right angle, thereby minimizing the contact of the AC noise field with your audio line.

DC power lines (like those coming from a 'wall wart' power adapter are not noise producers, but the wall warts themselves are notorious noisemakers. Keep your audio cables far away. You may want to run a separate power strip for these adaptors to keep them as far away as possible to prevent their transformers from inducing hum into your audio lines.

Sometimes the power supplies inside a piece of rack gear can induce noise in a neighboring piece of gear. In this case, try leaving an empty space in the rack between the offending unit and it's neighbors.

The last way in which noise can invade your otherwise perfect recordings is through a ***ground loop***. If two electrical devices are connected to ground (the ground is the third prong of the AC cable) at two different points, a difference in potential (a voltage) can exist between these two points. When you plug in an audio cable, the cables shield connects the two devices allowing a current to flow. Zzzzzzz.

The most practical solution is to ground all equipment to the same point. This minimizes the possibility of a difference in ground potential. Just be sure that the circuit can handle all that juice, or you'll blow a fuse. Luckily, most audio gear (with the exception of power amplifiers) draws a fairly low current.

Do NOT, I repeat ***DO NOT*** just remove the ground plug by 'lifting' it with an adaptor or cutting it off. Should a short circuit occur in a piece of gear and you touch that piece of gear, YOU become the shortest path to ground, especially if you are holding something grounded, like a microphone. Remember Jack Nicholson at the end of "One Flew over the Cuckoo's Nest"? Did he look like he could still engineer a good album? Nope, and the Chief didn't think so either.

As a last resort, albeit an pricey one, you may have to connect your audio through an isolation transformer. This is in essence a D.I. box. Because a transformer passes a signal by magnetic induction (through the air) there is no physical connection through which the 'ground loop' can occur. The drawback is the additional component added into your audio chain and the added expense of the iso. transformer.

Some software plug-ins allow you to remove noise *after* the recording is finished! This works on any sustained noise, like computer monitor buzz or AC hum. These software programs can actually be surprisingly effective and transparent sounding. All that they do is ask you to highlight a region where the noise occurs alone – like in between words on a vocal track, or in the silence before the song starts – and it analyzes the sound. The sound of the noise is then subtracted from the entire track mathematically. Always try it out on a copy of the track or be sure that you can undo the processing in case you are unhappy with the result.

Always be methodical in your search for the source of noise in your studio. Try powering down one piece of gear at a time until the noise goes away. Then you'll have the culprit – investigate its audio connections, power connections and proximity to AC lines or power transformers. Take it step by step and you will be able to reduce the noise in your workshop significantly.

Educated gear connection

I know and love the excitement of unpacking a newly arrived piece of gear – the *rrrrip* of the packing tape, the cloud of Styrofoam peanuts, the flutter of the user's manual being tossed aside and... Ahhhh! There she is!

I rush into the studio, grab a couple cables, make some quick audio connections, plug 'er in and start twisting knobs. After cutting a few tracks, questions start to pop up; How do I adjust a certain parameter? What does this knob do? Should the cat be allowed to chew on that power cable?

There are so many possible audio preamplifiers, processors and manipulators out there that it is hard to set any rules on how to interconnect them. There are, however a few MUSTS:

1) ***A microphone or instrument level signal must be brought up to line level to work effectively in another unit.*** This means that you must *preamplify*. Some units contain a preamplifier before their compressor or digital effects. Be sure to evaluate this preamp and see how you like it. If you have a superior preamp, use that and then go into the line level, not mic level, inputs of the unit afterwards.
2) ***Once you go digital, don't go back.*** The A/D and D/A conversion process can only degrade your signal. Even the best converter's goal is simply to minimize degradation of your original signal. Use the best converter you have to go digital and then stay digital all the way to tape (disc). Remember – any digital reverb or effect, compressor, EQ, etc., coverts your signal to digital. If you don't come out of this device via the 'digital out' you are converting it twice!
3) ***Don't dither on your way to disc.*** Some devices offer you the option of dithering (reducing the bit rate) at their output. Try to record at 24 bits all the time. Also make your mixes at 24 bits. Reduce it only at the last possible step (mastering)
4) ***Monitor your tape (disc) outputs at least when checking the sound.*** There are ways to avoid latency issues (as described in a later chapter) by monitoring the signal before it goes to disc. Just be sure that, latency aside, you listen to the output of your DAW before you start tracking those magical notes of yours. If you have a few boxes plugged in ***before*** tape, you want to be sure you are not just recording the screech of some piece of gear with it's eyes crossed in a feedback loop.

Getting a good signal to tape

The first step in creating a good sounding music recording is to record great sounding basic tracks. It is hard to make good tracks sound bad in a mix. It is even harder to make bad tracks sound good. Using your gear to its fullest capability is an art unto itself.

It is, of course, impossible to know what gear you have at your disposal for recording at home or in your studio. So we are going to look at the ideal recording setup, and then look at how you can imitate this as closely as possible with the gear you are using. This will get your gear sounding as good as it possibly can.

There are a few principles to apply when doing this:

1) **Select a proper microphone for the job**. You cannot undo your choice of microphone later. For details, refer to the next section on "Microphone choice and placement".

2) **Use balanced cables and minimize cable runs**. Shorter is better. It is also best to use balanced (three conductor) cables whenever possible. Most gear offers XLR or ¼" TRS balanced inputs and outputs. Use them! You will not be sorry - unbalanced cables are tone sucking noise antennas.

   Minimize the cable length from the microphone to the preamplifier. It is better to send a line level signal through a long cable than a mic level signal. This may mean locating your preamp close to the sound source. Listen to this: An acoustic guitar recorded by connecting the mic with 50 feet of cable (first), and the same guitar and mic connected by six feet of cable (second) ***(.wav files EX12a and EX12b)***. Do you hear the difference? Pay close attention to the high notes (open strings on the guitar) - can you hear how they are less present on the long cable recording? Do you notice the loss of definition in the upper midrange? These are the finer points at work.

3) **Set your preamplifier gain correctly**. Do a dry run with the sound source. Find out what the softest and loudest notes are going to be. Keep in mind that, especially in a rock or pop setting, people are going to play or sing a little harder when tape is rolling. Record the signal at a high enough level to take advantage of your recorder's full resolution, but leave 3 - 6dB of headroom to prevent clipping. You may need to track with a compressor or limiter set after your preamp to catch unexpected peak on the way to tape (disc).

4) **Don't put any more crap in the way.** After your preamp, get to tape as quickly as possible. This may involve connecting your preamplifier output directly to your sound card, computer interface or recorder inputs. A mistake that a lot of people make is to route their preamp outputs to tape through a mixer. By doing that you are putting so many more amplifiers in the signal path that you degrade your sound many times over.

5) **Use the best possible A/D converter at your disposal, and do not return to analog once you go digital**. You must beware of the hidden pitfalls of using a digital box before you go to disc! If you have something like a T.C. Electronics Finalizer or a Lexicon reverb, that is great – they have some great sounds in them. But

once you go analog into them, come DIGITAL out and connect to your sound card or interface via SP/DIF or AES/EBU. Otherwise, you are going through the A/D converter of the effects box, out through its D/A converter, and then into your computer or digital recorder through its A/D! Talk about torturing your poor signal!

6) **If you're recording on a computer, be sure that you or your recording software is not placing any processing in the signal path on the way to tape.** Don't use dither, EQ or other plug-ins on the input to tape (disc). Check your manuals to learn what is exactly happening on the way to disc.

To review, let's now imagine a perfect scenario, and use it to set up your studio recording chain.

- We have a fine quality, large diaphragm microphone set up for the singer. There is a short, high quality mic cable leading to a world-class mic preamp, which is only a few feet from the mic.

- The mic preamp is set to a proper level, and the output runs at line level into the control room.

- There is a high quality compressor or limiter as the next step in the chain, set to catch any unexpected peaks in the performance.

- The output of the compressor is sent to a top notch, low jitter A/D converter and fed to the digital input of the recorder.

- The engineer is awake and sober at the record button.

Other than the last step of the above scenario, this is what happens every day at the best studios in the world. So how can this help you?

Well, let's say that you have just a Shure SM-58, a Mackie 12 channel mixer, a Behringer compressor and you computer with an audio recording quality sound card running Cubase.

A typical setup I see all the time is this:

The microphone is connected to an input of the mixer. The EQ on the mixer is boosting some highs and lows to "get a fat sound". The mic signal is sent to one of the mixer subgroups, which are connected to the soundcard through the compressor. The engineer controls the volume of the microphone with the subgroup or channel fader.

Let's rewire this entire situation. If the above situation was you, try this and tell me what you think as the tracks start piling up.

Connect the microphone to an input, and take the signal from the 'direct out' connection on the mixer's back panel. Even many inexpensive mixers have this option. By doing this you by pass the EQ circuits, fader, subgroup faders, etc. Less electronics = purer tone.

Now, skip the compressor and go right to your computer's input. Adjust the mic level so that the loudest level is -6 to -3dBFS. Later you can use some plug-in processing or internal processors to compress peaks and bring up the levels a bit. The major advantage is that you will be able to change those plug-in settings as you work; compression that you do to tape is written in stone.

With this variation on the way to tape, you will improve the quality of your recordings in a big way. It is hard to hear the difference on individual tracks, but it adds up.

There are many of you out there working with an MBox setup or something similar from Mackie, Lexicon, etc. The nice thing about a dedicated box like the MBox, Saffire, MAudio 410, etc. is the short, self-contained path to disc. Plug in your mic and you're rolling. No other connections. But can this be improved?

Here is a series of recordings of acoustic guitar. The **first** take is just the mic into the MBox 2 ***(.wav file EX13a)***. The **second** recording is a Demeter Class 'A' tube mic pre into the MBox, bypassing the MBox preamp (line in) ***(.wav file EX13b).*** The **third** recording goes from the mic pre, to an Apogee A/D converter, and then *digitally* into the MBox (in essence bypassing the MBox entirely – it just passes along the digital signal) ***(.wav file EX13c)***.

Compare the recordings (and please excuse the shaky playing!) These types of interfaces can do a fine job of recording your signals, but it is important to understand how much of a difference each step in the recording chain can make. This will help you decide what is most important when investing you hard earned money in new gear for your all-consuming hobby…

Monitoring

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Now that we are wired up and ready to roll, before we start laying it down, let's take a look at how we are going to listen to all this wonderful music.

Being able to hear what is going on in the studio is of utmost importance, and this doesn't count just for the engineer, but for all involved musicians, producers, and assistants.

There are many fine studio monitor speakers available today, and I am of the opinion that it is not as important *which* speakers you use, but rather how well you know those speakers. So, instead of rattling off a bunch of brands, I want to look at the broader picture.

It is important to be able to hear the highest high frequencies as well as the lowest lows, but let's not forget about the real world. Many people still listen on a small home stereo or a boom box that won't reproduce frequencies much below 100Hz or above 10kHz. In this respect, it is important for you to have, in addition to some full range monitors of your choice, a pair of consumer fidelity speakers ready for comparison listening.

I used to use a Sony boom box that had 'aux inputs' via RCA connectors on the back. I could simply turn down the main monitors, switch on the boom box and see what was happening. Often I would find that those deep, satisfying low frequencies would either disappear on the boom box or just overload it's wimpy consumer power amp, causing the little speakers to blubber for help. The same goes for guitar tones. Bad on the boombox, bad everywhere - if they're smokin' on the boom box, they'll sound great anywhere else.

For your main high resolution monitors be sure:

- Decouple them from the surface they stand on with some foam padding. This will prevent exaggerated bass response by means of physical conduction through solid objects.
- Don't put them in a corner, this will multiply the apparent bass output, ruining the accuracy.
- Don't set them on the mixing board; get them up to a level wher you will sit up straight to listen. This is not only good for your back after long hours of audio work, but will prevent sound reflections off of the mixer which ruin the accuracy of the monitors

- Keep the listening volume at 85dB or below. It is right and good to 'crank it up' once in a while and see how it feels, but it is both unhealthy and inaccurate to work like that. First of all, hearing damage is real. When your ears ring, they're saying "you hurt me". Don't do that often if you like music. Second, Our ears naturally compress at higher volumes, so your sense of dynamics and detail is lacking at high volumes. Not very professional.
- ***Always follow the "Last on, first off" rule***. Turn your monitors/ monitor amp on last and off first. This will save you blown tweeters, popped cones and shattered nerves!

Next up in monitoring is the tracking room side. Headphones are very important here. You will likely need at least two pairs of headphones and probably four or more. I would suggest doing some listening in a music store and buying both a pair that sounds very accurate, and one that has a pleasing, consumer sound – extra lows and highs.

I find that some musicians (especially hobbyists) would much rather hear a flattering sound in the headphones to 'get a vibe'. Consumer phones tend to add some bass and treble boost to give the music some 'thump and sizzle' – impressive but not accurate. In a situation where you are tracking music in the same room as the recording equipment, you will need to be wearing headphones as the engineer, as well. You should have an accurate pair of AKG's (or similar) for more discerning listeners, such as yourself.

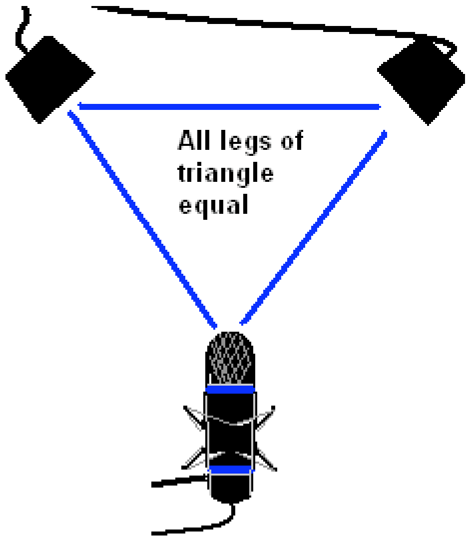
If you are tracking multiple musicians simultaneously, you will also need a headphone distribution amp, so that you may connect multiple headphones and potentially create alternate mixes for picky singers. Yes, it will be the singer.

In this case, you may want to go for that ***Behringer*** or other inexpensive unit. Sound quality is important, yes, but when the bass player sets his Grape Slurpy on your headphone amp and condensation seeps down the side, you'll be glad it was a cheap headphone amp.

All kidding aside, there are a variety of options available today, some of which offer connection via network cables, multi-band EQ for each pair of headphones and individual mix levels. You will need to look at how many features you think you will need versus what you are willing to pay. Don't forget that many things can give you an extra headphone amp in a pinch, from a little Mackie mixer to that boom box, all you have to do is feed it a signal from an aux output or mixer buss, and you're in business.

TRICK: Some vocalists just aren't totally comfortable with headphones, and if you're willing to take a little time and effort, you can blast him or her with loud monitor speakers – right at the microphone – with very little bleed into the mic. This trick is really like audio magic, and takes advantage of the *phase cancellation* phenomenon.

To set this up, you need to set up a pair of speakers and a microphone like this, in an equilateral triangle:



Then create a MONO mix of the music which the singer is to sing to. Some mixers have a mono button, or you can just pan everything to one side and use that. Feed this mono signal to BOTH speakers.

Now reverse the leads on one speaker (positive to neg and neg to positive); this put that speaker out of phase with the other. So the two mono versions of the music arriving at the microphone are out of phase, adding up to no sound!

It sounds like voodoo, but it really works! You should record a bit of the music coming out of the speakers and hear for yourself – if it is still loud in the mic double check your measurements.

The great part of this is that since the singer is not standing exactly at the phase cancellation point, they can still hear the music loud and clear. Tracking a vocal like this can be a great experience for a singer and may win you repeat contracts for your studio!

A digital dilemma: dealing with latency

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Latency has become such a buzzword in the recording world in the last few years that it is almost sickening.

There is latency in everything. When you take a class at the university, and the professor standing at the front of the room speaks to you (and I know you are sitting in the back row, slumped into you chair), the words he speaks must travel through the air (at 1100 feet per second) before they reach you. If you are so unlucky as to have one of those big auditorium collective college courses, you might be, well, 110 feet away from the professor. That is a latency of 100 milliseconds ($1/10^{th}$ of a second)! So, if you are accused of giving an answer too slowly, explain to your professor that it's all about the latency.

Every digital device (including the A/D and D/A converters) incurs latency before passing along the signal, so there is no such thing as latency-free *through your computer*. A well-tuned computer of 1GHz+ processor speed is capable of fairly low latency monitoring - check you hardware and software manufacturer's suggestions and look into support forums for experienced user tips on "tuning up" your computer for audio recording. Extensive operating system tips are beyond the scope of this book.

An excellent way of getting around the latency problem involves a little extra gear, but will work wonders. Since all the popular programs compensate for the difference in time between your performance and the audio arriving to disc (by shifting the recorded track the appropriate number of samples), you needn't worry about that. You just need to be able to hear yourself as well as the backing track with no strange delays between your playing and what you hear.

The 'blend' knob on all the current USB and Firewire boxes (Mbox, etc.) gives you a direct signal, totally dry and boring. I like to run the outputs of the MBox 2 into a little cheap mixer I have, and send a clone of the input signal to the mixer. You can do this with a second mic on the instrument or with a second output from your preamp, if you have it.

I then have a hands-on monitoring mix, which is a good feeling in itself. I sometimes get annoyed and uninspired when forced to mouse-click around when trying to be creative, and the mixer in front of me lets me grab a knob or shove a fader without affecting the signal going to disc. I also have an old Boss reverb hanging around on its last legs, but it's good enough to add a little 'verb to the monitor mix!

If you're not an old packrat like me with extra gear lying about, you can still grab a cheapo mixer used for under a hundred bucks and solve all you latency problems in one shot. And considering that the latency issue is one of the only things separating the big-time DAW's from us project-studio types; that is a pretty inexpensive upgrade!

For you ProTools LE users: This is my current DAW of choice, so I do have a few tips for you. Just for your reference I am using an Intel Mac (MacBook Pro) with dual 2 GHz processors and 1 GB of RAM. I opted for the Mbox 2, even though I am aware that Firewire is a bit faster (see definition of Firewire).

First off, for other Macbook Pro users, **only connect the Mbox2 to the LEFT SIDE USB PORT!** If you are not doing this, this will be the first source of error messages. The internal architecture is such that this is your only option.

Now, go into your hardware settings and select the lowest latency you have available (128 samples for me). Open up 8 tracks, record enable them and let 'er rip. If it doesn't hitch at the start, let it run for a minute. If you have dual processors, select 2, and set the CPU usage to the maximum possible (90%). See if you get an error upon stopping.

If you do get an error message, reduce the number of tracks and try again. Find your limits and remember them; you will have to be realistic about what your system can handle.

If you haven't gotten an error message yet, start opening up plug-ins on the tracks. Put an EQ on every track. Then a compressor. Sooner or later your computer will yelp for help. This is your limit at that latency setting. The good news is, by increasing the sample latency to 512- or 1024 samples you'll get many more plug-in instance for mixing time.

If you find yourself going nuts with a latency you can't seem to find, make sure you are not using any plug-ins on your master fader while tracking! The internal processing latency of many plug-ins is far too long, even if you're set to 128 samples in the hardware options. I did this and was very relieved when I discovered the problem.

Microphone choice and placement

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The microphone is *always* the first step in the recording process; we need to become experts in microphone use in order to be great audio engineers.

Some sounds are easy to capture, some are not so easy. And you need to consider the bigger picture at all times – how should this sound fit into the mix? Should it be up close in your face? It is it a background part that needs some 'air' between it and the lead parts? Should it sound crisp and bright, or warm and fuzzy?

Let's take the topic of mic choice and placement on an instrument-by-instrument basis, and I will provide some examples of the way these methods sound (in my home studio setting) to give you some points of reference.

The most important aspect of recording is to *use you ear*. Before you decide on a microphone and it's placement, move yourself around near the instrument as the performer plays, and find a 'sweet spot'. This will be a place where the instrument sounds balanced and clear – just plain subjectively *good*. Then, turn on the objectivity switch. Is the sound a bit bright overall? Or maybe muddy sounding or bass-heavy? Is there too much natural reverb in the space? These are questions that should affect your mic choice. Here are tips for different sound sources:

Too bassy/muddy: Opt for a small diaphragm condenser - they tend to have a bit less sensitivity to bass frequencies than large diaphragm condensers. Consider a mic with a 'bass roll-off' switch to tame unwanted low frequencies.

Too bright/trebly: Use a ribbon mic, which tends to sound 'warmer' and 'bassier', especially at close range. Large diaphragm condenser mics are a little less sensitive to high frequencies than small diaphragms, and will downplay harsh and bright transients. Consider a dynamic mic, their slower transient response can also downplay overly bright sound sources.

Too much ambience/reverb: Many home studios suffer from poor acoustics. For those of us living in apartments or spaces that cannot be heavily renovated into a music studio, we have to deal with the acoustic limitations handed to us. Three cheers for the directional microphone! Obviously, the cardioid mic somewhat ignores sound coming from the back, so this will reduce any unwanted echoes or reverb. So is a hypercardioid pattern even better? Strangely enough, it probably isn't. The extreme off-axis rejection of sounds by a hypercardioid microphone can sound quite unnatural. Hypercardiod mics are more at home onstage. Surprisingly, a figure eight pickup pattern (like many ribbon mikes) can often pick up enough ambience to give a realistic sound to the recording without picking up too much cluttered sounding reverb. Don't ignore this classic polar pattern – give it a listen and see what you think.

Acoustic Guitar Mic Technique

There are so many guitarists out there suffering from a lousy sound, that we have to start here. It works in your favor as an engineer that many people can play at least a few chords on the guitar, because for your own first experiments in recording acoustic guitar you will want to have a guitarist handy.

Start, of course, with a well strung, set up, and tuned instrument. Poor intonation, bad tuning and the crusty strings that were used at the original Woodstock festival will only cause you grief when trying to get a good sound. Always start with the basics before you start mic'ing up a source.

There is more than one way to mic an acoustic, as the old saying goes. **A pretty standard choice for a mono track is a small diaphragm condenser** like a Neumann KM84, Shure 81, or AKG 451. Place the microphone in front of the guitar, about 12" away and about midway between the sound hole and the bridge. Now angle the mic in about 45 degrees, aiming towards the point where the neck meets the body, like this:

This method provides you with a well-balanced sound most of the time. If the sound is lacking bass frequencies, try aiming a little more towards the sound hole. If you need more treble frequencies and pick attack, aim for the bridge.

TIP: To prevent a foot-tapping guitarist from creating rumbling sounds in the mic (due to physical conduction through the mic stand) you may wish to place a layer of foam under either the mic stand or the guitarist's foot. But don't forget, especially for blue players, that this foot tapping can be an integral part of the performance. I have even used an extra mic for the foot! (listen to ***EX_CHARLIE_FOOT.wav***. That is not a bass drum, butthe foot played back through an amplifier!)

I have also experienced big, mysterious rumbles in an acoustic guitar mic – caused by the performer *exhaling* at the mic! This is hard to avoid during a concentrated performance, so consider using a windscreen (pop-filter) between mic and guitarist. Here's what happened to one poor fellow at a session when we just couldn't get rid of the breathing noise:

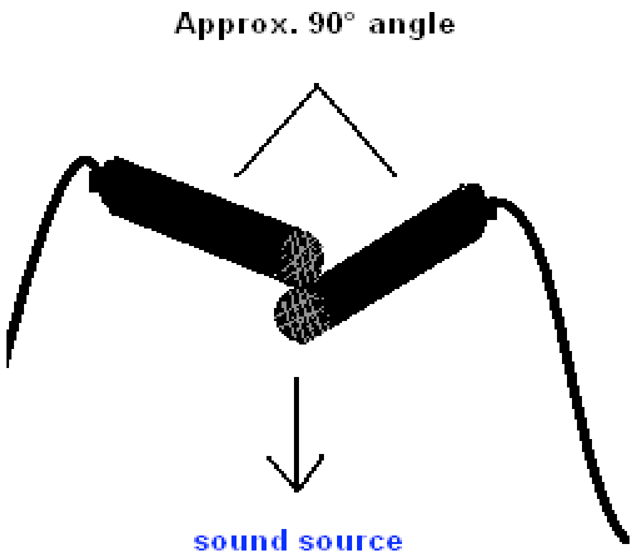
A good sport – and a hell of a snorkeler.

The next method is what I like to think of as "taking a picture of the sound source". Use a large diaphragm condenser (AKG C414, Shure KSM32, etc) and set it up 3 – 5' in front of the guitarist, as you might set up a camera for a close portrait. Use you ear to find a sweet spot while the guitarist plays, then place the mic there. Try an omni-directional pattern for starts. In many home studios, the room ambience and resonances will be less than ideal, so consider switching to a cardioid or figure-eight pattern if a trial recording picks up too much ambience.

Don't forget that many acoustics have a pickup nowadays. I personally hate the Ovation PXM pickup "Dave Matthews sound", but it has become popular, so we have to live with it. What I like to do is run an acoustic through an amplifier with a little edge to it. Listen to this broken-down Tom Waits style country intro, the tremolo-effected guitar is my favorite acoustic through an old Ampeg amp. The acoustic just had more twang to it than an electric! ***(.wav file EX_COFFEEBEER)***

A very full and detailed acoustic guitar sound can be recorded by using a stereo pair of microphones. The important thing is to keep the capsules of both mics as close as possible to avoid phase cancellations. For this reason, small diaphragm condensers are an excellent choice. You can either set them up on two different stands, or invest in a special clamp device that holds two mics at once.

Place the microphones at a 90° angle, capsules one on top of the other, like this:

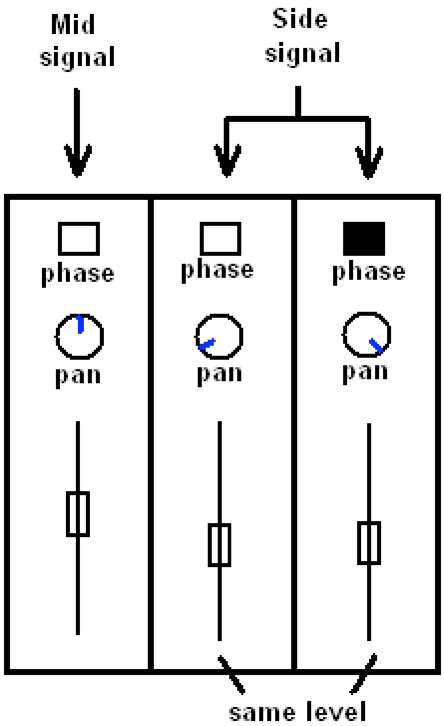


The M-S (mid-side) technique can provide a very detailed and accurate acoustic guitar sound. For locations with good natural acoustics, this is an excellent way to capture the ambience and still have the flexibility to adjust how much of this ambience is used during the mixing stage. Let's look at how to set up an ***M-S matrix***.

An **M-S** mic setup requires two mics, one with a cardioid pattern, the other set to figure-8. The ***Mid*** microphone (cardioid) faces the source and picks up the sum of the left and right stereo image (L+R). The figure-8 faces to the ***Sides*** and picks up the difference (L – R).

The major advantage of the M-S setup is that you are facing the cardioid (mid) microphone directly at the source, like the 'take a picture' technique above. This provides a clear image. The side mic, in figure-8, picks up the ambience, and can be adjusted when mixing to widen or narrow the stereo image.

The signals from both mics need to be decoded in an M-S matrix, which can be set up on a console like this:



Take a listen to these examples – First, a spoken voice moving in the stereo field ***(.wav file EX14)***. Then we have a stereo recording of acoustic guitar ***(.wav file EX15).*** The stereo image is adjusted by adding more of the decoded 'Side' signal and then returns to mono over the course of the track. A very real 3-D effect! We'll use this on other sources as well.

Electric Guitar Mic Technique

There are more voodoo stories about recording electric guitar than about than any other instrument: Modified amplifiers, variacs, a one-of-a-kind Shure SM-57 owned by producer X, miniature amplifiers cranked to the hilt, records cut in in taxicabs and telephone booths, entire studios repainted for "a bluer reverb", chickens turned loose in the studio to defecate on the 'chosen' amplifier...

Yes, many of these things are true. I have seen them with my own astounded eyes, heard them with my own forever damned ears. I will now tell you some of the stories to which I have not sworn mortal secrecy.

The one and only rule for electric guitar recording is:

"The right sound is the one that fits the song."

An entire encyclopedia could be written about the guitar sounds recorded through the years, so the ideas presented here will not be comprehensive by any means. But you will have enough ideas to fill up hundreds of defenseless little tracks with pulsating, slashing electric guitar euphoria.

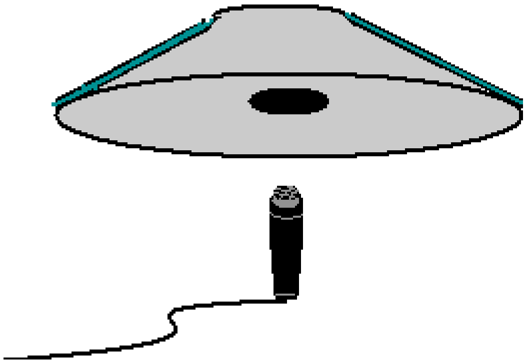
The idea of close mic'ing an instrument was, for a very long time in studio recording history, simply not a consideration. With a large number of musicians to capture on tape and a limited number of tracks and inputs, most sound sources were picked up from a moderate distance to sort of 'snap a picture' of the music. It is no wonder that the first multitrack recordings (and recording devices) were made by a guitarist - Les Paul. His innovations ushered in the modern era of processed, 'bigger-than-life' recorded guitars. Hail to the chief!

The basic tools you'll need: An SM-57, an amp, and a guitar. The possibilities with this modest tool kit are numerous. Add another '57 to the formula and we approach limitless. You can, of course, substitute in other microphones, such as the Sennheiser MD-421, MD409, or it's modern remake the e609. But for now let's look at and listen to some of the possibilities for microphone placement using the ubiquitous '57.

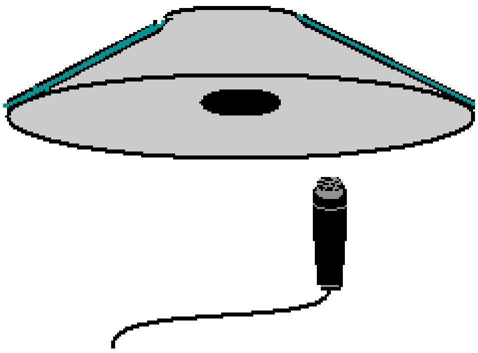
It is still a good idea to place an additional microphone 6-10 feet in front of the amplifier to pick up a sense of 'air' and space. An M-S stereo mic pair is even cooler, capturing a 3-D picture of the acoustic environment. But the the de-facto standard is a good close-mic'ed amplifier sound.

Now, have a look at a few diagrams of mic placement options, and listen to recordings of each mic position. For each example, there is a simple guitar riff played on a telecaster through a small tube amp combo set for a pretty standard 'crunchy' rock tone.

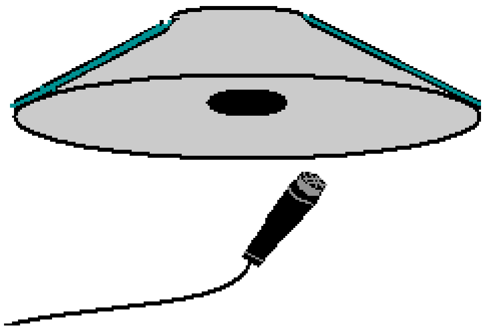
EX16) Mic pointed at center of cone, perpendicular to amp.



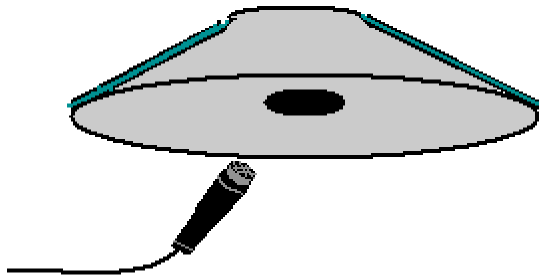
EX17) Mic pointed at side of cone, perp. to amp.



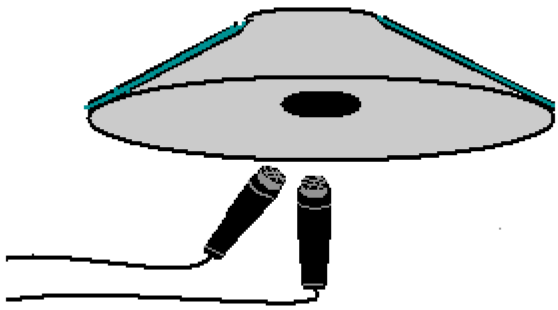
EX18) Pointed at side of cone, perp. to cone.



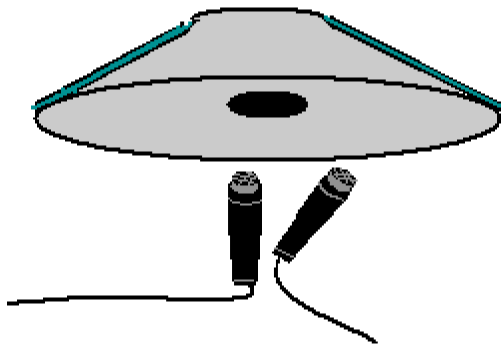
EX19) Pointed at center of cone, 45 degrees off axis.



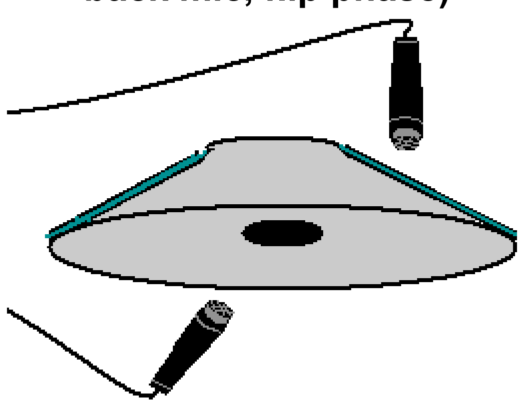
EX20) 1 & 4



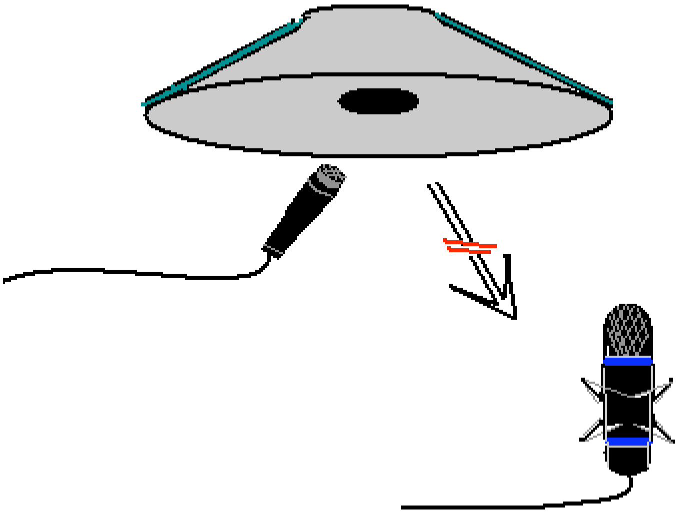
EX21) 1 & 3



EX22) Front and back of amp (choose best front sound, add back mic, flip phase)



EX23) Subjectively best sound, with room mic blended in.



Many big studios have a closet full of mics, so another way of implementing the multi-mic techniques above is to throw a whole melange of mics in front of and around the amp and studio, and then listen to different combinations in the control room. For those of us with less equiptment to spare, look at it this way; the diagrams and sounds listed above are starting points. An odd angle, a mic on the floor, at the side of the amp, or a happy accident where the vocal mic set up across the room gets the best sound - these can be great, and you shouldn't rule anything out.

The next step - if you have the right tools - is to try these techniques out with a ribbon mic. Listen again to the difference between a '57 and an Oktava ML-53. Ribbon mics love guitar amps, they pick up a smooth low end and a little more ambience. ***Shure SM-57 and Oktava ML-53.***

Now, on to the dastardly fun stuff. For kicks, I'm going to present you with a list of audio files, and then a list of ways in which the guitar amp was set up. Try to guess which is which, and then check in the appendix "answers" to see what you get right. You may not believe your ears. ***(.wav files EX24 EX25 EX26 EX27 EX28 EX29 EX30)***

1. A guitar played through a busted set of headphones - the earphone taped to the screen of a condenser mic.
2. Played through a cranked tube preamplifier, then run into a stereo. Mic'ed with a '57.
3. A '57 inside a toilet paper tube and placed back about 6" from the amp.

4. Guitar into Vox Pathfinder w/ 8" speaker($75 amp).
5. Guitar into Pignose G40V w/ 10" speaker ($200 amp)
6. Guitar into Marshall ---- 4x12" speaker ($$$ amp)
7. Guitar straight into ProTools, using Amplitube.

The great thing about a blind test like this is that you cannot start off by thinking, "The big Marshall amp sounds the best." You have to use your ears to decide which is which - and which you like - so you remain objective. Some of the very best sounds ever recorded have originated from some very unusual setups. The reason these sounds made it onto successful records is that someone listened objectively. **"If it sounds good, it is good."** (quote attributed to both musician Duke Ellington and recording gear designer Joe Meek.)

There are two schools of thought when it comes to effects and guitar tracks:

One school of thought is that you should record a guitar track without effects, just a basic well recorded sound, and then do any effects processing in the mix.

The other direction is to get the guitar sound you desire, with all the fuzz, tremolo, chorus, wah-wah, and echo already on it, and capture that on tape (disc).

The truth is, it is not possible to remove an effect from a track when it has been put to tape that way. You can't de-fuzz, un-chorus, or remove echoes. These things can, however, be added later with freedom to experiment and remove the effect if it doesn't fit. On the other hand, an effect can be integral to a performance, help you gain perspective on how the track fits into the mix, and just be inspiring to listen to!

The delay effect on the guitars in "Every breath you take" by the Police is a great example of effects making the track come to life. Anything by Jimi Hendrix is also a valid argument for printing effects to tape. The way the performer reacts to the dynamics of effects setting can be lost if the effect is not printed and something in the effect changes on the next playback or during mixing.

My advice is to be sure (or make sure your performer is) of the choice of effect before printing it. If the song was written around a particular tremolo and spring reverb effect, then it may be essential to get this onto tape the way it is. But beware of the temporary thrill, 'cotton-candy' effects (lots of fluff with no substance!) which are fun to play with for five minutes, but will sound out of place, annoying, or just plain silly in the track later.

Now a couple of fun tricks and helpful techniques:

- To aid in finding a good mic position, some engineers will put on a set of headphones and take the mic in hand in front of the amp. The headphones are set to monitor the microphone, and the amp is fed an input of white noise (random noise with an average of equal energy from all frequencies - like the T.V. with no input cables connected). Now, by moving the mic around in front of the amp, you can choose a position which doesn't sound harsh, emphasizes low frequencies, etc. as you choose. This is an excellent way to attain a natural EQ setting. If you get the right sound at the mic, you're on your way to a killer track.

- I imagine that almost everyone has heard of this trick now, but here goes. Get an assistant to twirl an SM-57 on a cable (preferably not your best mic or cable, just in case) in front of the guitar amp while the guitarist plays. This is a great, natural Leslie speaker effect and costs exactly $0.00 to build. **EX_swirly1** and **EX_swirly2.**

- To increase that funky ambience on a track - especially for Jazz or Blues - set a snare drum in the room with the guitar amp. Loosen the snares a bit so that they sympathetically resonate with some notes. Lower notes tend to trigger it more, but the randomness of it is what makes it cool. If a track sounds sterile and needs some basement club vibe, this is a great way to get some authenticity in there. On this example ***(.wav EX_IWANNAHOLD)*** I went for a blowing-up speaker kind of sound in the chorus of a cover of "I wanna hold your hand". This creates a sense of intensity in the refrain, and clarity again when it drops out in the verse!

- Double track with a "Nashville-tuned" guitar. This sounds like a great, expensive 12-string, and is easy to do. Simply replace the low E-, A-, and D-strings with lighter gauge strings and tune them an octave higher. Try using the usual D-, G-, and B- strings from a spare pack of strings, those gauges are about right. Then, double the guitar riffs, chords and melodies in the song on this guitar. Listen back and see what you like.

- This one requires some guts, an affinity for Hitchcock films, and a spare speaker. Just take a steak knife and play Norman Bates on an old 12" guitar speaker. Stab that baby full of holes. Then, to make it more rock and roll, spill a little beer on it and leave it in the sun for an afternoon. Many guitarists have used this technique to get more distortion and an authentic “been-abused-in-lousy-clubs” tone. I’m gonna replace a guitar speaker soon, so I’ll stab up the old one and post a sample!

- If you're lucky enough to have a piano around, try this great natural reverb effect. Stick a mic down inside the piano, near the strings. Then put a weight on the sustain pedal to allow the piano strings to ring. Place the guitar amp near the piano and mic as usual. Print the piano mic track to a separate track and blend it in for a very airy, cool reverb sound. You may want to pan the 'reverb' to one side and the guitar to the other... works great on acoustic guitars and even vocals.

Drum microphone techniques

O.K., you may be able to slide by with crusty acoustic guitar strings, a beat up amp, or a boozy old singer, but drum heads really do sound like crap when they're too old. Since the drums play such an important role in the overall impression of a song's sound, this is the place to invest (or get the band to invest) before cutting tracks.

If you can't, won't, or just completely forget to get new heads for the kit, then I'll try to help you save the day later (with some great drum samples) - but this won't do much for your overhead or room mic tracks!

Start off with 'Ambassador' or 'Pin Stripe' heads for all the drums. Be sure to stretch them out well: Mount the heads, then place the drums on a mat on the floor, new head up. Get someone of average build (don't use a sumo wrestler for this!) to stand with one foot on the head to stretch it well. Don't worry; both drum and head can handle it, just no trampoline tricks... Giving the heads a stretch is as important as stretching out guitar or bass strings, if you don't they'll de-tune during the song and ruin the take!

Generally speaking you shouldn’t dampen the heads with tape, foam, or other material to reduce overtones, but rather tune them out of the drum. If there is an annoying ring to, let’s say, the floor tom, try to solve it with as little material as possible. Start with a little ring of duct tape, sticky side out. Steadily hit the drum and run your finger around the outside of the drum head to find the offending spot. Place the duct tape doughnut there and see if that helps. If you need more dampening, stick a piece of cloth to the doughnut. Don’t overdo it – those overtones are what give life and breadth to a drum sound once the mix fills with other instruments!

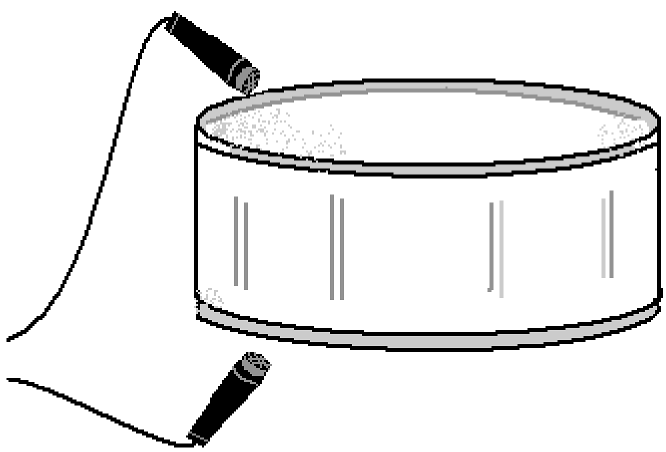
Also, since we are about to get into some serious multiple mic'ing keep in mind that phase is going to become a serious consideration:

when the same sound arrives at two different mics- one a fraction of a second later than the other – the resulting hollow "phased" sound can really suck the life out of your tracks. This is often present as a 'swishy' sound on cymbals, sometimes making them sound as if they fade in and out after being struck. We'll look at how to avoid these problems. This can be subtle at times, but is EXACTLY what this book is about - take care of twenty subtle problems and your mix goes from "O.K. for an amateur" to "Damn, dogk!"

Snare Drum Mic Technique

Aside from the lead vocal sound, and in some cases even more so, the snare drum is the most important individual instrument sound in a pop or rock mix. Producer Danny Kortchmar (with whom I have had the honor of working on many occasions) is absolutely strict about snare sounds in his productions - tracking doesn't start until the sound is knockin' him out. This can involve using amplifiers, odd drums, torturing gear... It doesn't matter how you get there, the important thing is that it sounds great.

The following diagram, using two Shure SM-57's is the standard, and fairly foolproof way of getting a good snare sound. Be sure that you phase-reverse the top mic – it moves away from the mic as the stick hits it – and should be reversed. The bottom mic will be O.K. then – in phase with the top.



Use the best preamp you have in your toolbox when on the snare drum. The best choice is something with a nice fat transformer in the path. A clinical sounding clean preamp can often sound too thin, even in "purist" recordings of jazz or acoustic music.

TRICK: So, you recorded a drum track, and later you realize that it would have been better to use a brass piccolo snare rather than that big old wooden hatbox that the drummer chose at the session. No problem – we'll switch it after the fact! Like this:

Set up a monitor speaker in the studio with the speaker facing the ceiling, and run your original snare drum track into it at a fairly loud volume. Then, set the new snare on top of the monitor speaker. Blast the old track into the new snare and re-mic the new snare track. The new snare drum will bang and resonate along, and re-track the old track very nicely! Be sure to record some room tone, too.

This method is not totally foolproof, but it does a nice job of following dynamics and ghost notes, unlike the sampling methods we will get into later. Try blending the re-tracked snare with the original if the re-tracked one doesn't have enough "oomph".

Kick Drum Mic Technique

A good solid kick drum is what you need to get your listener's feet a-stompin', so here's what you need to do:

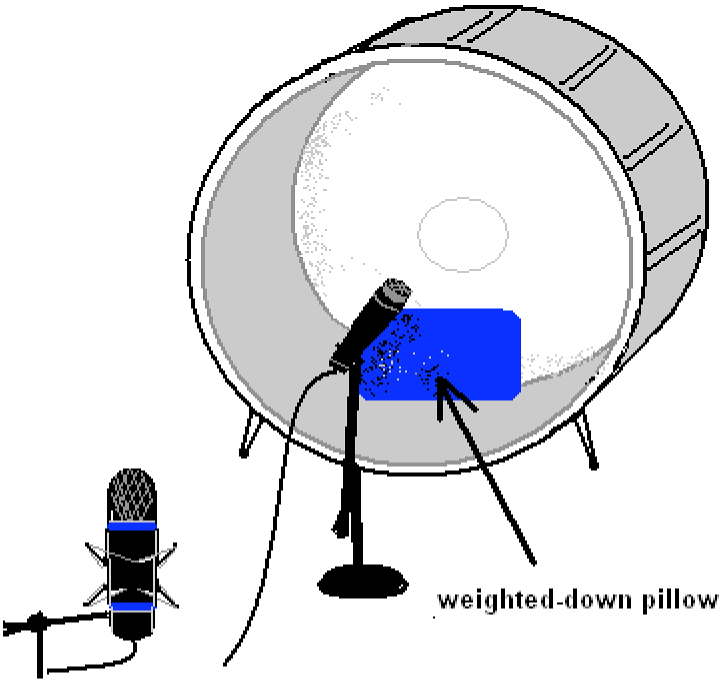
1) Make sure that the kick drum is tuned properly. You should look for a clear resonance and a strong fundamental tone. Some people remove the front head (more common for rock and pop) and some leave an uncut front head on (jazz and blues). The compromise is to cut a hole in the front head so that you can mic the beater head closely from the inside.

2) Good microphones for the inside (beater head) are the AKG D-12, EV RE-20, Shure Beta 52, or even the old Shure SM-57 (that's Chad Smith's kick drum sound on "Blood Sugar Sex Magic")

3) Add an ambient mic a few feet in front of the kick. A classic is a Neumann U67, but since very few of us have one, a large diaphragm condenser mic of any kind is good. Be sure to engage the -10dB pad. Ribbon mics are really nice and won't need a pad – just don't let blasts of air from the hole in the front head hit them!

4) For those of you who need that heavy metal "click" in the kick drum sound, get out your credit card. No, I don't mean that it's expensive, but that you should tape a credit card to the beater head where the beater hits. This will give you all the high end click you need, and is really easy to do. On second thought, use an OLD credit card. Or the singer's card.

5) Make sure the kick drum is firmly in place, so it doesn't slide across the floor during tracking. Now place a sandbag or a heavy pillow weighted with some stones or other weights against the bottom ¼ of the inside of the beater head. This weight and damping of the inside head will help you get a deep, solid tone.



An old studio secret for recording kick drum is to take a Yamaha NS-10 studio monitor (now hard to find) and place it in front of the kick drum, speaker facing the drum. Then the speaker wire is fed into a preamplifier and the speaker used like a microphone. I have to admit I've never tried this, but it sounds very interesting! You could use any speaker or monitor you like, you just need to solder a connector to get the speaker wire into the preamplifier.

Toms and individual cymbals

No big secrets here. Just a secret word of warning: Drummers hit things! Don't put your proud new Neumann microphones on those toms unless you want them to get a good whack with a stick. Sennheiser MD-421's are a durable, classic choice, and there are great little Audio Technica and Shure dynamic mics built for fitting into the tight spaces between toms.

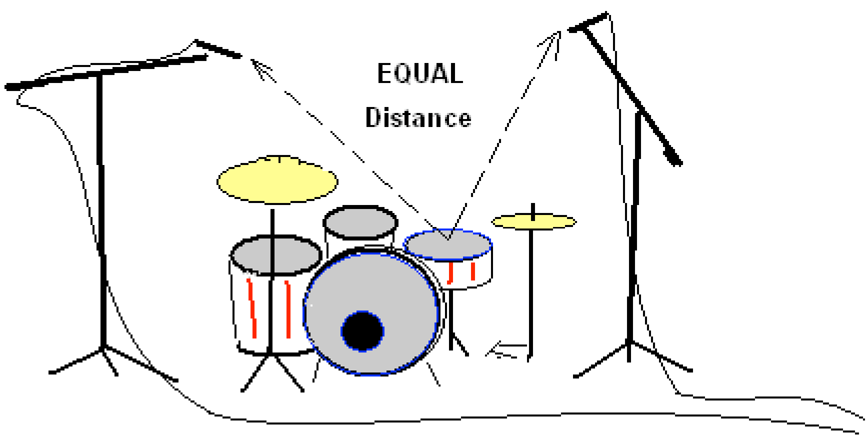
Some engineers will add an ambient mic (as with the kick drum) to the floor tom. A ribbon mic is also a great choice for this, although in most cases an extra mic is unnecessary.

Individual cymbals rarely need to be separately mic'ed - the overheads will do a fine job. I would, however, recommend a high hat mic in every case. A small diaphragm condenser is the right choice, any one will do. Just be sure that little blasts of air from the hats falling closed won't puff into the mic and cause distortion.

Overhead microphones

Here's where a nice pair of AKG-451's or Neumann KM184's will make a tremendous difference in your recording quality. I have a pair of Sennheiser MKH-40's which I prefer over any other mic. The overhead mics are going to pick up a stereo picture of the kit, a certain amount of ambience, and provide the clarity of the snare and cymbals.

Very important is, again, the way the snare will sound. An excellent technique for assuring a solid snare sound is to measure from the center of the snare to each overhead mic. Use a length of string, a tape measure, or just a short mic cable if that's all you have handy. Keeping the distance from the snare to each mic the same insures phase coherence for the snare drum relative to the two overheads.

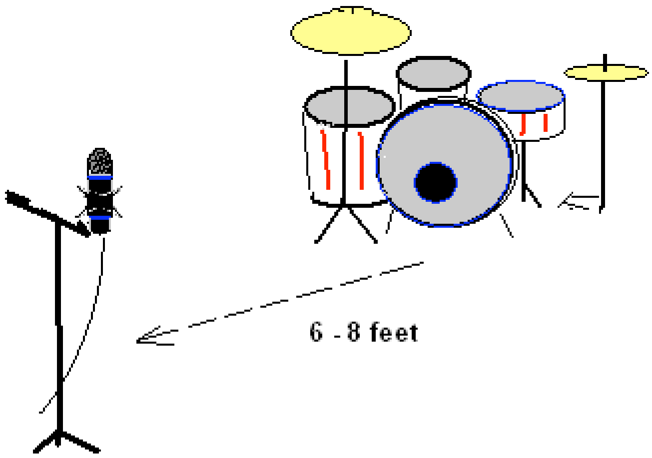


Otherwise, position the overheads within this measured distance to pick up an even blend of ride, high hat, and crash/splash/china/etc. cymbals. You should record a few minutes of drumming – requesting that the drummer plays all the cymbals at intervals - and listen back.

When does it end!? Room mics and front of kit

Now that you have all the individual pieces of the kit mic'ed up, you need to set up some mics to capture the room ambience. Even if you don't have an excellent drum room, give it a try – you may be surprised how a little room sound 'opens up' the sound. See 'triggering' below for a clever idea on handling the room sound.

Set up a large diaphragm condenser or a ribbon mic several feet in front of the kit. The best way to find a good spot is to walk around in front of the kit while the drummer plays. Listen carefully – when it sounds like the sounds really come into focus at a particular place, put a microphone right where your ear is. This is recording at it's most logical.



Room mics are most effective in a good room with proper acoustic treatment, but many garages, stairwells, and basements have good spots for some booming ambience. The most famous example is Andy Johns hanging a microphone a few floors above Led Zeppelin drummer John Bonham's head - in a stairwell. That sound became famous in the recording of "When the Levee Breaks" and has become a much imitated and sampled drum sound.

TIP: The 1:3 Rule – Generally speaking, the more microphones you use on a sound source, the higher the chance that things will start to sound bad. Phase cancellation, imbalances in tone quality, comb filtering, and other audio problems begin to creep in. In this respect, it is not wise to use an ambient kick mic, a front of kit mic, *and* room mics. You may wish to set them up to see which sounds best, but then remove your least favorite from the mix.
The 3:1 rule, however, states that as long as a microphone is THREE TIMES the distance away from the sound source as another microphone, you will almost surely have no odd audio issues. For example, if you use

a front of kit mic five feet away from the kit, your room mics should be FIFTEEN feet from the kit. If your room is only twelve feet long, then move the front of kit mic in to four feet or less!

To get your room mics cranking, set the stands up high, and set them close to the wall and near the ceiling for starts. Listen back to them with heavy compression – see the compression techniques section – and then try some different positions. Some engineers recommend facing the mic towards the wall, and as close to the wall as possible. You will need to experiment and get to know your recording space. In case you are recording in someone else's studio, ask a staff engineer where they have had luck. Just remember to compress that sound!

Samples, triggering and sound relacement

Now that you've gotten your good drums sounds the hard way, you get to learn how the pros *really* do it. They cheat. Yes, they do mic things carefully and use great gear and get super sounds, but then they cheat – and you can too!

With a DAW, you can look right in close at every drum hit you've recorded. You can then create an extra track and line it up right next to, let's say, your snare drum track. Then, you can download one of the great recorded snare sounds that I have provided and line it up with the snare sound you recorded. Now play it back and BOOM! Your snare sound has been pumped up to commercial record size. Here's a track where we recorded the drums in a friend's living room because of the nice wood floors (there's no added reverb on the drums - it's natural) ***(.wav file EX_DRUMS_W_SAMPLE).*** The ambience and overheads were nice, but the kick and snare were layered with samples because they just didn't 'snap' in the mix.

This can be done with any drum hit – kicks, snares, toms, etc. Even cymbals! Sometimes you may wish to add several samples – one with a crispy high end, another with deep, clear lows, and so on. You can even sample doors slamming, handclaps, phonebooks ripping in half, whatever your devious little mind desires.

Here are some great drum samples for you from my collection. ***(.wav files SNARE1, SNARE_CRISP, SNARE_HI1, SNAREHI_2, SNARE_HI3, SNARE_LONG, SNARE_LO1, SNARE_LO2, SNARE_REBOUNDS, SNARE_SHORT, KICK1, KICK2, 808KICK, KICK_CRUNCHY1, KICK_CRUNCHY2, KICK_REBOUNDS,***

***KICK_ROUNDSOUND*, and *KICK_SOFT*.** You may want to download the zipped files for these.

The really good news is that there are a variety of plug-ins available now which can do the tedious sound replacement work for you. You should still look carefully at the replaced track in comparison to your original to be sure that the replacer plug-in has done an accurate job.

Triggering a sound from a drum hit is an old method for juicing up kits. Time honoured methods are the 60Hz kick drum trigger and the snare drum white noise trigger.

The kick drum 60Hz trigger goes like this: You need a track of a 60Hz sine wave. This track should have a gate applied to it. Send a copy of your kick drum to the gate trigger input. When the kick drum hits, the 60Hz signal booms through. Set the gate release time to adjust the length of low end sine wave boom.

The snare drum white noise principal is exactly the same, you just substitute the 60Hz sine wave for white noise, then trigger the gate with the snare drum.

Have fun with the trigger idea – you can also set sounds to trigger off the high hat, toms, whatever you like. You may wish to trigger the room sound off the kick and/or snare drum if your room sound isn't top-notch – this will keep the room sound under control, and you can use the gate hold time to control the length of it. You could trigger a guitar track off the high hat for a tremolo-style effect. You could blow the raspberries into a microphone, loop it up, and trigger it off the toms for a fart stampede. Sky's the limit.

Loops are another way to bring some variation and dimension into your production. There are some great discs available with well produced hip-hop, rap, house, and other funky beat material which can really add to a song. I often use a loop for writing purposes, and then later build some fills from individual hits or have a live drummer perform to the loop.

I would recommend "Loopzilla", The "L.A. Riots" discs (BIG samples), "Vinylistics" (the ultime old school sounds), and for some real great kit samples "Drum Works" and "The drummers of Motown" CD's. I've also had luck creating my own loops by mixing samples with my own performances and then cutting the loops into usable 1 or 2 bar sections.

When "cutting" your loop, be sure to identify the beginning of the first downbeat and the end of the loop very clearly in your editor. Listen carefully to multiple repeats of the loop and make sure it grooves. You will often need to make a reference bar length (in samples) from the main loop you wish to use. I have often found that loops that are supposedly the same tempo and length can vary conisiderably. Always make one loop the master reference and relate all bar lengths to it. In this way you are assured a consistent tempo throughout the song and won't have problems with drifting beats and loops.

And, finally, I should mention the programs like "EZDrummer" and "BFD". These programs are essentially sample libraries of very well recorded drums set up so you may play them like a MIDI instrument. There are many layers of velocity (that is to say, the drums sound very natural because they were sampled many times at different 'hit strengths'), and a large selection of sounds.

I'm a convert, I admit it. I love tracking real drums, and you can't get as personalized of a performance with BFD as you can with a drummer, but the time saving - and tone saving - factor is undeniable. We'll listen to some ways to process the BFD sounds after we explore processing. They can get reeeeeeeeealy good!

Equalization

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Right there in the word equalization is the word 'equal'. Although engineers have developed numerous wild tricks with equalizers through the years, the most basic and important concept is 'to make things equal' or 'to balance the sound'.

Let's imagine that you are an engineer in the early days of recording. That famous rockabilly band just laid down a real foot-stomper of a take. But while listening to the playback of the ONE microphone on the ONE speaker, everyone agrees that the bass is just too loud.

"Well, ya'll," the singer says, "Ah ain't never singed it that good before, an' Ah don' think Ah kin sing it that good agin!"

"So, son," The producer looks over at you skeptically, "Kin yew balance out that sound, sorta eequahlize the instrahments?"

Nodding, you turn to the two story high equalizer-machine at the side of the room, dial in 100Hz, and begin cranking down the level one dB at a time until the producer exclaims,

"Well, gaw-lee! It's just like black magic!"

And the band members turn and level a suspicious stare at you...

Yes, that is what it's all about – having a door at both sides of the studio; *equal* chances of escape from dangerous superstitious musicians.

No, wait, it *is* about balancing sound. In the example above, the bass frequencies were *made equal* with the others, giving the impression of a more natural and balanced mix.

Fast forward to the days of multi-track recording, and we now multiply the ability to readjust – to equalize the sounds - by however many tracks we have! What a great thing! I mean, how can there possibly be a bad sounding recording made anymore?

I ask myself that question all the time, and the answer is that the power to *change* something doesn't necessarily come packaged with the ability to change it for the *better*. On that note, I'm going to walk you through some typical and some not-so-typical audio situations and give some useful hints, secrets and examples on how to make your recordings sound much better through the use of equalization.

The ropes

Before we start twisting knobs, let's keep our eyes on the big picture. Unless your recording is of a solo instrument, you need to make a sonic place for every instrument in the mix. As a hobby chef, I like to think about cooking a great meal when EQ'ing instruments and setting them into a mix.

The first thing every chef knows is that the raw ingredients have to be great to start with. Wilted lettuce cannot be saved with dressing. No matter how much gravy you dump on that dry turkey breast, it is still going to have poor texture and taste. So make sure your tracks – your raw ingredients – are as tasty as you can make them.

Of course, all chefs *also* know that ***the secret is in the sauce***. Asparagus, new potatoes and thinly sliced Black Forest Ham is definitely improved with a ladle of hollandaise sauce! Just like your well played, well mic'ed drum tracks will sparkle to life with the right EQ settings.

Most importantly, as the chef puts together the meal, a balance is created between salty, sweet, sour, and even a little bitter to create a full taste experience. Yes, we may like sweet dishes, but if every ingredient of every course of the meal is sweet, then the meal is one dimensional and unsatisfying at best.

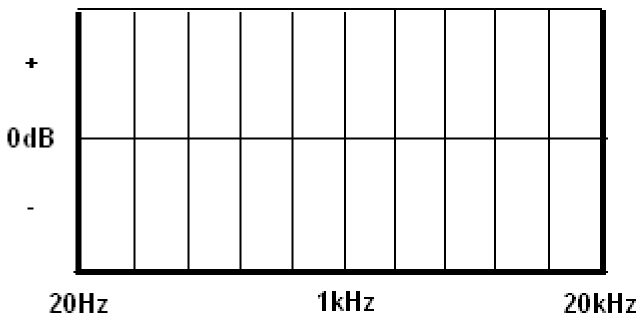
Again, the same goes for your mix. Bass/treble, warm/clear, and full spectrum/midrangey and filtered – these are some of the 'flavor' counterparts of the EQ world. You can't make all the instruments fat and bassy, or your mix will sound the way it feels to eat a bowl of pork rinds in melted butter. My former college roommate can be contacted for details on that feeling. gorillatickler@yahoo.com

You will need to understand the basic types and parameters of EQ. There are plenty of propeller-headed names for the various types of devices and filters, but all the names do is describe *how* the internal circuits boost or cut certain parts of the frequency spectrum in order to alter the sound.

Sound is energy transmitted through the air like waves transmit energy through water. Sounds are composed of a combination of different frequencies, measured in Hertz (cycles per second). Some sounds are fairly simple, like a single note blown on a flute, and others very complex, like the sound of a drum.

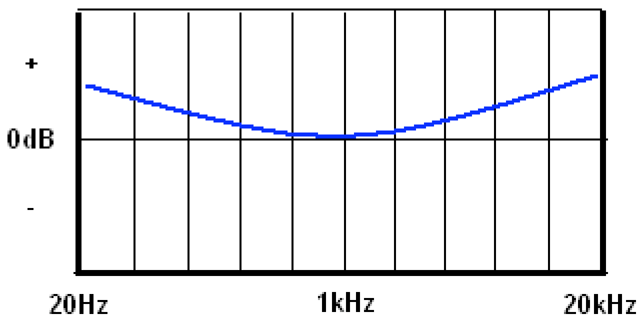
We can hear deep bass frequencies (20Hz) all the way up to (what is for human ears)high treble, although some people argue that we can ‘feel’ even lower and higher frequencies. I believe that to be true to a certain extent as well, but we must accept certain limitations of our sound gear, speakers, etc. and at some point say, “O.K., 20Hz – 20kHz is plenty for recorded music, let’s just get along and record something!”

The spectrum of sound is often represented on a graph like this, with low frequencies on the left and high on the right. The horizontal center line is the reference level for ‘no change’ between what comes in and what goes out:

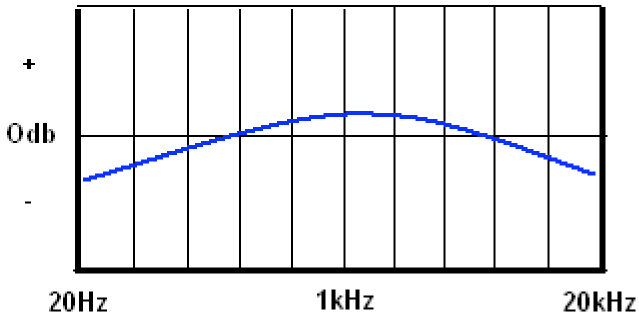


A line is then indicated on the graph to indicate the frequency response of a device like a microphone or equalizer. What the line is telling you is that “If you were to put a signal through me, this is how I would affect it.”

To better explain this, let’s imagine that you turn up the bass and treble on your home stereo to make it sound ‘louder and better’. The difference between what goes into that EQ and what comes out would be somewhat like this:



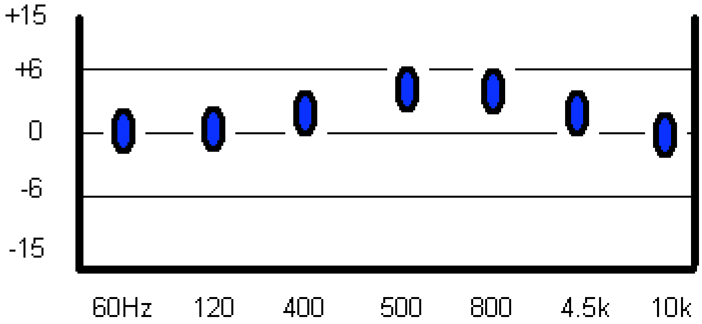
Another way to look at this might be that you have a microphone that accentuates mid-range frequencies. The difference between what goes in and what comes out would be this:



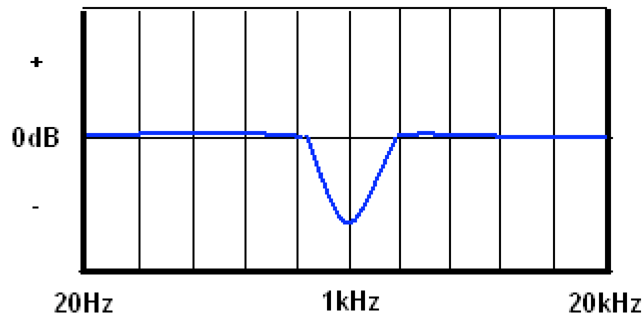
So the last thing to understand is what types of devices we use to equalize. There are the ***bandpass and shelving EQs, graphic EQs.,*** and ***parametric EQs.*** Keep in mind that Esq.. are also referred to as 'filters'.

Shelving and bandpass filters operate just like the 'low', 'mid', and 'high' knobs on a stereo receiver. The low shelving filter boosts or reduces only sounds ***below*** a fixed frequency (usually about 100Hz). The high shelving filter boosts or reduces only sounds ***above*** a fixed frequency (about 10kHz). A bandpass filter boosts or cut a middle range band of frequencies within a certain range (about 500 – 5kHz or so).

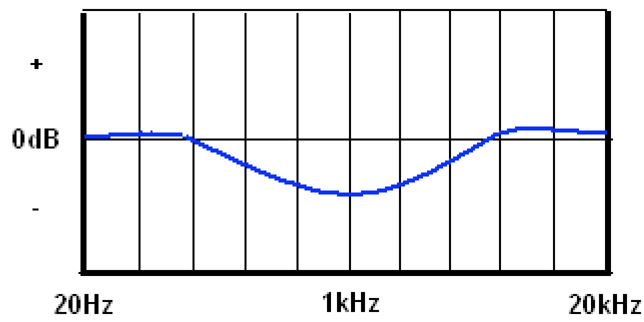
Graphic EQ is also familiar to us all. It is the type of EQ with all the little 'sliders'. The whole frequency spectrum is broken up into bands (a small range of frequencies). There can be as few as 5 or as many as 31+ bands in a graphic EQ. Graphic Esq. offer a little more flexibility in tailoring the frequency response of the EQ circuit and therefore shaping sound. Like this:



Parametric EQ offers the engineer the greatest flexibility. For each filter on the parametric EQ (there are often several bands), adjustments may be made to the frequency affected, the amount of boost or cut and the bandwidth (or ‘Q’) of the filter. Setting the the ‘Q’ adjustment to a narrow setting allows us to focus in tightly on a particular frequency, like this:



We may also set the ‘Q’ wider, to cover a broad band of frequencies:



Adjustments to the ‘Q’ are continuously variable between the widest and narrowest settings, offering a tremendous range of options.

The advantage of the ‘Q’ adjustment is in the flexibility this setting offers us when sculpting a sound. We can boost or cut only a very narrow frequency range to emphasize a particular quality of a sound or cut out a problem frequency. We can also set the ‘Q’ to wider sounds for a broad range adjustment to tailor the overall frequency response of a sound.

Before you reach for the EQ knobs, doctor, be sure that the patient is at the right level in the mix. If you start saying, “It needs a little more treble. Good. Now a little more bass. Uh oh! Now the midrange needs a boost. There! – perfect!”, then you should shut off the EQ and adjust your level. No unnecessary surgery, please.

Then, when you think you are ready to EQ, follow the next few rules:

- **Cut rather than boost.** Look for a problem you can equalize down, rather than something you wish to increase. This will sound most natural and lessen the chance that you will hear harsh artifacts of the EQ circuit.
- **Limit your EQing to about 6dB of gain change.** It is easy to start zeroing in on frequencies and cranking them up or down, but this often leads you down a dead end. If you have paid attention to good microphone and tracking technique, you will rarely need more than a few dB of adjustment. Bad mixes occur because they are built on a foundation of overprocessed tracks – nothing sounds natural.
- **Be careful of high 'Q' settings.** Unless you are in a very well acoustically treated monitoring environment it is possible that you are not hearing certain frequency bands accurately, because of room acoustics. High 'Q' settings, especially when boosted, can sound nasty when listening later in other environments.
- **Always A/B your EQed sound with the original.** A/B is a term we often use meaning "compare 'A' – the unprocessed sound - with 'B', the processed version." You will often find you need to make less adjustment to keep the sound natural, a little more to fix it, or even that you need no EQ after all! This is very important, and the best engineers A/B a sound both on it's own and A/B it in the mix until they are happy. This is a good habit and will save you having to later undo a whole house of cards built on poorly EQ'ed tracks.
- **Audition different EQ's.** The difference in sound quality between EQ units varies widely. For example, the ProTools EQ 3 is quite clean and clear, whereas the funkier JoeMeek and Steve Massey (http://www.masseyplugins.com/) EQ's add a whole new dimension to your sound.

BIG HINT: **In order to an offending frequency**, set the 'Q' fairly narrow, set the gain for about +6dB, and then sweep the frequency through the range where you suspect the nastiness to be. Go slowly while the sound is playing back and you will find a point where it sounds worse. Now, cut that frequency by 2dB or so, and widen the 'Q' a bit. Listen again with the EQ bypassed and the again with it on. Does this help? Does it still sound natural? If yes, then you have done well. If not, start again on your search for the right frequency – this takes some practice. I call this the "boost and sweep" method, and it is the standard way among many, many engineers for determining how to EQ a sound. Practice it and you'll also become a pro EQ'er.

Vocal EQ

We are intimately familiarity with the human voice – we hear it , for better or for worse, every day of our lives. We are all good judges of what sounds natural, and are quick to notice when something sounds strange.

Strange vocal sounds are therefore instant attention getters. But let's start with a natural sound.

Simply using a mic with a fairly flat frequency response and not messing with the sound any further attains the most natural vocal sound. This would be proper for **acoustic music, jazz and definitely for classical performances**. If the sound needs a little more 'air' on top, a few dB boost with a high shelving filter set to 12kHz will open up the top end a bit without sounding unnatural.

A **pop or rock vocal** has different requirements to get it to fit into the mix properly. First be sure you have a good level in the mix! Consider the interplay between the vocal and other instruments, especially things like buzzing distorted guitars or 10-finger piano parts. Hopefully the song is well arranged...

Before you start EQ'ing that vocal, is it possible to make space for the vocal by cooling out some frequencies in other instruments? The most natural vocal sound is to not EQ it too much other than a hint of high end, so consider that twice!

Now the singer's personal voice character should come into consideration. Are they a bit nasal sounding? In that case, try a little cut in the 1kHz to 4kHz range.

If the singer sounds unclear, you may try a boost in the above mentioned frequency range.

In general, you are going to find that narrow 'Q' cuts or boosts to a vocal will sound unnatural. If a vocal sounds out of place in the track in general, you may wish to try a very wide 'Q' cut or boost. Sometimes a wide midrange boost can give a vocal enough bite to ride on top of a big rock song. You may want a wide, warm low end boost on an intimate jazz track. Be sure to keep these adjustments within just a few dB or run the risk of skewing the whole mix into sounding odd and unprofessional.

Background vocals should be kept both at a level and an EQ setting that makes them serve the lead, not overpower it. If background vocals cut through too much, even at lower levels, try dipping out a little in the

1kHz-5kHz range. This will make them ‘softer’ and ‘breathier’, and set them back in the mix while retaining fullness of sound.

You can also have fun with vocals by filtering off all the lows and highs and leaving only the midrange. “Breaker one – nine! Breaker one – nine!” Run this sound through an amplifier or amp simulator and you’re on the radio from somewhere out on the lonely highway... You can also do this to one track of a doubled vocal to add an attention-getting resonance to the mix.

Most importantly, compare your work to commercial (or your favorite – it doesn’t have to be commercial) CD’s. Find a recording in a similar style with a similar sounding singer and A/B your recording with the other. Make general adjustments from there. Be sure to listen and try to discern EQ curve from room ambience, reverb and/or compression effects so that you don’t get confused. This also takes practice.

Acoustic guitar EQ

Acoustic guitars often need a bit of help. Not every guitarist out there is playing a freshly strung Martin, Gibson, Taylor or other $2000.00 + naturally good sounding instrument.

Typically, acoustic guitars need a boost in the high treble (12 kHz) to create a larger-than-life sense of clarity and sparkle that has become recording convention. Alternatively, sweep a parametric band up to 20kHz, set to a Q of 1.0 and boost away (+3 to +6dB or more) – this creates an increasing rise all the way up the audible frequency range.

The next typical adjustment is in the low-mid frequency range. Again, the better the guitar and acoustic environment, the less this is necessary. If there is a ‘muddy’ or ‘cardboard’ sound to the instrument, set a parametric EQ to 200Hz with a 6dB boost and a somewhat narrow ‘Q’ (maybe 1.5 or so). Now sweep slowly up thru 600Hz while listening to the recording. At some point the sound will be swallowed up in that ‘muddiness’. Stop there and reduce the gain to -3dB or so and do some A/B listening.

700 – 1.5kHz can contain a sort of plastic, dull sound on acoustic guitars. Try a little cut here if you’re hearing a cheap, artificial sound. Always A/B it!

Be cautious of the area from 2 – 10 kHz. I have found that you are often better served re-cutting a track with a different mic or guitar than monkeying around heavily in this frequency range with more than a slight, wide cut or boost. You tend to lose the sound of certain notes or chords rather than balance the instrument. You must do some trial and error work to learn the ropes.

A final trick I like on acoustic guitars is to set a high pass filter (a steep 'Q' shelving EQ) to cut off all frequencies below 40Hz. Sometimes you can get a mandolin like sound by placing a capo on the neck and cutting up to 120Hz or more on the low cut filter. It doesn't sound completely natural, but can often compliment a second, natural sounding track or just 'tighten up' the overall sound by leaving the lowest low frequencies to the drums and bass.

Let's hear the sound of the ProTools EQ3 versus the wonderful Steve Massey EQ. Neither is 'better', you would need to see how it fits into your mix to make the choice. Listen to my favorite rockabilly riff again with a 3dB low shelving and 3dB high shelving boost through the ProTools EQ (first) and through the Massey EQ (second). ***(.wav file EX34.)***

Electric guitar EQ

An entire manual could be written on the art of EQing electric guitar. There are simply so many possible sounds, from clean sparkling sounds to rabid distortion and everything in between. So what I am going to do is make a subjective list of impressions for particular frequency ranges. Hopefully this will help you focus into areas you wish to cut or emphasize. Keep in mind that the ranges are approximate and will vary slightly with the room, mic, guitar, amp and song!

Afterwards we'll look at a few tricks for some special sounds.

20Hz – 60Hz: Generally, electric guitars and amps do not produce frequencies this low. These frequencies may be cut down if there is any 'rumble' or 'boominess', especially in the room sound of an electric guitar recording. Some larger amps may, however respond to pick attack and special guitar tricks with some low frequencies that go this low.
70Hz – 100Hz: Cut here if the guitar sounds flabby or boomy. Add some here if you need fullness.
120Hz – 200Hz: These are the fundamental frequencies of the lower strings. Many instruments contain a lot of energy in this area, so you may need to focus in on particular regions of this band and boost or

cut a little in different frequencies for instruments such as snare, guitar, piano, and vocals. Often, you should just leave this area alone!

200Hz – 500Hz: This is the famous 'mud' area. Use the technique from the section on acoustic guitar EQ to identify and reduce problem frequencies.

600Hz – 1kHz: 'Cardboard' or 'Honk'. Sometimes you want a boost here for that 'Queen' sound, or to bring the guitar forward in the mix. Other times you want to cut back in this range, especially for more 'chunk' in heavy metal guitars.

2kHz – 5kHz: This range contains a lot of the attack of the guitar sound. Cut or boost as you need to settle things into the mix – after you double check the overall volume level in the mix, of course.

6kHz – 10kHz: Contains the brightness of the sound, but can be 'harsh'. Cutting here can make a high solo smoother sounding.

10kHz – 16kHz: Most guitar amps do not produce frequencies much above 10kHz. If an amp is 'hissy' or there is high frequency noise, you can usually safely cut back here to reduce it a bit. If the amp was nice and quiet, you can boost a bit to add some 'air', especially to clean guitar sounds.

16kHz – 20kHz: Not much going on way up here. Play around and see if you find anything!

Some tricks and hints:

- If you are creating a mix with many different guitar parts, consider creating an EQ 'zone' for each part based on the range it is performing in. For example, cut the lows and some highs off of a middle-range lick that repeats often. Then, cut some of that remaining middle range out of a complimentary part.
- Dip out a lot of energy in the 1-2kHz range to get a 'larger' sound out of amps that were played through small speaker cabinets. Listen to this riff recorded through my trusty Vox Pathfinder 10 Watt. First, just as it was, next with a dip in the mids ***(.wav file EX35a and EX35b)***.
- EQ the guitar signal ***before*** it goes to the amplifier! This drives the amplifier in an entirely different way than EQing the mic'ed signal later. This is especially effective for getting chunky, midrange sounds which would sound too harsh if EQ'ed after the amplifier.
- A very wide mid range boost (Q of 0.5, frequency 1kHz, +2dB) can be just the ticket for a rock guitar sound. Many modern amps have a midrange cut built in, and it can be refreshing to hear more of that range in a mix.

Bass guitar EQ

Most good bass guitars require very little EQ. What is very important is that the bass is played at the proper dynamic for the track, that is, not too softly on a loud song or vice versa. The harder a bass is plucked, the more high frequency attack tends to jump out. This is true of both electric and acoustic bass and is strongly exaggerated if the string slaps back against the fingerboard.

The most important frequencies to consider are:

50Hz: This is generally the lowest fundamental tone produced by the bass, although tuning a bass down lowers the fundamental. If there isn't enough 'oomph' in the bass, try boosting here
100 – 200Hz: Also an important bass range, as many consumer boom boxes, etc. do not reproduce 50Hz signals very well! You will want to balance out this frequency range so that the bass is not too sub-bass for the average stereo.
200 – 500Hz: Again, the muddy range. It can also be very important in giving a feeling of 'body' to the instrument esp. for jazz bass sounds, so be careful about cutting too much.
700 – 3kHz: This range contains a lot of the attack and plucking string sounds of the bass. Very important for slap styles.
5kHz +: the sound of fingers on the strings and the harmonic content of higher notes lies here. If you are using a shelving EQ, you can set the shelf frequency much lower than you would for most other instruments since a bass will rarely sound 'harsh'.

Acoustic bass tends to produce stronger fundamental frequencies than electric bass due to the increased string length and body size. There is a lot of 'feel' down around 40Hz on the low notes of acoustic bass!

For electric bass, you may wish to blend a mic signal with the direct signal (check your phase!) and use the strong points of each signal to create a round sound. For instance, accentuate the clean, clear low end of the direct signal and boost some midrange on the amp signal for string attack and general vibe. Remember, you can't boost what isn't there, and since the amplifier/speaker duo acts as a *bandpass filter* (lowest lows and highest highs just aren't reproduced) you may need that direct signal for the 60Hz sounds.

Strings, piano, and classical instruments

I have lumped together a few instruments here into a sort of 'leave it alone' category. When dealing with classical instruments ***in a classical setting*** (piano, flute, oboe, clarinet, violin, viola, cello, etc. – think orchestra) you should use EQ only to correct acoustic problems with your recording space, such as a resonance at 300Hz, for example.

Piano has the widest range of any instrument, and if the performer uses a lot of this range, as in a piano concerto, cutting/boosting EQ will sound more like the performer played some notes softer/louder, rather than actually affecting overall tone.

With woodwinds and strings, the sound signal is relatively simple and EQ tends to disturb dynamics more than tone as well.

With that said, a pop or rock context is another scenario entirely!

Strings may sound great if they are high and low pass filtered and distorted, or woodwinds (think 'Strawberry Fields' where the Mellotron fake woodwinds sound so nice and crunchy). Also, the typical 'muddy' 300 – 600Hz range on other instruments sounds lush and full in strings. What a nice way to arrange something into that spot in the EQ spectrum...

A piano can be turned Honky-tonk by filtering some highs and lows off, adding a touch of chorus and a few random, high-Q parametric EQ dips for those missing keys...

A rock and roll piano can benefit equally well from the kind of wide midrange boost noted above for guitars. 1kHz, Q 0.5, +2 or 3dB.

Recording orchestral instruments can be a challenge in the small studio mostly due to the limitations of the acoustic environment. Orchestral instruments require a bit of distance from source to listener (mic) for the sounds to develop, especially true for low frequency instruments. Try to back the mic off 10 feet or so if you can, and try different mikes to see which pickup pattern provides both a realistic picture of the instrument while suppressing unwanted ambience. Then you can adjust your EQ a bit to reduce bad resonances. Hey, give it a try and then experiment, what have you got to lose?

Drum EQ

Now it's time to play with the grownups. We've learned to mic the drum set properly. We've recorded beautiful, solid tracks of well tuned and well played drums (right?) Now it's time to make 'em shine.

Because of the rich frequency content of most drum sounds, we have a lot of leeway in sculpting the sound with EQ. From full and tubby to thin and papery and everything in between, drums take very well to EQ manipulation.

In a jazz, blues, or orchestral setting a natural unaltered sound is often the most fitting. So, don't do 'too much'.

For rock and pop we want bigger than life, and 'too much' can be great...

Let's take the kit one piece at a time:

Snare

The snare plays a central role in pop and rock – cracking out that infectious backbeat – and needs to sound as good as it can to insure a happenin' mix.

The tricky thing with a snare sound is getting a balance between the low end 'thud' and impact, enough midrange 'body', and some crispy high end for definition. To tell the truth, most major recordings use ***layered samples*** of snare drums (also other pieces of the kit) to get a bigger-than-life sound (as discussed earlier) This is done by finding each snare hit in your DAW's editor window, then creating some new tracks and lining up other recorded snare drum samples (onto the new tracks) with your recorded snare track. A friend of mine recently did some work for Matchbox 20, and used ***four snare samples*** in total to get the sound they wanted.

Typically in such big productions, you get the original snare sounding nice and full. Then blend in a sample of a snare with a lot of low end punch, for more impact. Then add some nice crackin', snappin' high end sample for that extra 'cut' and brightness. The blend should serve the mix, not only sound good on it's own.

Even if you decide to go the sample-layering route, here are some EQ ideas to get your original snare track sounding as good as it can:

20Hz – 50Hz: Unless it is a very large snare tuned fairly low, there isn't much down here to boost; you would likely wind up boosting bleed from the kick drum.
60Hz – 100Hz: This is the range with the deep 'oomph'. Try sweeping a 6dB EQ bump ('Q' of 2) through this range until you find the sweet spot, then reduce the boost to 3dB or so and adjust your 'Q'.
120Hz – 200Hz: Here lies the punch and impact of the sound. Try the same trick as above. A little boost in this range insures good translation to low-fi systems.
200Hz – 500Hz: Although this is typically the muddy area on other instruments, be cautious of cutting too much here on snare drums, lest you lose 'body'. Try it and see, that's the best way to hear what I mean by 'body'.
600Hz – 1kHz: This range can be tin-can and annoying. You may, however, need a boost here if the drummer plays a large, snare tuned deep. Try and listen.
2kHz – 8kHz: Here is the crisp and crack to the drum, where the sound of the snares comes out. Use the boost and sweep method to search for good and bad and even out the sound if needed.
10kHz – 20kHz: Add here for more clarity and presence. Too much boost will make the sound 'papery'.

Dirty trick: With all the great amp-simulator toys and plug-ins out there, consider beefing up a wimpy snare sound by running a copy of it (or an aux send) through a virtual amplifier. Flip through the amp models while listening to your mix until the sound strikes you. Be sure to watch the blend between real and amped to keep it in check. You may need to use your editor to re-align the processed track with your original to compensate for processing latency.

Kick drum EQ

Equally important in your mix is the big old bass drum. This is the backbone of the downbeat and must be solid and deep as well as clearly defined.

The same sample layering tricks are often used on bass drum, especially where very deep, clear tones are desired. The old **Roland '808'** kick drum sample has been a staple of rap and hip hop for decades, and everything from car doors slamming to phone books dropped on floors have been recorded and mixed in with bass drums on commercial records. These are ideas you can easily experiment with on your DAW at home.

The basic tone of the kick drum is very style dependent, as covered in the section on microphone placement. Now let's look at some style based EQ tips.

20Hz – 50Hz: Some engineers radically boost these ultra low frequencies, some either ignore or even cut them. You will have to experiment, as this depends heavily on the instrument.
60Hz – 120Hz: Use the boost and sweep method to find the fundamental of your drum and give it at least a 3dB boost. This is standard for most music styles.
120Hz – 200Hz: A little bump here can give the drum some more punch, but can interfere with bass or guitars, so be sure to consider the EQ of your other instruments.
200Hz – 400Hz: A judicious cut in this range is important for pop and rock. Boost, sweep, find, and destroy. Don't do that for jazz and be cautious for blues, some of that muddiness can add to a swampy blues vibe.
500Hz – 1kHz: If the kick sounds 'flabby', look for a place to cut in this range, but be careful not to remove all of the sound of the wood.
2kHz – 5kHz: This is where the 'slap' of the beater lives. Don't adjust for jazz or blues. For pop and rock you'll want a little boost. Use a lot of boost at a narrow 'Q' (1.5 – 2 or more) for metal and punk, to help cut past all those guitars.
6kHz – 10kHz: There is some clarity and definition in this area, but be careful of too much boost, as you can get 'springy', twangy overtones in this range.
10kHz +: Quite often this range contains bleed-in from cymbals. Probably best left alone, but if cymbal bleed is bad, you can cut here to reduce it.

Don't forget the 'credit card on the beater head' trick from the mic technique section. This gets extra 2-5kHz frequency content into the microphone so that you don't need too much EQ boost when mixing, which can bring up bleed from other drums/cymbals. This technique is crucial for recording punk/metal.

Try the amp-simulator method (as described for snare) for the kick as well. It can really bring out the impact and snap of the kick drum if needed.

Overheads/Cymbal EQ

Overhead and cymbal microphones often pick up a fair amount of room ambience, so you will need to keep an ear out for problem resonances from the very beginning.

Here are some very subjective descriptions of the sound of some frequency ranges for cymbals. Boost or cut within these ranges to increase or decrease that timbre. Sometimes a narrow 'Q' setting can pull down an annoying sizzle or ringing tone without affecting neighboring frequencies. Use your ears and bypass the EQ often and listen to be sure that you are not over-processing.

Also be sure that you don't mix cymbals too loud. High hat is often needed to 'drive the track', but listen to how low crash cymbal are usually mixed on commercial records.

20Hz – 60Hz: Not much here. You can safely filter out these frequencies to get rid of room muddiness and low end bleed.
70Hz - 120Hz: Not much in the cymbals, but the toms have some important low end here, so check out what you want to add to the toms, like a little 100Hz bump.
200Hz – 500Hz: 'gong' (lower) and 'clang' (higher)
600Hz – 1kHz: 'annoying' in cymbals, tubbiness in toms. A little cut can be good here. Just don't loose too much 'wood' or 'body' from the kit by cutting too much.
2kHz – 10kHz: contains both the sizzle and the hardness of the cymbals as well as the sound of sticks striking and tom drum heads. If there is harshness anywhere, use the bump and sweep method to search and cut the offending frequencies. Don't overdo the cut and make sure your cymbals aren't mixed too high.
10kHz – 20kHz: The high frequency sparkle and clarity of cymbals. A little high shelving boost is often nice, but don't make 'em sizzle like bacon grease.

It can be difficult to get a smooth and clear cymbal sound without screwing up the toms, so please make sure your cymbals are mixed low enough. Always compare your mix to others! Then, when a tom fill comes along, use your DAW's volume automation to boost the tom fill. In the case of a regular mixing board, get the drummer to ride the fader up – and then back down...

Frank Zappa used a suave technique when mixing drums. He would make a submix of the toms and cymbals, then send a copy of that signal to two separate stereo mix busses. One bus would then be EQ'ed to boost some high midrange and treble, the other bus would get a fat low end boost and then be run through a compressor. This way, Frank could easily blend the 'crack' of the high end with the 'fat' of the compressed low end. Making a general mix adjustment to the toms and overheads was a breeze.

I developed my own take on this idea by creating a "crunchy" submix. With the popularity of hip hop and drum n' bass sounds, adding an element of lo-fi to your drum sound can fatten the mix and provide another mix possibility for creating breaks, fills and drops. Use an aux send or a buss to get a copy of the drum mix to a stereo channel (or a pair of channels). Then, set up a stereo EQ and a stereo compressor. On the EQ, set a high pass filter to 60Hz with a very steep cutoff. Cutoff filters cause a 'bump' – a resonance – at the cutoff frequency. In this case, a cheap EQ (or an older plug-in) may work better because this resonance will be more pronounced. Do the same for the high frequencies. Cut everything above 8 or 10 kHz. The lower the cutoff, the more the sound will seem crunchy and 'lo-fi'. Now give a hefty boost at 100Hz as well. This will give you the 'thump'. Now, pack the whole EQ'ed mess into a compresser and smash it to hell. Adjust your attack to let the 'thump of the kick drum through'. Back off on threshold if you feel that the dynamics get lost. Listen to this example to hear it in action:

We'll look at how to set up these techniques on a DAW like Pro Tools in the "Mixing" chapter.

Individual Toms EQ

The sound of the toms will vary greatly depending on their size, from dinky six inch Roto-toms, to massive witch's-cauldron floor toms with a fundamental frequency that can interrupt whale migrations.

Proper tuning and drum tone comes first. Toms can sound pretty bad if you didn't do your homework, so make sure you spend time getting you sound right before you track!

Look for a balance of attack, fundamental tone sustain, and even the reaction of the snare drum's snares for some rattle and sizzle. Make sure not to exaggerate the low end so much that the mix suffers from huge, syncopated bursts of low end energy at every change in the song.

20Hz – 50Hz: Floor tom territory, usually in the room mikes, when the long low frequency waves have had time to spread out. Don't overdo it.
60Hz – 200Hz: Fundamental frequencies of various toms are in this range. Make sure you've got a balance of punch in this range with higher frequency definition.
200Hz – 500Hz: Here is the mud again. Go easier on cutting this in the toms, as tom hits occur less often and need body. If the song contains a tom groove, you may need to cut more.
600Hz – 1kHz: This can be the cardboardy, muddy range in smaller toms, and often sounds 'flabby' in larger toms.
2kHz – 5kHz: Impact and definition.
6kHz – 10kHz: Crack and snap. Make sure this isn't too prominent – even this out with the low 'punch' of the fundamental.
10kHz – 20kHz: Not too much here, again, this can contain some cymbal bleed.

Big hint: The best way to get rid of cymbal bleed and background noise on toms is to use your DAW editor. Simply cut out the silence (well, it may contain sound, but no tom hits) between the tom fills by using the volume fader or just cutting out regions of the track. Make sure to fade the edges in and out to avoid pops and clicks. This will make your tom fills jump out clean and clear. It really improves the tracks.

General percussion EQ

Most typical studio percussion (tambourine, bells, shaker) should not be EQ'ed, but simply chosen for their sound in the first place. One of the best tools a small studio can have is a collection of percussion trinkets.

Most are not very expensive ($20 or so), and others can be made by jingling keys, bead necklaces stapled to a board rubbed with a drumstick, rice, corn or beans in a box or container, jingle bells from Christmas, etc. Walk around with an open ear for percussion and you'll find it everywhere. A unique sound can really make a track.

Of course, a little high end (10kHz +) boost can help a tambourine sparkle. Just be careful not to make things sound artificial unless that is specifically the sound you are after.

Congas, bongos and other tom-like ethnic percussion can be treated to the "toms EQ ideas." Try to give things a little distance from the mic when recording, as mentioned earlier, to capture an even picture that needs less EQ.

Cowbell and other harsh percussion can be cooled out by cutting around 1kHz, although this often sounds like you have lowered the volume rather than used EQ. Maybe that's all you need to do – balance your volumes. Unless you *know* that you need **more cowbell**...

Other keyboard instruments

Wurlitzer, Clavinet, Rhodes, Moogs, synthesizers, the list is too long to get into specific settings for every instrument, so here are a few hints.

Wurlitzer, Rhodes and Clavs are usually pretty well balanced and warm sounding on their own. Very important is the amplifier the keyboardist plays through (if they're using one). If something sounds harsh, use the bump and sweep method to find and reduce the offending frequency. This is also a great place to experiment with amp simulator plug-ins. You may want to keep a direct signal track as well as a mic'ed track so that you have more options when mixing. With a direct line signal recorded, you can always run the signal back into an amplifier and have an opportunity to re-mic the part.

Moogs and other classic synths generally need no EQ, unless they are hissing and need a low-pass filter, or if an oscillator is creating a heavy low frequency which needs to be filtered out to avoid distorting the track. Use your ears and work with the musician to create a sound that fits into the track and will therefore need less extra EQing.

Sampler and synth sounds are so incredibly varied that the choice of a fitting patch for the song is where the work is done. After that a careful evaluation of the tone is necessary. Often, there is a fake, cardboard sound to synth strings, horns and pianos/keys. The perfect remedy is to search around 1kHz or so with a parametric EQ 'bump and sweep' and then use a fairly wide 'Q' setting with just a few dB reduction. Then, add a boost of high shelving EQ around 12kHz to add some air and realism to the sound.

Be cautious of low frequencies in sustained synth sounds. Synth presets often add lots of deep low end so that the sounds will impress potential customers. You may need to reduce these lows to ease the sound into the mix with the bass, drums, guitars, vocals, fire trucks, and step dancers.

From Baglama, Banjo, and Bouzouki to Xylophone and Zither

Many of the stringed folk instruments need no EQ at all. Their development has been based on fitting into their musical position through the sculpting of the body and the tuning of the strings.

When used in pop or rock recordings, mandolins, banjos, dulcimers, zithers, etc. generally need only a mild boost of high end EQ (+2 dB at 10 kHz shelving) to make them competitive in the mix. Perhaps a little cut at around 400Hz for clarity as well. Before you carve up that banjo sound with EQ, make sure you are really sure it really fits into that heavy metal ballad...

The same thing goes for vibes, marimba, xylophone and other pitched percussion. A touch of high EQ will suffice. These instruments produce very pure tones and EQ will act more to boost or cut single notes or a range of notes rather than affecting the overall timbre of the instrument.

Brass instrument EQ

Crucial to a good brass sound is the choice of microphone. Trumpets and Saxophone can produce some very harsh tones when mic'ed up close, and you need to have done your mic technique homework!

20Hz – 60Hz: Only in tuba! Probably OK to filter off of other brass instruments to cut wind noise or mic rumble.
70Hz – 200Hz: The 'warmth' area of the sound. If you cut too much here, you'll have a kazoo track.
200Hz – 600Hz: This contains the 'honk' of the horns. You can "reduce the goose" by cutting here but, again, cutting too much can create kazoos!
around 1kHz: Harsh area. Cut here to cool out the 'bite' of a sound.
2kHz – 5kHz: Definition of notes. Can be dipped down a bit to make a part more 'background'.
6kHz – 10kHz: Blowing sounds and metallic tones.
10kHz – 20kHz: Breath and sparkle. Too much boost makes things harsh.

Sometimes a ***dynamic EQ***, like a de-esser, is more appropriate for horns. Tuned to a problem frequency, the de-esser can be set to reduce the frequency only when it exceeds a threshold that you set. This prevents the sound from becoming thin overall, but will reduce harsh peaks. More on this in the next section.

Dynamic control: Compression and Limiting

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Controlling dynamics is simple. Ready? Here goes:

"If something is too loud, turn it down. If it is too soft, turn it up. Thank you."

That work for you? Well good, then let's move along to the next chapter.

Just kidding. The important part to volume control is understanding *how* to make these volume changes, or even more specifically, ***how a compressor or limiter makes these changes***.

Let's imagine that you are recording a vocalist who is singing a ballad which has a wide ***dynamic range*** (see definition if necessary). You set the microphone gain so that you have a nice level going to tape. Then, as the song progresses, the singer sings louder and louder as the song builds. At some point the level will be too high and your recorder will clip or distort. So, being a good engineer, you begin turning the level down as the singer gets louder. Then, as the song comes to a quiet section, the singer begins to whisper sweet nothings – you turn the level up so that you don't just record nothing...

What you are doing is ***compressing*** the dynamic range of that signal. You make the loud parts quieter and the quiet parts louder, reducing the range form soft to loud. Now, imagine that you have to do this for 23 takes of the vocal. Getting tired of riding that fader yet? How about we invent a machine to do it for us.

That is all a compressor is - a dynamic controller; easier living through technology. So, what info do we need to give this machine to make it work the way we want it to? Here goes:

1) Tell it when to start turning the signal down. This is called the ***threshold*** – the level above which the compressor begins working.

2) Tell it how much it should turn the signal down. This is the ***ratio.*** We dial this in as a ratio how much goes in to how much comes out. For instance a 2:1 ratio means that for every 2dB of signal level above the threshold the output will only increase by 1dB. That is a fairly mild compression. 8:1 means that the signal must increase 8dB above the threshold for even 1dB more level to come out. And so on.

3) Tell it how quickly (or slowly) it should start working when the signal goes up. This is the ***attack time.*** A slow attack acts like it takes the compressor a moment to react, much like a person would act. A fast attack takes advantage of the fact that a compressor is a machine and can clamp down on that signal faster than a person can react, catching peaks in the signal very quickly. The attack time can vary from a few milli (or even micro-)seconds up to a full second.

4) Tell it how quickly (or slowly) it should stop working. This is the ***release time***. When the unit reacts to an input signal above the threshold, it pulls the output level down. When the input signal goes below the threshold again, it will return to normal, but how quickly it does this is important. The release time can range from a few milliseconds up to several seconds. Release times mustn't be too fast, or strange audio artifacts and sounds can occur.

5) ***Input and output level*** (output is sometimes called "makeup gain") allow you to control the signal level within the compressor. If you use a high ratio (like 8:1) you may need to raise your output a little to bring your signal back up to useable levels, especially if your threshold is set low.

6) ***Knee*** is an option offered on some compressors, usually "hard knee" or "soft knee". This has to do with the way the compressor reacts as the threshold is approached. It is independent of the attack time, but still shapes the attack of the compressor - does it cut in hard and quick, or soft and easy? If you want to hear the compressor working, hard is good. Soft knee is a smoother way of compressing whereby the compressor begins compressing by ramping up the ratio as the threshold is approached.

7) ***Compress or Limit.*** Some units allow you to make a choice of a range of settings. Limiting is just like compression, except the ratios are generally higher 10:1 up to "infinity:1" ("Make the freakin input as loud as you want and you'll only get one more dB outta me!") and the attack and release are often faster on a limiter, allowing you to truly "limit" the maximum output level.

There are plenty of hardware and software ***dynamic processors*** (that's the blanket word for a compressor, limiter or expander) out there, so why do some cost only pocket change while others cost as much as an automobile?

Since we are investigating gear as well as how to use it, let's look at the price/performance aspect before we get into the details of compressor function. As we know by now, the parts which go into a unit are a major factor in the price of high quality gear. So you'll realize right away the a ***Behringer, Alesis or ART*** compressor for a couple hundred bucks or less does not contain high quality audio transformers, discrete circuitry, stepped potentiometers or hand soldered point-to-point wiring.

This is not to say that you can't make a fine recording with an inexpensive unit, just that, again, *you may wish to be conservative in your use of such a unit as a link in your 'audio chain' during recording*. Once you record with the compression in the track, you can't remove it if it turns out to be the wrong setting!

The best bet when tracking a voice or an instrument is to err on the side of less compression. I have overdone it myself too many times – running an acoustic guitar into a compressor and just sqeeeeezing the life out of it because it sounded cool in the headphones while tracking, only to find out later that I ruined the dynamic of the performance.

The major advantages of high-end compressors are twofold:

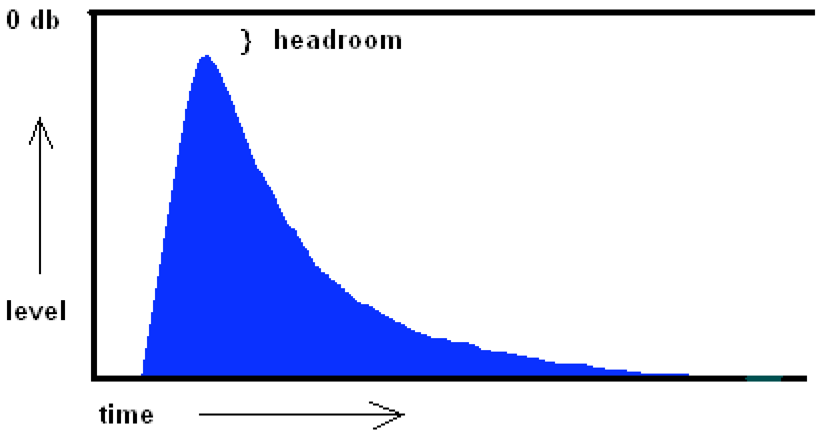
First, the high quality components do not (or very minimally) degrade the sound signal. Compressors can cause a loss of high end clarity, and this is something we try to avoid at all costs. Low cost units simply do not contain the proper components to preserve your nice signal's integrity.

Second, some compressors are desired for their transparency, some for their 'funky' qualities. A unit by Avalon, Crane Song, Focusrite or George Massenberg may be required for clean and "invisible" dynamic control in mixing or mastering, whereas "classic" units from API, Neve, Universal Audio or Urei, or modern reproductions and variations on classics from Chandler, Empirical Labs, Manley, Purple Audio, or Joe Meek are meant to color your sound in a particular way.

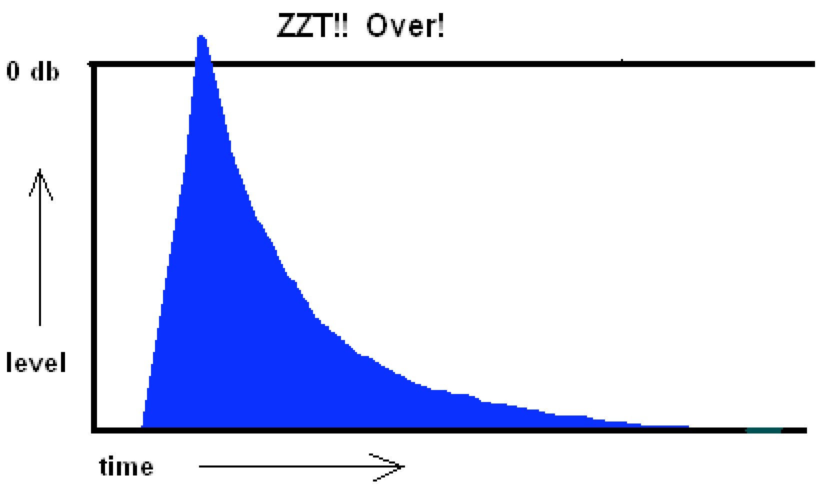
I must admit that because the ever-increasing quality of software compressors, I have sold off most of my own collection of classic gear, holding onto just a pair of Urei LA-4's which I really love. We'll hear some examples of audio using these as well as some other units and plug-ins when we get to the settings for instruments part.

Right now, let's get to know what happens to your signal inside these units a bit better.

I always like using the example of a drum hit, like a tom or a snare drum. With it's fast attack and strong signal dynamic, here is a simplified version of what a drum hit looks like in your editor window as a waveform:



Let's imagine that that is the level you have set for a good, firm snare hit. You have a few of dB of ***headroom*** (level to go before hitting 0dB as a safety net). But what could happen during tracking of the song is this:



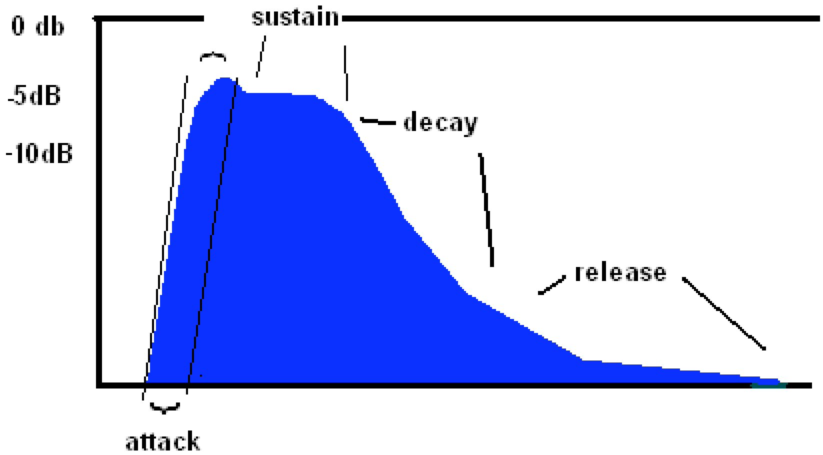
Oops! The drummer got into it and sent you a big loud rimshot hit, causing your recorder to clip and make nasty noises. Let's do the following:

Set a compressor in the audio chain between mic preamp and recorder, and set it to a 8:1 ratio and a threshold of -5 dB. Remember, this means that for the signal to rise over -5 dB, the input signal must go 8dB over the -5dB threshold for each 1dB increase. That means that to get to 0dB, the input signal must increase 5 x 8db = 40dB! That should keep our energetic drummer under control.

What happens now? Well, as long as the signal stays under -5dB, nothing. But as soon as that signal goes over the threshold, the compresser turns it down according to the ratio, and your hard hit snare rimshot will stay within bounds.

O.K., those are the basics of using a commpressor as a saftey net. This is an important feature, and is the original purpose of the device. Of course, using a device in a creative, sound manipulating was is where the fun is at! So let's look at the signal again in terms of it's four areas of dynamic development; ***Attack, Decay, Sustain and Release.***

Imagine that this following diagram is a visual representation (again, like in the edit window of a DAW) of a note played on a piano. A piano has a fast, hard attack (from the hammer), sustains briefly before the sound decays to a lower level, and then the note is release (disappears) in a bit of ringing strings and room ambience:



Keep in mind now that the attack, sustain, decay, and release I am speaking of are properties of the sound, not of a compressor. Yes, the compressor has attack and release settings, but keep those separate in your mind.

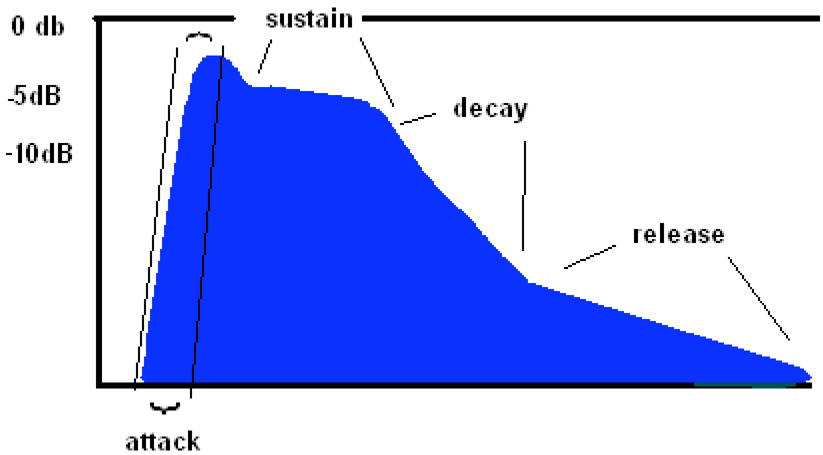
The neat thing about the compressor is that we can manipulate this signal to create a ***different dynamic***. We can use the properties of a compressor to shape this signal in the way the dynamic develops; more or less attack, sustain, decay and/or release can be dialed in with a few tricks.

Imagine that we want a piano player to perform on a rock track which includes a lot of percussion and acoustic guitars. In the mix, the piano seems to lack enough attack to 'cut through the mix' amidst all that percussion and jangly acoustic guitar. We also want to increase the sustain of the piano where the player held some nice chords in the bridge. Let's start twirlin' knobs:

Let's set the compressor to start affecting the signal at -10dB (set threshold to -10dB). This way we'll be sure to affect the signal to a fair extent. Then dial in a 4:1 ratio, and set the attack time medium to long and the release time fairly fast. Turn up the output (makeup gain) 3-5 dB.

When a note or chord is played on the piano, the compressor will react after briefly letting through the attack of the note (because of the piano's fast attack dynamic and the compressor's slow attack *reponse* time). This attack will sound louder, because we turned up the compressor's output gain. Then, as the compressor kicks in, it will reduce the output gain, but increasing the gain quickly after the signal falls below the threshold again (due to the fast compressor release time).

The final signal will be, dynamically speaking, more like this:



The attack will exaggerated through the combination of the increased output gain and the time it takes for the compressor to kick in. Through this visual example we can remember a rule for compressor settings:

Slow compressor attack times increase the attack of the signal.

The other way around is also true; use a fast attack setting to "catch" and reduce the attack of the signal. This is especially important for limiters which are most often used to catch the peaks of a signal (especially when mastering a mix) to stop clipping so that the signal can be made louder.

Now, notice that the ***sustain of the compressed signal is increased***. Once the compressor pulls down the output volume, there is room to bring it back up as the compressor releases. Even as the sound starts to decay, the compressor continues to bring the level up, back to our original gain setting; this sounds like more sustain!

In fact, this increase in sustain as well as the louder volume of the decay and release of the notes (due to the raised output gain) makes the whole audio signal louder to our ears. This is another rule which applies to all audio, not just the use of compressors:

The subjectively ***"loudest" of two signals*** measuring the same level will be ***the one with more sustain***.

For this reason, a shout in a cathedral sounds a lot louder than a shout in a carpeted living room full of furniture. The echoes and reverberation in the cathedral add to the sustain, and therefore the loudness of the signal. The shout isn't really any higher in level! You can explain that to the ushers as they show you the door...

Now that we've looked at the theory of these things, let's move along to some tips, tricks and audio examples.

KEEP IN MIND: It is assumed throughout that you have kept your recording levels set so that the average level is at around -10dBFS, with maximum peaks around -3dBFS. If your sound source recorded higher or lower in level, the recommended threshold settings will need to be adjusted accordingly – threshold is the setting which will vary most within the recommendations, based on the source recording.

Vocal Compression

Generally speaking, any vocal track will benefit from a little compression. A matter of fact, even when I recorded classical music, I used to sneak on a little 2:1 compression (or even 1.5:1 – very little) to make the recording sound better. No one was ever the wiser, and they used to say my recordings just "sounded better". [If you read this, Steve, – Sorry for being sneaky, but she sounded great, right?]

Vocal compression brings out the sense of 'breath' and 'air' making the recording more lively and 'in-your-face'. Remember, it is the increase in output (makeup) gain and perceived loudness which accomplish this.

On the other hand, an over-compressed vocal begins to sound flat and mechanical (this can be used to interesting effect) and sometimes 'fatiguing' = hard on the ears.

The key is to strike a happy balance. On a cranking Rock track, you will likely need more, less on a quiet ballad. In general I would suggest tracking the vocal with just a little 2:1 to prevent clipping until you get a feel for it. You can always play around with compression in the mix and experiment to your heart's content without writing anything in stone.

Here are some basic starting points for working on a vocal sound:

Style	Ratio	Threshold	Attack/Release
Soft acoustic music, Jazz	2:1 or less	-5 to -10. Lower thresholds will sound dynamically flat	Start with slow Att/ Rel. and adjust faster for most natural sound
Blues/Classic Rock	2:1 – 10:1 Some Blues and rock singers sound great if they are "squeezed" more with 10:1. If things a sound unnatural, back down to 2: or 4:1	-10 to -20. A low threshold with a 4:1 ratio will make a singer sound very "in your face"	Start fast and slow it down to taste. The stronger dynamics of Rock may require you to quickly reduce volume peaks
Pop	2:1 is a good start. For a really punchy Beatles-y sound, go 4:1 to 8:1	Adjust threshold down while increasing makeup gain to "squeeze" as desired	Start fast, ease up. Setting things to fast can make the compressor create 'pumping' or 'breathing' sounds
Heavy Rock	At least 4:1. You'll need to tame some wild grunts and screams, so be ready!	-10 is a good start. Be careful not to go too low (-25) or you'll bring up noise and breathing	Fast is good to catch loud peaks. If the track starts sounding too flat, bring your threshold up and slow down the attack to allow more dynamic

De-essing

Very important in processing a vocal is getting ***sibilance*** under control. Sibilance the 'sss' sound that can be very prominent in a vocal track, "***S***illy ***s***inger ***s***lur***s*** his word***s***". This is exaggerated by the proximity of the singer to the microphone combined with the little blast of air we release with the 's' sound. This can be about as pleasant as a poke in the ear, so we need to control it.

A De-esser is a compressor which is sensitive to an adjustable frequency range – from about 2kHz – 10 kHz. We can dial around to find the exact siblilant frequency, which varies from singer to singer.

If you don't have a dedicated de-esser, a regular compressor with a ***side chain*** input will do. The side chain allows you to put a signal into the control circuit of the compressor to affect when it compresses.

What you need to do is to take a copy of the vocal signal (through a buss or an aux send) and run it into an equalizer. Use a fairly narrow 'Q' setting, and boost 6 or more dB around 4kHz. Listen to this EQ'ed signal and sweep the frequency up and down a bit until you hear the sibilance really jump out.

This EQ'ed signal should now be connected to the side chain input of the compressor. This can be done by way of cables (the old fashioned way) or simple by way of selecting the proper inputs on your software compressor.

The end effect is that only the sibilance will be reduced. Set the ratio to 4:1 for starters and use the threshold to control the amount of reduction (lower threshold = more reduction). Be sure to set your attack and release times VERY fast, so you don't chop off non-sibilant syllables!

Guitar Compression

A good rock n' Roll guitar track becomes fatter and punchier, acoustic guitars gain presence and sustain, and blazing solo gets smoothed out and brought forward in the mix without hurting the ears - all with can be yours with good compression technique.

The key to good guitar tracks lies in controlling the attack. A heavily distorted guitar needs a bit more punch, that 24th fret wah-wah and fuzz guitar solo needs less attack (or maybe a de-esser set to cool out annoying frequencies!)

Since overdriven or downright buzzsaw fuzzed-out guitars are so common in recordings, it is important to remember what distortion is - a ***clipped*** signal. Clipped means that the waveform was chopped off a bit because the level got too high; the signal is compressed! This audio haircut also chopped off some of the ‘punch’ and ‘impact’ of the sound, so let’s create some!

Set a ratio of 4:1, turn the attack to it's fastest setting, and set the release to medium-fast. A release of approximately 100 ms. is good - for more active guitar playing set it faster so the compressor recovers quicker, but not too fast, or you'll lose the effect. Now listen to the guitar track with the threshold set so that your compressor is reducing the gain around 3 - 6dB. Slowly increase the attack time until the guitar track starts to gain some attack. Now listen to the track in the mix. Continue to adjust the attack time to find the right blend. Listen to this example to hear how this can breathe some life into a track: A Keith Richards style guitar riff (first) and then with some the compressor tuned in (second) ***(.wav files EX37a and EX37b)***.

An interesting side note: Since I was having trouble with the amp’s power supply, there is some 60Hz hum in the track. Notice how the compression effect makes this appear louder - the overall level is increased, for better or for worse!

The key with electric guitars is to create an audible effect. The electric guitar is not a very natural instrument anyway, and we can use more audible changes to create more interest.

For clean electric sounds, try a 6:1 ratio with a -15 threshold and a fairly fast attack and release. This will get you a Police “Every Breath You Take” kind of dynamic as the compressor allows a hint of the pick attack through before clamping down on the note to create a smooth texture. Increase attack time for more ‘pick’. John Frusciante (RHCP) is a big fan of compressed clean guitar sounds. He exaggerates it to the point where pick scraping, string noise, and the dirt under his fingernails can all be heard on the track. Try a 6:1 ratio and dig DEEP with the threshold; -20 or more. Start the attack time very fast. Then slow it down until you get the right ‘edge’ to the sound. Use the release time to allow more dirt and grime to rise up after the attack – the faster the release, the more fretboard scum will float to the surface. A hard knee compressor is essential for Frusciante’s sound.

Acoustic guitar tracks can both lose and gain from compression. A minor disadvantage of compression is the way in which lower level sounds become louder. In the case of acoustic guitars, the squeaking and sliding sounds of the strings can be made loud and irritating by too much compression, especially when the strings are fresh and new. It is important to pay attention to these sorts of distracting side effects and get them under control. In the case of string squeaks, try less compression, a de-esser tuned to the squeak frequency, or automating volume dips at the squeaks in your editor window.

On the bright side, acoustic guitar tracks gain balance, solidity and a sense of depth and presence from the right amount of compression. The settings are very material-dependant, but here are a few basic tips:

- **For an active, strumming accompaniment track**, try about 4:1 ratio, a threshold of -10 and a medium fast attack and release. If there are any hard peaks or uneven dynamics, try speeding up the attack and release to catch the peaks and smooth out the performance. Be careful not to squash out any dynamics which are essential to the song arrangement!

- **For a fingerpicked part,** try a lower ratio (2:1 or 3:1) and dig a little deeper in the threshold, down -15 or more. This often brings up a sense of presence and proximity to the instrument. Keep the attack and release on the slow side to start. If you want to hear the compressor more, speed up the attack and release and opt for a 'hard knee' setting if available. Listen again to the fingerpicked example how it is (first) and then with a compressor "pushing" the guitar and it's attack forward (second) ***(.wav files EX38a and EX37b).***

- **For capo'ed guitar parts**, be cautious about too much compression. The capo always seems to 'squash' the dynamic a bit anyway. I would recommend a limiter-style setting to catch the odd loud note, so that you can increase the overall level of the part. Try a 6:1 or 8:1 ratio with a higher threshold (-5 or so) so the compressor isn't doing more than catching big peaks. If this setting isn't happening, try the settings for the fingerstyle idea above and see if that helps. Always consider the option of NO compression!

- **For big juicy Beatles style rhythm guitars,** try really sqeeeeeezing that guitar track. On some Beatles songs you can really hear the compressor working and 'pumping' the sound – a sort of attention-getting shaky modulation. What the hell – turn everything all the way up! (Well, watch the output knob – don't blow your speakers). Cranking the knobs to the hilt is a great way to get to know your gear anyway, and many a musician has found a cool sound by saying “Hey, what does this knob do?” and twisting away. In all seriousness, you will need at least an 8:1 or 10:1 ratio, a threshold of -20dB, a medium attack and a fast release for that modulation effect. Crank the input knob to squeeze more. Listen to some strumming as is (first), and then with the Bomb Factory 1176-style compressor squashing the sound (second) ***(.wav file* EX39a *and* EX39b).** Notice the increase in the presence of the pick attack and the interesting distortion added to the body of the sound.

Drum and Percussion Compression

Due to the strong transients contained in many percussion sounds, we often need to limit the maximum level of the percussive attacks to fit the sounds onto tape (disc) with a high enough level and without clipping. For this reason, a limiter is an excellent tool in recording or mixing drums and percussion.

Another advantage to dynamic control on drums is the way in which the resonant tones of the instrument are made relatively louder. When the compressor or limiter “clamps down” on the transient and attack part of the sound, the volume is lowered. With the proper release setting, the signal level is raised again as the drum sound decays, and the “body” of the sound is made louder.

- **Setting your limiter:** Try not to overdo limiting during tracking. You can’t get those transients back, so err on the side of caution until you become confident in your processing skills. Set the limiter to reduce peaks by about 3 – 6dB as a start. This will allow you to increase the overall track level, and you’ll hear more body to the drums. In a rock or pop context you really can’t go wrong with this setting. Ratio should be 10:1 or more, and the attack and release very fast.
- **Compressor settings** should be made carefully on snare, kick, and toms. If you dig in too deep (low threshold) and use a high ratio, you’ll start making popcorn – your drum hits will sound like little pops with a weak transient and no body. Always start with a 4:1 ratio (or so) and don’t bring the threshold down below -10 or -15dB. See how this sounds in context while adjusting attack and release. Make sure that your drums have a nice balance of “snap” (attack) and “body” (release is fast enough to bring up resonant tone)
- **Cymbals often need more compression in rock tunes.** Don’t squish them too much going to tape, but be sure to try higher

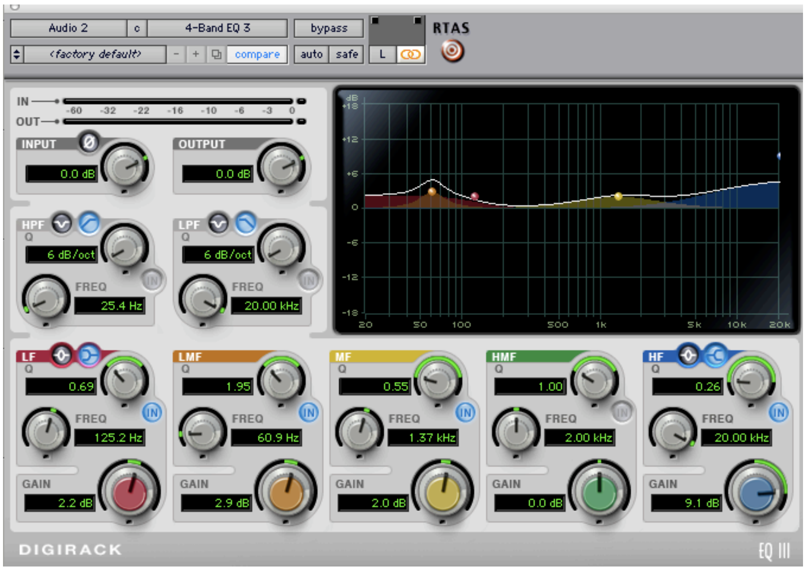
compression settings when mixing. Some of that smashing, pumping compresssion sound can add life to a track. Try a low threshold with medium attack and release settings for a classic kind of compressor response.

- **Squeeze the hell out of room mics.** The most famous compressor setting for room mics is probably the "Urei 1176 with all the buttons pushed in". Although we can't all afford the 1176 or a repro of it in hardware version, there are some great software plug-ins out there that do the sound very well. Engineers have found that by cranking all the knobs and pushing in all the buttons at once, the 1176 will do wild and demented things to the incoming audio. This sound makes room mics on drums sound especially good. The trick is to find a good balance of this sound in relationship to the rest of the kit. Start with the faders for the room mics all the way down and then blend in more of the squeezed room mics until it sounds good in context. This method can really add that professional sense of space and "solidness" to your tracks that is subtle but makes a tremendous difference.
- **Be careful of percussion compression** on things like bells, tambourine, triangle, etc. These small bodied instruments need to let their transient cut through the mix – that's what they are there for. If you want more body to the sound, try reverb or (if your acoustic environment allows) a room or overhead mic for a little "air". I find that compressed tambourines sound fake and plastic in a mix. Concentrate on a good even performance before compressing!
- **Bus compress your drums.** Once you have all the individual sounds compressed and limited, run everything through a stereo compressor. This often provides the 'glue' to make the whole kit sound smooth and together. Many engineers also run the bass into this compressed submix to create a tighter rhythm sound. Try a 3:1 ratio and a medium fast attack and release. Pull down the threshold so that the compressor is just working on louder hits. Listen in context and try squeezing more by lowering the threshold and raising the output. Listen to this with some guitars, vocals, etc. playing along. Switch the compressor in and out to see how you like it. Sometimes this effect is not necessary, sometimes it sounds great when set to really work the sound hard. Use your judgement. Remember that less processing is often better for overall sound quality, so remove the compressor if you don't think that it helps much.

Let's listen. Here is a sample of stereo drums performed by BFD. The sounds are well recorded in a world class studio with great gear. Now listen step-by step as some clever ProTools EQ and Dynamics processing cranks up the sound, adding life and a much more unique sound than the basic part had.

.wav EX40) The unprocessed drums.

.wav EX41) ProTools EQ3 adding the following EQ curve:



.wav EX42) The basic drums with Steven Massey's "TapeHead plugin added. This does a fair simulation of analog tape - notice the mild distortion "warmth added" and the broad midrange boost.

.wav EX43) The basic drums with some 4:1 compression via Bomb Factory's BF76. Like this:

Compare the dry track with this and notice the difference. I have made the settings a little heavier than I'd probably use in a mix in each case, so that the effect is obvious.

.wav EX44) Here we have the drums packed through all the effects listed above.

.wav EX45) This is the same as EX44, but with Massey's Limiter cutting the peaks down a little and cranking the volume. WOW! Compare this to the original track!

Keyboard instrument and horn compression

Less is usually more when it comes to brass and keyboard instruments. The main point is to get the dynamics to fit into the track. The heavier the arrangement (more guitars, horns, keys, background vocals, etc) the more you may need to compress individual instruments so that they can hold their own in the mix.

Proper arrangement is of highest importance, followed by relative level settings. Before you reach for the compressor, make sure that the balance of keyboards to guitars or horns to drums, etc. is all well mixed.

When you find that a piano part sounds too thin in the mix even though you were happy with the original sound in terms of micing and EQ, then it may be time for some dynamic control.

- **Listen to The Beatles "Lady Madonna"** – that piano is heavily compressed, and makes it sound almost like an electric piano because of the sustain it gives the instrument. That sound can hold it's own in the mix, it has body and some snappy attack. You can get a similar sound with a 8:1 ratio, pretty fast attack and release and a low threshold. Just be aware that you actually *lose* dynamic range with this setting! Yes, the sound is cool when your rocking along, but it would be totally inappropriate for a ballad which needs to provide some dynamic counter-play to a vocal – it wouldn't allow for a delicate touch or subtle phrasing.
- **Many synth patches are already compressed,** so check internal keyboard settings to see what's in there. Maybe you would prefer a compressor that you have in your rack to an internal keyboard one. Check it out!
- **Horns need special care.** I always hated the horn sounds on Rolling Stones records like "Exile on Main Street" even though the album is brilliant. I think that someone overcompressed them and

the loss of the attack dynamic makes them sound like kazoos! A little soft compression can make a horn sound smoother and breathier (by bringing up interesting low level sounds). Just be careful not to overcompress the transient edge or you'll be in the kazoo chorus of the local Lion's Club parade.

- **Funky, Stevie Wonder style Clavs and other keys** can stand a little compression. A bit of limiting can cut over-the-top transients and make the sound fuller and fatter. Be sure to remember that rhythmic, syncopated parts play as much of a percussive role as the do harmonic/melodic and should therefore retain a percussive quality.
- **Ghostly, unusual effects** can be pulled out of pianos, Wurlitzers, Rhodes, etc. By heavily compressing them and reducing the attack significantly. These kind of settings create sounds which are artificial sounding – more of a pad or effect than a normal sound. Radiohead has done some cool experimenting on albums like "Kid A" and there are many more new sounds to be discovered by abusing compressors in combination with keyboard instruments.

Multiband Compression

Another of the major audio scene buzzwords in the last 5 -10 years has been the multiband compressor. Many engineers (whether they admit it or not) can't live without them anymore, and the difference in quality they can make on an individual source or an entire mix can be astounding.

First of all, what *is* a multiband compressor, and what can it do for you? Think of it as four compressors (usually – some have more or less) in one unit. The audio spectrum is then divided into frequency bands and these frequency bands are compressed individually. So, as some heavy low frequencies are being tamed, the high frequencies aren't also pulled down in volume briefly. Same thing goes all around – a blast of buzz-saw midrange on a distorted guitar can be compressed a bit without ducking out low end power...

What this all adds up to is a very intelligent compressor which operates more invisibly across the sound spectrum, allowing the engineer to control peaks and therefore raise the loudness (overall average level) of the mix. A normal compressor does this as well, but heavy compression or the odd peak in the audio can cause "pumping" and "breathing" in the sound – the sound of over-compression and bad mixing.

Multiband compressors are fairly easy to operate, as many have a selection of presets which operate smoothly. There are, of course, enough parameters – four stereo compressors! – to become rather crazed from knob tweaking. The key to proper multiband compression is not to overdo it! The initial sound of the m.b. compression is satisfying – loud and fully of excitement and clarity – but beware of making your carefully recorded mixes sound like F.M. radio.

I find it more useful to view a m.b. comp as a dynamic EQ; when certain frequencies become too loud, the m.b. comp just reaches in and briefly turns them down. This can be very useful in the home studio where odd room resonances and unfortunate sound reflections add up to an imbalanced EQ spectrum. This is especially true with home recorded drums. It can be very hard to get a balanced and punchy sound, and if you use a parametric EQ to define problem frequencies and then tune a m.b. comp to control these frequencies, you can get a much fuller and clearer sound than with EQ and compression alone.

Always be sure that your mix sounds as good as you can get it before you start compressing, and this goes double for multiband compression. If you add too much low end to bass and drums in your mix and then crank up the m. b. comp, you will still hurt the low end of vocals and other natural, evenly EQ'ed sounds. There are varying opinions on whether it is O.K. to mix thru a compressor. Some say yes, some no – I say mix with a comp if you like, just also save a non-compressed version for the mastering phase. I would, however, advise against mixing through a multiband comp, as this can really skew your impression of what is happening.

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Now that you're tuned in to the major methods for adjusting the sound of your tracks, lets look at some inventive ways to add interest and depth with effects.

Effects processing and design

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The use of effects such as reverb, delay and flanging is what set the late 60's apart in recording history as much as the style of music developed then – technology inspired by people. In some cases the experimentation with effects led to the composition of songs – people inspired by technology.

The right reverb, a shimmering chorus, a rhythmic delay or a frenzied distortion tidal wave – these sounds can make or break a track. Musicians such as Radiohead, Producer Trent Reznor, and Guitarist Tom Morello of Rage Against the Machine and Audioslave have proved to us again in recent years that there is still undiscovered territory in the land of effects.

Let's take a tour of some of the most common effects devices and the ways they are most often used and misused. Thinking up new ways to use effects and experimenting with these ideas is key to creating your own unique sound as an audio engineer, producer or musician.

Reverb

The classic effect is reverb, and is really just short for "reverberation" which is a complex combination of multiple echoes and sound reflections which sounds not as one distinct echo, but rather as an ambient decay or "trailing off" of the sound which is independent of the instruments own resonances.

The first reverb captured on recordings was simply a result of the acoustic environment in which the music was recorded. Want more reverb? Record in a bigger room.

The next step was taken when clever engineers began to place individual instruments into lively acoustic environments such as stairwells or grain silos. Listen to an Eddie Cochran recording to hear this.

As technology and the clever engineer developed further, the reverb send was invented. It was a simple idea. A copy of the signal to get reverb added was sent to a loudspeaker in a large room, empty tank, available bathroom, etc. On the other side of the reverberant space was a microphone to pick up the reverb created in the space. This mic was

returned to the console and sent to a fader that, when raised into the mix, added reverb to the instrument(s) sent to the reverb chamber.
Plate reverbs still in use today operate on the same basic principle.

Spring reverb, long available in guitar amplifiers, uses a set of springs and again the same principle to create a little decay on the notes you pluck into that old amp. This is why you can hear a “sproing” when you thump an amp with the spring reverb on – the physical motion of the springs is converted into sound.

Digital reverb devices use processors to mimic the way in which multiple reflections and echoes of a sound build what we hear as reverb. The quality of the processing is of utmost importance, because our ears are very critical of audio clues as to what type of physical environment we are hearing. The industry standard has long been the reverb processors from **Lexicon**, although there are so many companies creating quality processors that you must use your ears to decide which is best for you. Always remember that no matter how many whiz-bang little tricks a processor does, the most important thing it will do for you is a decent room, hall and plate reverb. These are things you will use in almost every mix you do; a flanged fuzz-wah-echo is not.

This next example uses a high-end Lexicon plate reverb. Surprisingly enough, even their lower cost units sound pretty great although maybe you could beg, borrow or, um, rent a PCM-90 or similar for one day while you mix? ***(.wav file EX_GREATPLATE)***.

In the interest of de-mystifying reverb names, keep in mind that the type of reverb most often describes an acoustic environment or a device. Hence “room”, “hall”, cathedral”, “auditorium”, “chamber”, and the like are usually meant to approximate the sound created by the reverberation found in those types of locations. Some presets on reverb devices have clever names like “Elvis Vox” or “Rockin’ vibe”, so you’ll need to listen and look at the basic reverb in the preset menu to decide if it is right for your track. “Plate” reverbs, of course, try to approximate the sound of a physical plate reverb – a very smooth decay with no ***early reflections*** to give a clue as to the room dimensions.

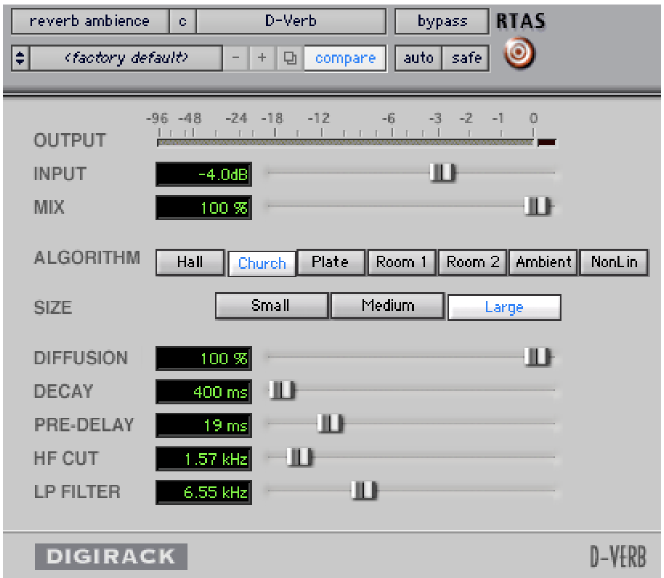
So that brings us to the point where we need to understand the finer points within the reverb sound itself:

- **Predelay**: This is the difference in time between the original sound and the begin of the reverb. Usually measured in milliseconds.

- **Early reflection**: This portion of the reverb gives us the spacial cues our ears recognize as the size and shape of the space. In a room, hall, chamber, etc. the sound source reflects off the walls, floor, and ceiling, and makes a quick return to our ears. These reflections are typically very short echoes, and their level and tone quality gives us a sense of the type of material in the room as well. A tiled bathroom has very loud, bright early reflections. In a concert hall these reflections will take a little longer to arrive (due to the size of the space) and will likely be diffused by the design of the room to prevent distinct echos which can interfere with the timbre of instruments. A carpeted living room will have early reflections with less high frequency energy, due to the absorbing qualities of carpets, curtains, upholstered furniture, etc. Plate and spring reverbs, due to their artificial nature, lack early reflections, making them suitable for creating sustain without a sense of 'room'.

- **Decay**: The length of time it takes for the reverb sound to trail off. It is important to find the right amount of reverb decay in your mix – too short and the reverb is ineffective, too long and things become muddy and indistinct. The trend in recent years has been towards shorter reverb times. In the 80's, many recordings were loaded with long reverbs (especially on drums and vocals!) for dramatic effect. Current production trend is to create a sense of "space" and "room" without long artificial sounding reverb trails.

  TIP: Decay can be a very interesting parameter to manipulate: Imagine a tiled bathroom with the decay of a concert hall! An auditorium, church or concert hall with the decay turned down can become an excellent ambience reverb! This is currently a popular variation on the reverb scheme. Here is a little percussion sample with a church reverb set up to have an unnaturally short decay. It actually sounds quite like a good studio space.***(.wav file EX47).*** The effect is on at the start, turned off, and then back on again so you may hear the difference.



- **High frequency damping**: In a natural space, high frequencies are absorbed more rapidly than lower frequencies. This effect is also dependent upon the materials in the reverb space. A stainless steel tank will bounce high frequencies around a lot longer than a carpeted living room. Try adjusting this parameter to control the brightness of the reverb and therefore, again, it's "attention getting" factor. Sounds with more high frequency content simply grab our attention more. If you are happy with the length of a reverb (decay time) but it still sounds too "obvious" try turning down the HF damping. If your reverbs sound "muddy" or indistinct, try reducing the HF damping for a brighter, clearer sound. You may then need to lower the level or the decay of the reverb to compensate!

- **Reverb Density**: The many later reflections (after the louder early reflections) which make up the decay of the reverb are, naturally, separated in time. "Denser" reverbs have these reflections placed closer together, making a smoother reverb well suited to drums and percussion. In lower density reverbs the reflections are more distinct, nice on vocals and non-percussive instruments, but can sound choppy or grainy on percussive sounds.

- **Diffusion**: This parameter relates to the behavior of the decay portion of the reverb. In layman's terms; the more diffuse the reverb the "smoother" it sounds. The disadvantage to high diffusion can be a loss of the sense of 'room' and 'shape' to the reverb. A room reverb typically has a low diffusion, a plate or concert hall reverb a rather high diffusion. If your reverbs get cloudy and indistinct sounding, again, reduce this parameter to gain shape and clarity and consequently reduce decay and level to compensate.

Getting to know your reverb parameters is one of the more important steps in creating clearer, bolder mixes. Always turn a critical ear on your reverbs and try to make them distinct and be sure that they serve the mix and the overall vibe of the song. An intimate vocal backed by an acoustic guitar may call for a room or ambience type reverb to serve the intimacy of the song, even if a big juicy plate reverb sounds flattering!

Also consider the possibility of adding or removing reverb effects during the course of the song. Perhaps a touch of plate reverb should sneak into the chorus or bridge of that intimate vocal and acoustic guitar piece – that could be just the touch that adds a little drama to the song without creating a big washed-out sound the whole way through. It also creates a happening within the song, helping to keep the short attention spans of the general public engaged!

Make your own reverb: Don't forget the origins of reverb – actual rooms or spaces of some sort fed with a speaker broadcasting the sound source and microphones to pick it back up. Even if you can't make a lot of noise where you live and/or record, consider taking advantage of a friend's office after hours, parent's house when they're away, etc. to create yourself a reverb chamber! You don't need to do this live when mixing - you can just record the ambience back to a track or two and mix it in as you like later. You can even control the pre-delay by sliding the track around in a DAW editor window! Listen to this example of a bathroom used as a reverb chamber on an electric guitar melody ***(.wav EX46).*** It has a kind of hard, bright character that we though was right for the intensity of the song.

Delay and echo

First of all, let's define the difference between the two. **Delay** is generally thought of as an exact repetition of the sound source later in time. Sometimes the level is lower or the delayed signal is otherwised processed, but the basic idea is an exact repeat. An **echo** is meant to simulate the way a natural echo sounds – the repeated signal will be lower in volume, and have less high frequency content as in a natural acoustic environment.

Since echoes are in essence a type of delay (especially when created on a digital processor), let's look at the parameters of a delay and get to know how these adjustments can help us.

- **Delay time**: is simply the amount of time between the original sound and the delayed signal. This is usually measured in miiliseconds and can vary from a few milliseconds up to several seconds. Longer delays generally require more processing power. **TIP**: Keep that in mind when setting up delay plug-ins on host-based processing, such as Nuendo or Protools LE! Use the plug-in with the shortest maximum delay time that suits your needs. Longer delays with the delay time turned down still eat processing power!

Very important tip: use this formula to determine the delay time for your music. The formula is based on the fact that note duration is linked to the tempo of the song. You'll need to know the tempo of the song in B.P.M.:

(BPM / 60) x (1000 x note value) = delay in milliseconds

Example: (120BPM/60) x (1000 x 1/8th note) = 250 ms.

It's a pretty simple formula, especially since there are so many songs at 120BPM!

TIP: Once you have your exact delay time calculated for the type of note (eighth, sixteenth, etc.) adjust it a little longer or shorter for a more organic effect. Making it shorter than the actual note value creates urgency or a sense of racing ahead. Longer than the note value sounds relaxed and spacious, even lazy or drunk! Use these ideas to further enhance the way you effects speak to the mix.
Use the millisecond calculator for reverb predelay times as well – this can really help clarify your mix by making sounds occur on the beat.

- **Level/repeats/decay**: Level is simply the volume of the delayed signal. Repeats is obviously the number of times the input signal is repeated by the delay. Some delays allow you to adjust the decay of the volume over multiple repeats of the signal. This provides that trailing-off echo effect.
- **High/Low filters**: Some delays allow you to filter the delayed signal. Removing highs aids in creating a classic echo effect. The good old rockabilly echo can be cooked up with a short eighth note or sixteenth note delay, three to five repeats, and filter off the highs down to about 2 kHz or so. Filtering lows is useful when adding delay to bass-heavy sources. Too many

repeats of low frequency material will cloud up and imbalance your mix, so use that filter! Finally, filtering highs and lows can give you that CB-radio effect which is popular right now. I don't know if this is a good thing. I hear far too many filtered whole note (1 bar later) repeats of vocal lines in crappy modern pop... Think of a new use for this idea!

- **Modulation**: An interesting think happens if you change the delay time while a signal is passing through the delay unit – the pitch changes! This is the principle behind chorus effects! A mild modulation of the delay time (longer-shorter-longer-shorter-etc.) occurs, and the pitch varies slightly on the delayed signal. This can also give a little life to echos, especially for guitar, synth or brass solos. Be cautious – it draws attention to itself. To create a chorus effect from a delay, set the modulation and then reduce the delay time to a very low setting, this way it seems as if the source signal is modulating because you can't hear a distinct echo.
- **Flanging**: A flanger is really a zero-delay chorus in a way. You simply add a time-modulatedcopy of a signal back to the original with no delay between the two. This can't be done with all delay units, because many do not allow for a delay setting of 0 ms. The original flange effect was created by synchronizing two tape machines which were playing back duplicate tapes. One machine was then slowed down and released repeatedly by literally pushing against the flange of the tape reel by hand! This technique was also pioneered by recording genius Les Paul!

Using delays to design effects is one of the basics of creating your own sounds. We just looked at how to create a chorus from a delay unit. I also found it very interesting that Butch Vig (producer for Nirvana among many others) preferred to create his own reverbs from delays. He found that reverb units sounded fake and seemed to just add a lot of mud to the mix, so he stuck with delays! The concept is fairly simple, you just need a few delay units. You will really be creating ambience, rather than reverb, as the complex reverb tails (decay) are far too complex. The best way to start is to use two delays, one panned to the left, the other to the right. Set the delays for about 40ms. Add a some HF roll-off for a natural sound (unless you want it to sound like a tiled room), and set for no repeats as a start. Begin adjusting these delays and get a feel for how this affects the sense of ambient space. This is a very important method for adding depth to close-mic'ed recordings!

Take a listen to these home-made delay/ambiences:

.wav file EX48a* and *EX48b) Just a straight up kick/snare pattern. 48a is dry, 48b adds delay/ambience. The settings are slightly different for the left and right channels (both around 50ms.) and the LPF is cranked to to around 1kHz on both sides. Keeping the left/right settings slightly different builds a sense of stereo depth.

***.wav file EX 49*)** Here's a funky Wurlitzer part put through the same delay. It starts with the effect in, then the effect is off for the repeat. Notice how you miss the ambience when it's gone? If you like the part you have my permission to steal it.

Pitch shifting

Real-time pitch shifting involves digital signal processing, and is best exemplified by the excellent devices manufactured by Eventide. The main parameters is, obviously, which note you would like to shift to. This is usually measured as an interval (or a pitch change in *cents*) above or below the original note. You also may blend the pitch shifted note with the original to create a harmony.

"Smart" pitch shifters are able to create fitting harmonies for a particular key. This is important in some instances because you cannot just add a major third (for example) above any note you play and get diatonically correct – that is to say 'in the right key' – notes out of the pitch shifter.

Aside from the typical use of a pitch shifter to create harmonies, here are a couple other cool ideas to try:

A popular vocal effect uses pitch shifted short delays of a vocal panned left and right. Try about 30 ms. delay on the left , 20 ms on the right. Then, change the pitch of one side up about 12 cents and the other side down 10 cents. This will give you a thick, vibrant sound on a vocal track which sounds 'larger than life'. Vary the delay times and pitch change to taste.

A way to add interest to reverb or echos is to feed the input of the reverb or delay with the output of a pitch shifter. You can then have a guitar solo echo an octave lower. Or higher. Or try pitch shifting a vocal reverb up a diatonic third for a sense of harmony without a definite harmony track. Try several tracks pitch shifted to different octaves, each with a tremolo or gate on it – set to different speeds! There are a lot of possibilities here, so start experimenting! These ideas are especially easy to set up in a DAW.

Volume based effects

Aside from compression and limiting, there are other effects which can be created by creative volume control. The interesting side of this is what can be done by combining and altering the standard uses for these effects.

- **Tremolo**: A steady increase and decrease in volume of a sound creates the classic tremolo effect. This was especially popular in the guitar sounds of the 50's and 60's and is experiencing a bit of a comeback. The recent Green Day hit "Boulevard of Broken Dreams" prominently features an electric guitar with heavy tremolo, almost a **gated** effect.

- **Panning**: The volume in the left speaker reduces as the volume in the right speaker increases – and the sound source seems to move from left to right. That is panning in a nutshell. Aside from dramatic sweeping-across-the-soundstage effects, panning can be used as a sort of stereo tremolo effect by rapidly panning from left to right. Panning delays or echos can be used to add interest to vocal tracks, percussive sounds or solos.

- **Gating**: Is the hard cutoff of the volume of a signal. This can be set to occur periodically (like the Green Day example above) or to occur when a sound falls below a certain volume threshold. Gated reverbs were popular in the 70's and 80's and were a big part of the Phil Collins Genesis drum sound. The reverb output is run through a gate, and is set to cut the reverb off quickly after a short decay time – Ba-boom – boom – boom!

- **Volume swells**: A classic pedal steel effect from a standard guitar, a piano turned into a synth pad, ghostly orchestral drum rumbles from a regular kit – all of these effects can be created by using volume swells. By removing the attack portion of a sound completely and then fading in the body of the sound you can turn your regular selection of sounds into a whole new palette of sonic colors. This is easiest accomplished in a DAW by using the volume automation function. Just turn the volume all the way down and then create automation events to raise the volume up to the desired level after the attack. Fading back out is sometimes very effective as well.

- **Reverse audio**: There are effects boxes which can play back snippets of sound in reverse, but the true reverse effect is best done nowadays with the “reverse audio” function in your DAW or sampler. Cymbal and drum hits in reverse are a classic effect, previously created by reversing the tape reels on an analogue machine, but now even easier to line up to the beat in your editor window. Don’t forget to reverse pianos, horns and guitar chords as well as percussion parts, odd bits of ambient noise and other unusual sources – you never know what you may find! Don’t overdo it, but spend a rainy afternoon looking for a new sound which could set your mix apart from the rest! Listen to the reversed echo effect on this guitar solo ***(.wav file EX_FREEDOM)*** along with those low ominous Moog notes, it helped build the freaky vibe we wanted.

Mixer layout and mixing

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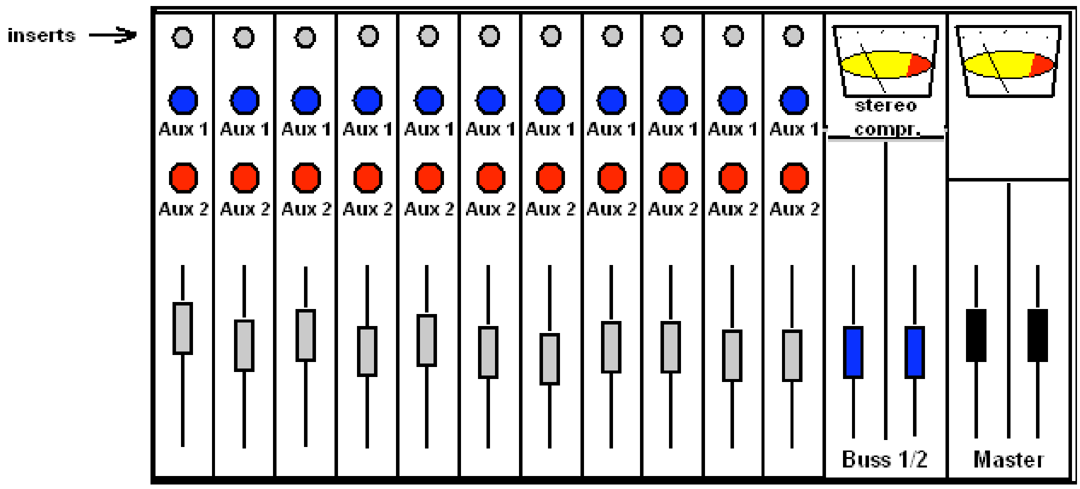
I have mentioned many mixing tips and methods for signal routing throughout the text so far, now it's time to add a few more important clarifications, methods and a few good checklists to help you keep your eye on the ball when putting all your precious audio tracks together into a song.

One of the most important things to get clear in you head is signal routing. Let's take a small standard mixer as an example, and imagine that we have a song from a simple vocal, piano, bass, and drums combo to mix. We'll add some reverb and compression and look at ways of handling the tracks, effects returns and stereo outputs. There are many ways of mixing, but for those of you with little experience with bussing, inserts, and auxilliary sends and returns, this should make a nice tutorial.

First of all, let's look at our mixer. It doesn't look too fancy here, but trust me – it's a well built little unit ☺

We'll assume that all the individual tracks have been optimally EQ'ed using all the tips in the EQ chapter and some careful listening. Same goes for compressor settings.



Let's imagine that :

- The first three input channels are drums (there would likely be more, but there's no need to illustrate a huge mixer that won't fit on one page), and the drums are EQ'ed and balanced relative to one another.

- Input four is the bass. In the insert point of the bass there is an external compressor inserted (bass audio out to compressor, compressor output returns to channel).

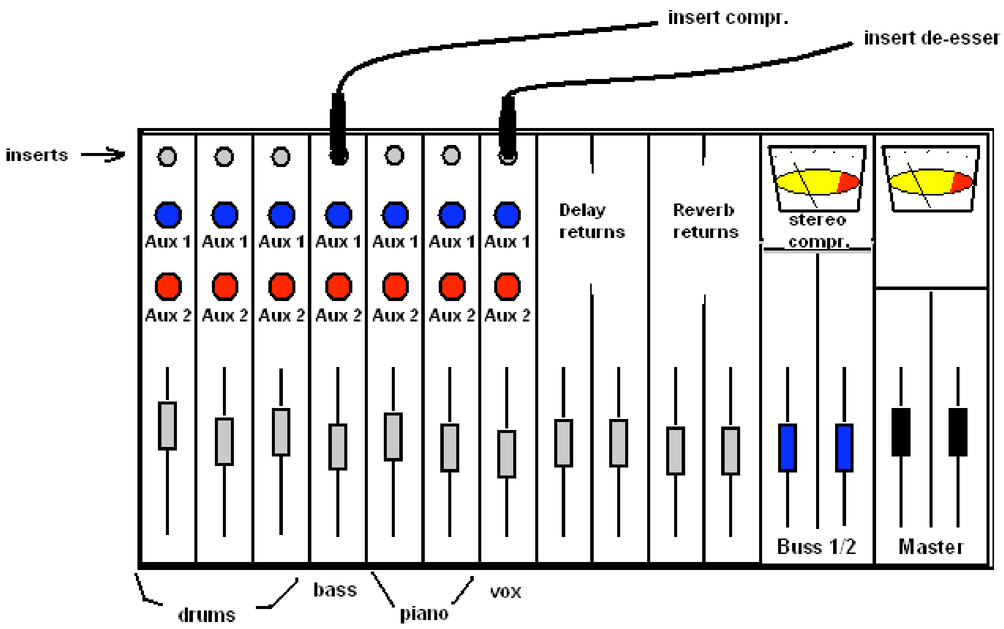
- Inputs five and six are the piano, recorded in stereo. If an M/S matrix was used to record it, then you would use the setup shown in the “Mic technique” chapter.

- The vocal is on channel 7, and we’ll imagine that since the recording was a little “ess-y”, the insert is connected to a de-esser.

- Aux 1 feeds a delay set to create a stereo ambience effect, and is added to the bass, piano, and vocal. The drum room mics provide natural ambience.

- Aux 2, a plate reverb, is turned up a bit on the snare drum, piano, and vocal.

Let’s look at the mixer again:

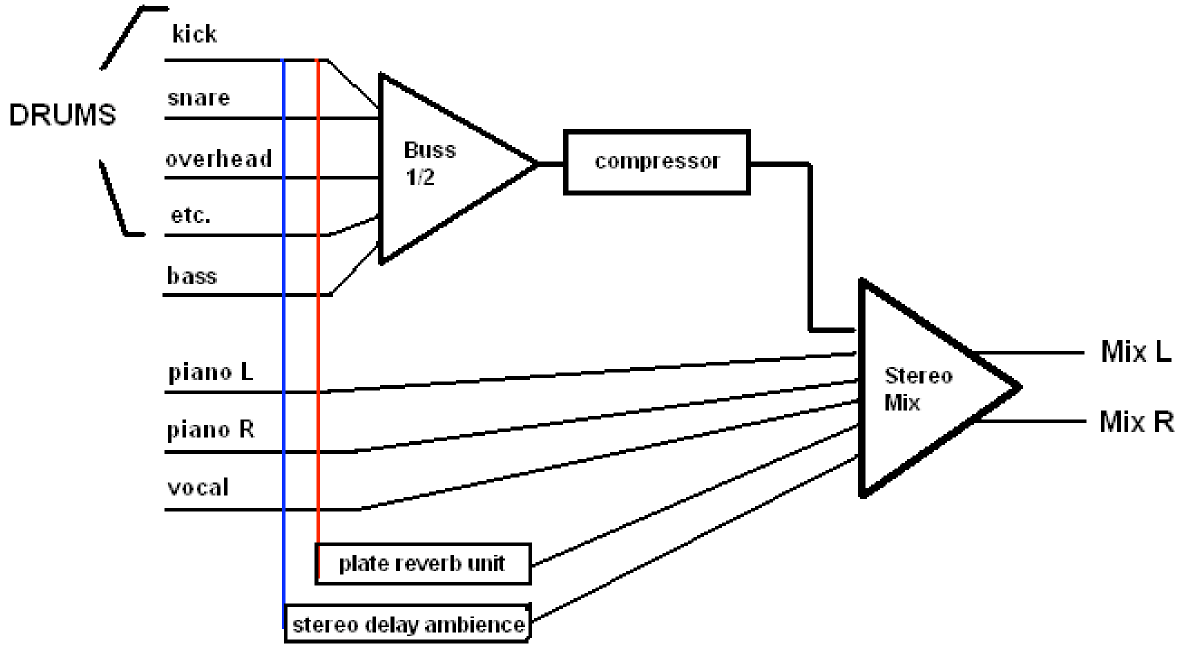


Now, on the mixer there are typically selector switches at the bottom of the channel strip which allow you to choose where the signal goes after the channel fader. The choices are typically “master” (or stereo out), buss 1/2, buss 3/4, on so on for however many busses you have available.

So, we'll send the piano and vocal to "stereo output", and all the drum channels and the bass to "buss 1/2". After passing through buss 1/2 and it's compressor, the signals will continue on to join the piano and vocals in the stereo mix.

The aux sends take a copy of the signal (at the level you set) and feed it to the units they are connected to – and return it through their channel to the stereo master fader.

It is important to understand and visualize this signal flow. You can imagine the signal flow like this:

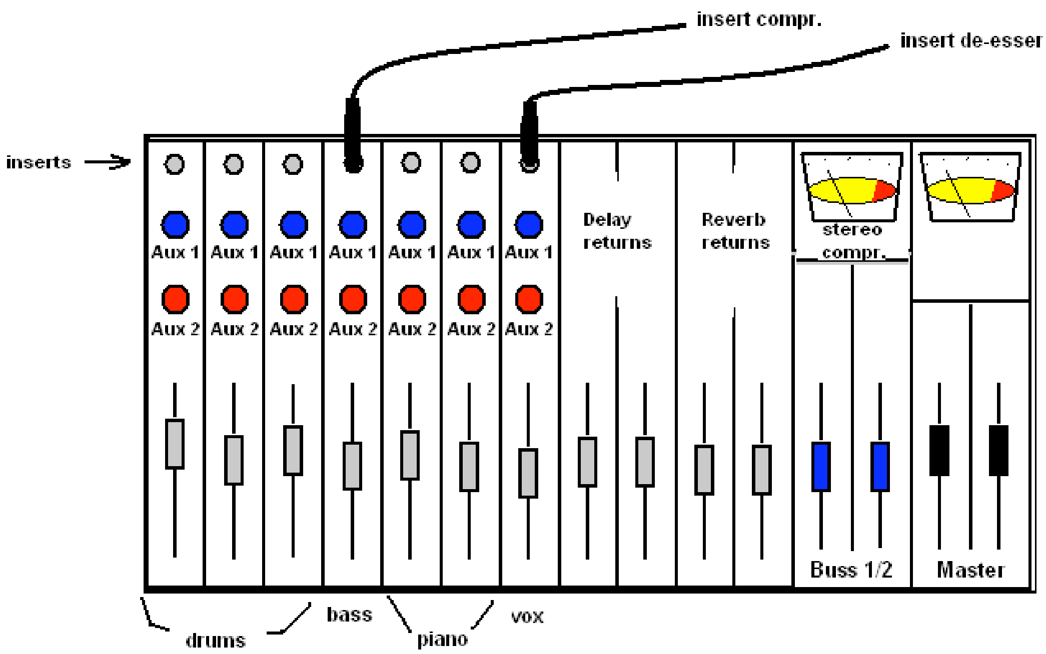


The point is to realize where the signal goes in every case, so that you know where to find each adjustment for each track. Keeping your levels consistent is also extremely important.

It would be poor technique, for instance, to send too little level to the effects devices (from the aux sends) and then make up the missing level by cranking up the aux return faders. This would amplify both the low level noise sent out by the effects units which are not recieving an optimal signal, and could add amplifier noise from the return channels themselves.

Important note: I like to use faders for aux returns, it's just a more "hands on" feeling than a little aux return knob. Just be careful NOT to ever turn up the aux send on those return channels! You will create a potentially speaker- and eardrum-popping feedback loop!

Back to our mixer:



Strapped across the Buss 1/2 faders is a compressor. Into this buss go both drums and bass – this is a great way to get the drums and bass to blend well and create a rhythm section sound. The compression of these instruments together should be kept mild, but enough to control any strong peaks within the buss group. This help to keep rhythm section dynamics from also compressing down the vocals along with them.

The only thing left for you to do is to get your basic stereo mix level and begin making fine adjustments. The most common way of building the mix is starting, as many great mix engineers like to say; “from the ground up”. Start with a good kick/bass relationship. Then add the drum kit’s other pieces. Make sure this relationship is groovin’ . this is the backbone of your mix. Next, bring in other accompaniment instruments (in this case, piano). Make sure that you are leaving enough ***headroom*** (space below 0db-VU or –dFS) to be able to add further parts.

If you run out of headroom, it is a good idea to build up from the bottom again quickly, making sure that any fader level changes don’t adversely affect your careful compressor settings!

Once all your levels are good, decide on how much effects/ambience you want in the mix, and adjust the relative amount (aux sends on individual channels) and the master effects levels (aux return faders). It is quite helpful at this point to switch monitors, listen in mono, or take a rough mix out to the car.

The most important volume level decision to make is the level of the lead vocal. Getting this level right is a matter of practice and much listening to rough mixes in other environments, especially when you are working in a less than optimized acoustic space. Common industry practice for those who feel that they are still unsure when the clock is ticking is to run a couple of extra mixes, one with the vocal up about 1dB, another with it down 1dB. Then you can listen and choose later.

The real way to pan

It is possible to **pan** instruments (place them somewhere left to right in the stereo field) in three different ways, one simple, the other two more complicated, but far more effective.

The simplest solution we have is the relative control of volume between the left and right channels. This is the simple 'pan pot' control found on all mixers. It is effective to a degree, but nowhere near as convincing in creating depth and space as the next two methods.

The second possibility is to use a stereo mic recording of an instrument, and pan the left and right sides of the recording somewhat skewed. For instance, pan the left channel hard left and the right channel to center. The differences in the mic signals will enhance the depth of the sound, and the panning scheme will locate the guitar in mid-left field.

The final method returns us to our old friend, the digital delay. This is also a fairly processing-economical method on the modern DAW – it requires only a very short (maximum 50ms.) delay.

To set up the digital delay pan, place your source instrument on the left channel, and a copy of that track on the right channel (You can also just send the output of the delay, set first to 0ms., to the opposite side). The sound will appear to be in the center.

Now, add a delay to the right channel, and increase the delay time by a few milliseconds. The image will take a different position, dependent on this delay time. Listen to these examples:

A percussion loop volume-panned 85% left ***(.wav file EX50)***

The same loop panned 100% left, with a signal delayed 4ms. in the right channel. Notice how the pan position appear similar, but is much more lifelike and 'spacious' ***(.wav file EX51)***

The possibilities are interesting, considering that many DAW's allow us to automate parameters such as delay time...

More mixing tips

Let's look at a couple more important points which initially took me a little while to understand. Maybe I can save you some time and frustration!

- **Pre/post fader aux sends**: A pre-fader aux send will maintain it's set level, no matter what you do with the fader. If you lower the channel fader completely, the aux send will still send audio to the reverb, delay, etc. that you have connected. As a result, you will still hear the reverb, delay, etc. coming from that channel. This can be useful for creating ambient effects when dropping out parts (reverb from background vocal is all you can hear...) or if you wish to simply maintain the relative levels of effects when making fader level changes.
  More common is the post-fade aux send, which will reduce the send level when you reduce the fader level. This is important if you wish to fade out a part and not still have it's effects returns sneaking into the mix! Keep this in mind if you're going crazy because you pulled down that tambourine fader – but it's still in the damned mix!!! I've pulled out a few hanfuls of hair like this when overwhelmed with a new console in an unfamiliar studio.

- **The Frank Zappa drum mixing technique** : I promised to explain this before, so here goes :

1) **Get your basic drum sounds EQ'd to taste.** Then get your relative levels set. Again, build from the ground up.
2) **Select both Buss 1/2 and Buss 3/4 on every drum channel.** Don't send them to the stereo output (master fader), they'll get there through the busses. Frank was rumored to have used this idea on just the toms and overheads, but you can do as you like!
3) **Insert a stereo EQ on Buss 1/2, an EQ and a compressor on Buss 3/4.**
4) **Set the Buss 1/2 EQ to boost +6dB at 10kHz and give it a little bump at 2-3kHz with a Q of 1.0 or so. Cut 5dB at 400 Hz, also about Q 1.0.** This is your "crack" group – the drums will have definition and clarity here.
5) **Set the Buss 3/4 EQ for a nice bump at 60Hz and 120 Hz. Keep the Q fairly narrow. Then add some low shelving from 200 Hz with a little boost.** Run this into the compressor with a slow attack

and medium fast release. Now, when a drum is hit, the compressor allows a low end blast through before clamping down on the volume – this is your “Fat” group. adjust attack time to taste and to fit the song.

So now as you mix, you can easily balance your drum sound between “Fat” and “Crack”. This is especially useful when you have a very instrument-heavy mix. As you add in instruments, you may notice that the drums need some minor adjustments. Instead of tweaking every single drum channel, you can make easy, balanced moves with just a few faders. In this example, I wish I could go back and just add a touch more bass and less drums, but otherwise you can hear the snappiness this method brings ***(.wav file EX_DRUMSNBASS).***

Keep in mind that if you do this with plug-ins, you will need to consider the latency of the plug-ins on the busses! Some DAW’s do this adjustment for you, but listen carefully for phase or delay problems between the two drum subgroups.

- **Create a soundstage**. This is a very important general decision in mixing. Be sure that the artist(s) have a strong say in this process, and do some serious listening to recordings which you know and respect to get ideas.
  The goal is to create an acoustic environment within your mix. Instruments should have a sense of space and distance. Choose a basic ambience and apply it to key instruments. Always compare with recordings you like. Refer to the effects chapter for help in making adjustments to effects units. Try to create a cohesive soundstage for the music which unifies the instruments. Sounds which don’t fit in will stand out – for better or for worse.

- **Bit rate questions**: Your final mixes will eventually wind up as a 16 bit CD, so you should mix to 16 bits and dither and all that, right? **WRONG**. The conversion from 24 to 16 bits is a ***Mastering*** process. We will get to that in a minute, so leave your mixes as stereo 24 bit audio files. You should ask the mastering engineer which file format they prefer and bring them the proper files.
- **To compress or not to compress**? This is the question. It is pleasant to listen through a compressor while mixing. It gives you a sense of what the final product will sound like. The downside is that uncompressing the mix, should you want a different style of compression when mastering, is nigh on impossible. There are two

possiblities to overcome this problem and maintaining a bit of flexibility during mastering. Neither takes much extra time.

1) Monitor through the compressor as you mix, and then disconnect the compressor when you run down the final mix. Be sure to note (or save in software) the compressor settings and bring them to the mastering session. This allows the mastering engineer to know what you really want, but allows him or her the flexibility to make their fine-tuned decisions as to how to make your mix sound best.

2) Simply run down two mixes. Make sure you label or name them appropriately. Run a compressed one and an uncompressed one. It is still likely that a good mastering house will have a much better compresser than you do, but sometimes those moments just happen, and your compressed mix will be "The One".

A Mixing Checklist

There are so many little details, so many little dials, lights and knobs, that you may want to print this list of questions to ask yourself and keep it handy when mixing:

- Did I listen to each track for noises or glitches? Fix these before starting.
- Did I disconnect microphones or unnecessary devices from mixer inputs to avoid unwanted noise?
- Did I EQ individual tracks to my liking?
- Did I compare my sounds to pro recordings/favorite recordings?
- Did I build my mix "from the bottom up"?
- Did I leave enough headroom as I went along? If not, start over and get it right.
- Did I consider my "soundstage" and pick fitting effects?
- Are any instruments overcompressed or over processed?

Now, get a rough mix happening and then:

- Listen over different monitors/systems.
- Consider the lead vocal level.
- Take a break for lunch/dinner/midnight snack, then come back and listen again – and make comparisons to other recordings.
- Be sure of all the parts and arrangements. This is the last chance to change anything.

- **VERY IMPORTANT: Don't do any fadeouts at the mixing stage!** Leave the fadeouts for mastering**.** In the digital domain, a fade does a LOT of math on your signal, save this math for the very last step, so it doesn't ruin your resolution.

Time to run the final mix:

- Am I using 24 bit resolution?
- Am I saving the mix in a folder/directory where I can find it again?
- Did I turn off dithering plug-ins?
- Did I decide to make a mix with master compression/limiting/EQ, and if so, did I make one without?
- Burn a CD, take 10 and listen on another system
- Crack a cold one, you're done for today.

Mastering Tips

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Anyone who is not a professional mastering engineer – and I'm not – knows they are getting into hot water by saying that they can give advice on the subject. The Pro Mastering Engineers are like Illuminati; they may walk among us every day, but they hold secrets that none of us will ever comprehend fully.

I will, however, give up my few accumulated secrets, quote a few mastering Masters, and help you to avoid some of the pitfalls of self-mastering (master-bating?)

We have to start off by assuming that the production, recording, and mixing have been carried out to the best of your abilities. After all, "Garbage in, garbage out." You have a 24 bit stereo mix with no extra compression, limiting, EQ, dithering, analog-tape-hype-simulation, and so forth.

To get a perspective on what mastering is, we should think about the original purpose of the mastering process – to create a final version of the recording for mass production (master) which will translate well to all possible listening systems. Originally, this involved filtering off low frequencies and dealing with phase problems which could cause a record player needle to jump the groove. This slowly transformed into a need to have one's record seem louder on the radio, to grab the listener's attention (Motown!). The current state of affairs is to produce masters with as much low and high end frequency range as possible (for the big pricey systems) while still sounding great on podcasts, MP3's and sattelite radio!

The most common misconception about the mastering process is that it is only about making things louder. This is only partially true. The fact of the mattter is that records which are consistently too loud cause the listener to lower the volume to avoid the pain threshold, contributing to the "giant little sound" phenomenon. The giant little sound is a recording with high level but no dynamics. It is technically possible for anyone with a small batch of modern DSP plug-ins to create a master wherein the entire song is cranked to precisely 0dBVU – you can't get any louder.

This 0dBVU cranked-up master will likely be completely unlistenable. The dynamics will be crushed, creating a wall of noise which vaguely resembles the original music.To listen to the music at all, you would have to turn it way down – the "giant little sound".

There are a few basic things you MUST do to get decent masters from your mixes:

- Let someone else do the mastering! I know, we all like to be in control of our pet projects from beginning to end. What you really need after all that tracking and mixing is a fresh set of ears. If you can't afford a pro mastering house, then see if you can talk a friend or colleague engineer into giving it a try. Objectivity is your friend when it comes to the final stage of the production.

Then, if you choose to master a project:

- Listen in the most neutral monitoring environment you can gain access to. This is not the time to switch listening environments often, as can be helpful in mixing. You want to build consistency from song to song and create an album out of multiple songs. Predicatbility of monitors is key. You may even be better off working in a large living room on an audiophile kind of stereo, rather than in a small tracking studio. The reflections off the console and walls in a studio can be poison when mastering.
- Maintain the highest bit level possible through the mastering chain, and dither ONLY at the end, and ONLY when changing bit rate. This is the same as I have recommended from the very start, by changing bit rates in the digital realm, you are subjecting your music to a lot of math. Avoid this as long as possible.
- Use the most stable master clock you have available. Some high quality converters available (such as those from Apogee) can provide you with an extremely stable master clock, providing you with a low jitter transfer of your audio all the way through the mastering process.
- Use multiple tools for metering. A VU meter, a dBFS meter and a spectrum anaylizer are all helpful in the visual aspect of mastering, and all tell you different things.

    1. **The VU meter** tells you how loud the material sounds. Since it responds to average (no peak) levels, it gives you a good idea of where your loudness sits. If you can get the VU meter hanging around the top of it's range without sounding overcompressed, you're doing good!

    2. **The dBFS meter** doesn't tell you much about the average level, since it responds so quickly to peaks – but that's what it's there for. In the digital age, going over 0dB causes ugly

digital distortion, not smooth analog overload. The dBFS meter tells us if we are in the danger zone.

3. **The spectrum analyzer** gives you a visual representation of the frequency content of the audio material. It may help you top better identify problem frequencies until you gain your golden ears, or if you are in a less than optimal mastering environment. Listening to commercial recordings through the spectrum analyzer is an excellent way to get an idea about the frequency balance of pro recordings.

- **LEAVE HEADROOM IN THE MASTERS**! There are far too many annoying sounding recordings out there which are compressed and limited into the top 6 dBFS of the available dynamic range. You should read Bob Katz's tips on mastering at www.digido.com – he is one of the industy's best, and he makes a very strong argument against overcompression and low headroom! Try to leave a few dBFS headroom for the occasional peak, and have most of the material kicking around -6dBFS.

- **Digital limiters are our friends**. The excellent L1 and L2 Ultramaximizer plug-ins by waves, as well as many other imitators out there do an excellent job of allowing us to increase the average level of masters with no audible artifacts. Just start be setting the limiter to cut 3 to 6 dB off the peaks, and you will get an immediate loudness boost without degrading your signal much at all.

- **Do not assume that you need any particular processing**. Yes, it is likely that the mix will need a little EQ, and maybe even compression. But don't start with some "swiss army knife" setting and start mangling the mix. I have made this mistake myself too many times. "Oh, this EQ and limiter sounded good on the last song, so I'll start here!" Bad idea. Always listen to the song through, and then imagine what could make it sound better – fully, more intimate, more dynamic, etc. Whatever the *song* calls for – not what the mastering engineer would like.

- **Use the sweep and boost EQ method**. In the same way we EQ'ed individual sources, you should identify potential problems in the mix to be mastered. But don't assume that you need EQ, you may not if you did a well balanced mix. If it ain't broke... And if you do EQ a whole mix, try to keep the adjustments small – remember the EQ now affects ALL instruments equally. Cutting out a bad bass resonance is going to make the vocal sound thin. If you're doing all your work in-house, consider a remix if you have big EQ problems.

Major TRICK: A cool EQ method is to use a parametric EQ instead of high shelving. Sweep all the way up to 20 kHz, and set a fairly wide Q (0.5 to 0.8) then boost about 3dB. This gives a steady slope up in the high frequencies, and sounds better than a regular high shelving. Compare the high end to commercial recordings. Boost up to 6dB if you think itstill sound O.K. – yu can boost quite a bit like this.

- **Use compression wisely**. Here is where a **multiband compressor** can really shine. You can get the bass end pumping without crushing the highs. The key is to identify the different frequency bands and set the *crossover* points properly. The crossover point is generally where the fundamental frequencies of the next higher group of instruments lies.

  Let's take our earlier example of drums, bass, piano, and vocals. The low drums and bass guitar will live below 200 Hz. Piano may also play some low notes, but will likely play chord inversions with a fundamental above 200 Hz. That would make a good first crossover frequency. The range of the piano fundamentals will likely not exceed about 1kHz, so this may make a next good crossover point. A lot of vocal energy will reside in the 1kHz to 8kHz range, and cymbals will have their high frequencies above 8kHz, making a last good crossover point. You can then fine tune these ranges while listening. Keep in mind that a faster attack and release may be necessary in the higher frequencies, and a slower attack and release in the bass. You should take a look at some of the devices preset settings and even audition them to get a feel for how the complex M.B. Compressor works

  A good old **stereo compressor** is not to be ignored! **Remember, with the right settings, you can a stereo compressor to give a mix some "oomph"!** The way in which a stereo comp moves the dynamic of the whole mix at once can really make a master dance and pump with the beat. **You need to start by finding the beats on which you'd like the compressor to work**. Set a high ratio, like 8:1, the attack time fairly short and the release medium. Lower the threshold (starting from zero) until you see the meters reacting to the beats you'd like to emphasize.

Adjust the release so that the compressor recovers (the gain reduction is less) before he next big beat accent of the song. If the release is too long, you'll squash the overall dynamic. Too short and you'll catch and kill all the nice little accents.

Then, reduce the ratio to 2:1 or less. Now, adjust your attack so that you allow a good "oomph" of beat through before the compressor starts working. The effect should be subtle now. Bypass the compressor and listen for a few minutes. Now switch it back in and listen. Do you like it? You should also be able to increase the average level another few dB after the compressor.

- **The final step should be to dither your signal down to the proper bit rate.** This will most often be 16 bits. Apogees UV22 process is really nice sounding in my opinion. If you have multiple dithering options, it is worth mastering through a few different ones and taking the masters around to a few different listening environments for a critical listen.

  Check your sample rate, too. Make sure your output file will be "16 bit 44.1kHz" for CD production. I have often made the mistake (easy to do with Protools) of dithering to 16 bits, but saving the output file as 24 bits! Du-oh!

The “In-House Master” checklist –or- Mastering for Dummies (dumb enough to do their own mastering!)

For those of us who can’t (or won’t) afford to let a pro do their mastering, here is a list of things to do, and the order in which to set them up, to do what I like to call an “in-house master”. It won’t be as good as the pros, but will do a nice job of shining up an indie record or a demo to make it a little more competetive sounding.

1) **Set up your mastering environment**. Quiet, calm, no disturbances. Best monitors you can get your hands on. Set the volume to about 80 – 85 dB, no more, no less. try not to have the monitors set too close to walls or where you’ll get an immediate reflection off of a mixing console. Mastering at home is a compromise.

2) **Make sure you’re using the right files**. There are often many mixes lying around. Get them all organized and lined up in your DAW or on DAT, whatever, before you start.

3) **Once you go digital, stay digital, and use the best master clock that you have.** Dither only at the very end. Check to make sure that any digital processors you have in the chain are not dithering and/or changing bit rates at their outputs. This is about the 100th repeat of this advice, but you really can’t afford to forget it now!

4) **Set up your mastering processors in the following order:**

- ***Limiter***; set to catch a 3 -6 dB peaks and give you a little more level.
- ***EQ***; use the boost and sweep method to find, don’t EQ more than a few dB.
- ***Compressor*** (multiband or stereo, don’t kill your nice dynamics!),
- **stereo imaging stuff**; such as “Spacializers” and so on. Only if you like it or feel you need it, this is the place. Don’t overdo these, they can sound cool at first, but stupid later - and can ruin your mono compatibility and general clarity.
- **Dither**; to 16 bits with your best dither.

5) **Do the song fade-outs, if necessary.** You may also need to plan in the amount of silence you need before tha next track will begin if you don’t have a CD burning program which allows you to adjust this value!

5) **Save your file in the proper format and bit rate**.

6) **When you have all your masters done, you need to sequence the album.** This is not as fun as tweaking and juicing up the mixes, but is just as important. You will need a dedicated CD burning software which allows you to adjust the time between tracks, or have already planned the pause into the master... **Always listen from one song into the next to decide the relative volume of the tracks**. And don't change your monitor volume! Be aware of the style of each song – with the big race for louder masters that is going on, don't do something stupid like cranking up a ballad so loud that the following "rocker" of a song sounds quieter! I beleive that this is still punishable by tarring and feathering in 32 of the 50 states –know your local laws!

7) **Save all your mastering settings!** One of the reasons that Mastering Versions of many analog devices are so much more expensive than their studio counterparts (take a look at www.mercenaryaudio.com) is the stepped switches built into them. This allows exact repeatability of a mastering setting. It may happen that an artist is later unhappy with an arrangement and cuts a section from or adds a section to a song and needs a re-master. The artist may also just want a minor addition of low end and is otherwise happy. You never know - you'll be happy you saved those settings!

Listen to these examples to get an idea of the progress from mix to master. ***MASTER_EX1.wav*** is an excerpt from the mix with no mastering processors added.

MASTER_ADD_LIM.wav adds the Steve Massey limiter to trim a few peaks.

MASTER_LIM_COMP.wav includes the limiter and the Pro Tool native compressor to raise the average level. Notice the way everything suddenly sounds 'fatter'.

Finally, I make a small correction in the low end to compensate for the compressor bringing up the lows, and add a little wide midrange boost and a touch of highs in file ***MASTER_L_C_EQ***. Ready for CD!

Final thoughts and Words of Wisdom

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Above all, the musician's performances are the stuff that makes music sound great. The engineer's responsiblity is to accurately capture these performances and aid in acheiving a tone that supports the performance. When an engineer cooks up a great sound for a track, this becomes a part of the overall performance.

The inspiration living in a fuzzing, feedbacking guitar tone is a good example. Listen to Jimi Hendrix's volcanic solo in "Foxy Lady" – do you think he would have performed it the same way if the outrageous tone did not work with him?

Keep this in mind as you record performances, whether you are the musician as well or just engineering on someone else's song.

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When mixing, keep an objective view of the song. Every song has some kind of message, and the mix is the stage on which the recorded performer stands. The overall vibe created should support this through proper volume balance, choice of ambience and effects, and an interesting arrangement.

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Mix with your ears and your hands, not with toys, boxes and effects. The bells and whistles have their place, but they come second to a proper balance of sound. Always try to mix without radical EQ and compression – don't squash and mangle the musician's performances right from the start. If this is an artistic statement you are reaching for, make sure, again, that it serves the song.

The best mixing engineers in the business start by pulling up the basic tracks, listening and then setting relative levels. The best sound is a natural one, and a good natural sound can only be enhanced with effects and processing. Get your sound happening by starting with those beautifully recorded tracks of yours, and if you wander off track while mixing, don't be afraid to go back to start and begin again.

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Always make backups. Always make backups. Be repetetetive – when working with digital systems, music can become irreplaceably lost. It

WILL happen to you, I promise. So be prepared and when you ask your computer to open up the project and it says, "Project? What project?", you won't wind up doing expensive damage by kicking it's little silicon brains in.

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If you're running your own little home studio, use those rainy days when you want to do something but are burnt out on the music to:

1) **Clean up wiring and improve connections**. Don't forget the coils of cable are antennaes and that cable runs along AC chords create noise. Also, think of connections which you have had to make repeatedly and see if you can wire them more conveniently.

2) **Build yourself some kind of improvement**. Check out http://www.johnlsayers.com/phpBB2/index.php for some project ideas. Great site with some cool projects.

3) Grab yourself some new ProTools plug-ins for free at http://www.masseyplugins.com/ can't beat the price, and the full versions are very inexpensive. Way to go Steven!

4) Read up on some of the best production and arrangements ever sone at http://www.recmusicbeatles.com/public/files/awp/awp.html . This series is at times a bit too analytical, but it draws attention to some production and arrangement ideas which may change the way you listen. Also, Alan's repeated reference to the wisdom, **"Avoid foolish consistency"** is not to be underestimated – in any style of music.

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Thank you again for your purchase of this book! Please keep an eye on www.joedocmusic.com for the dates of each update to the book. In the coming months I will be creating many more audio examples and revisions/additions to the text.

I am also glad to listen to your mixes, masters, questions, complaints, and most of all – praise ☺ at harrykipper@yahoo.com or through the website. Happy mixing!

Appendix 1: The guitar files

(listen to files page)

Alright here are the solutions to the guitar tones:

#1 is EX26
#2 is EX24
#3 is EX30
#4 is EX28
#5 is EX25
#6 is EX27
#7 is EX29

It's amazing the way simple tricks can create such a variety of sounds. Not all of them are top notch in terms of clarity and tone, but sometimes a full range tone will eat up too much space in a track, and a little sound that's been run over by a few trains can really fit the bill.