Producing Drums

Drums are the fundamental component of music; “The Bed Track” as we call it. All other instruments are recorded to the rhythms of the drum performance. All instruments eventually trace their roots back to Africa where rhythm was essential to the spirituality of the tribal culture. To this day we notice the rhythm of the drum from the native peoples of Canada, to a dance hall, to someone just tapping their feet to a beat. We are all intertwined with rhythm.

Recording drums is one of the most challenging situations you will ever come across in the studio. The microphones you use and how you place them is important in the initial preparation. A tuned kit with new heads is the standard operating procedure for all drummers. No amount of EQ or signal processing can resurrect a cardboard sounding drum kit after it's been recorded. The room you are recording in is very critical also. Rock prod/eng’s prefer large live rooms to capture the ambience of the kit. As for pop drummers, they are often located in booths or small rooms that are reasonably dry sounding so you can get a present sound.

The Kick Drum

For the kick drum, you should use a dynamic mic - the bigger the diaphragm, the better low-end pick-up. I've used RE-20's and AKG-112’s. The way I usually mic a kick drum is to stick the mic inside the kick drum (I always remove the front head) about 3-6 inches in front the front head. That's a good place to start - you can move the mic around and find the best sounding location. I usually stick a pillow or blankets against the rear of the drum head to minimize ring. You'll get more attack the closer the mic is to the beater - you'll get more overtones farther away. On certain occasions we use two mics. A small diaphragm dynamic (421) close to the beater to get the attack of the bass drum. The other mic, a large diaphragm, further away from the head to pickup the low end of the resonance of the bass drum. This allows you to have the flexibility to control the mix of the attack and resonance. With the close mic you can EQ from 2-4Khz to get the attack. Anything higher than 4Khz will just make the attack sound thin. With the close mic try to avoid aiming it directly at the beater. This prevents dramatic changes in the attack sound of the bass drum. If you place the mic slightly off axis and EQ the mid range (more than you would if the mic were on axis) the attack sound of the bass drum will be more even. With the large diaphragm mic place it closer to front of the drum (where the head has been removed) this will allow you to get more of the low end resonance. If using a large diaphragm condenser make sure to pad it down (use pad on mic) and place a Kleenex over the microphone to prevent the capsule from being overloaded by wind. With EQing the bottom end you need to know if you would like the bottom end to be heard or felt. EQing between 30-60Hz will allow you to "feel" the bottom end only on large speaker systems. If you need to hear the bottom end EQ between 60-100Hz. This will allow you to "hear" the attack of the low end on smaller speaker systems. The bass drum also produces a lot of low mid range frequencies that tend to not relate to themselves. This usually occurs between 300-600Hz. Be prepared to remove some of these frequencies, which will allow the bass drum to sound tighter and punchier.

As in any situation using two mics you need to be prepared for phasing problems. This problem can be solved by flipping the phase on one of the mics or moving the position of one of the mics.

The Snare Drum

The best way to capture a great snare sound is by close miking it with a dynamic cardioid-pattern mic that can handle a high SPL and keep leakage to a minimum. The legendary SM-57 is an excellent mic for the snare, it is the choice of many professional engineers, and it's what I use myself.

The classic approach for miking the snare is to place the mic 1-3 inches over the snare rim opposite the drummer and 1-3 inches above the top drum head. The mic should be at about a 35-degree angle downward. I usually try to also angle the mic inward (away from the hi-hats) to avoid leakage from the hi-hats. You can also mic the bottom snare head for some added top end but remember to reverse the phase. If the drummer is playing with brushes try using a small diaphragm condenser cardioid microphone. The condenser will have a larger pickup pattern to capture more of the performance. With EQ the snare drum has three basic regions: Low end 100Hz (depending on depth of snare drum), mid range (crack) 3-5 kHz and top end 10 kHz and above. In rock, snare drums you tend to desire a lot of the mid range/crack and low end. In Pop you tend to desire more of the top end over the mid range.

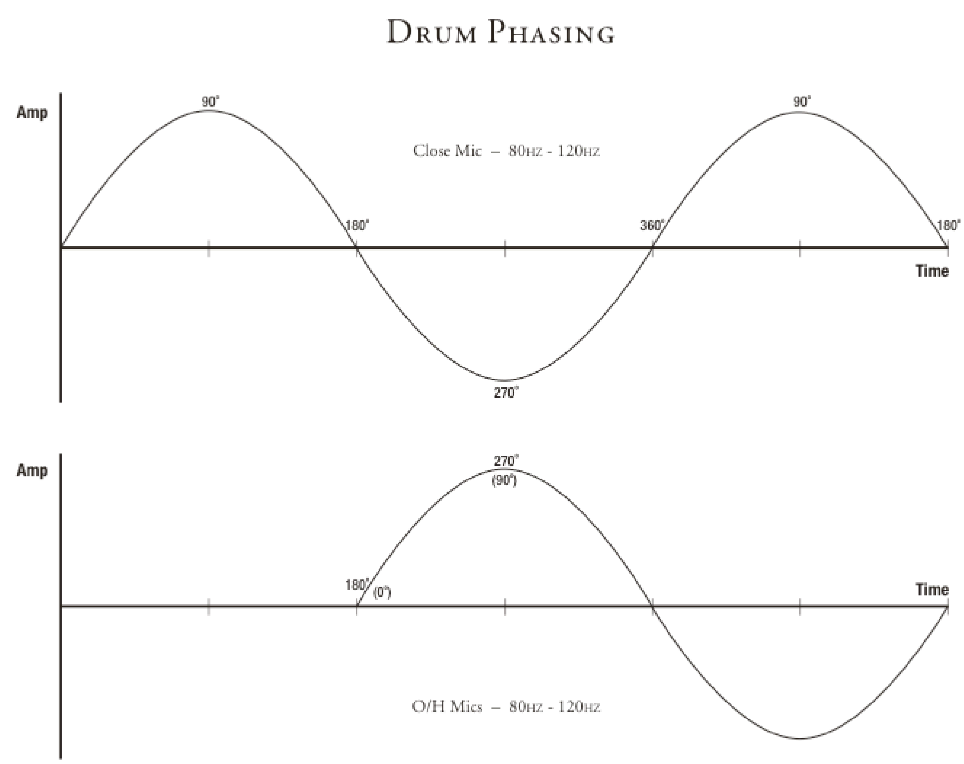
The Toms

It’s best to mic each tom separately. Again, small diaphragm dynamic mics work the best, and SM-57's and Sennheiser 421's are a good choice for their tight pick-up pattern and high SPL. Small diaphragm condenser mics are great for getting more top end but remember to insert a pad so the mics don’t overload and be prepared for cymbal leakage.

The best approach to miking toms is to place the mic 4-6 inches above the drumhead at about a 45-degree angle over the head. If you pick up a lot of overtones, a little duct tape in the right spots will kill the overtones, or if you have a noise gate you can gate out the overtones. I always move the mics around to capture the right balance of attack and resonance. In EQing toms there are four different ranges: Low end 80-120Hz (depending on size of tom) Low Mid range 300-600Hz, High Mid range 2-4kHz and Top end 10kHz and above. In an average EQ setting on a tom you would see a boost in the Low end, High Mid range and High end as well as a cut in the Low Mid range.

The Overheads

The drum overhead mics are really supposed to capture the overall sound of the drums, not just the cymbals. Condenser mics such as U-87s and AKG 414s are the first choice for overheads, and one popular miking technique is with a spaced pair of mics (on boom stands) mounted 2-3 feet above the drums -the right mic pointed at the right cymbals, the left mic pointed at the left cymbals. Remember that when raising overheads the acoustics of the room will factor into the sonic equation. When mixing direct drum mics with the overheads this most likely causes acoustical phasing problems, this happens in the low frequency range. The low frequencies in phase with the snare drum mic have a tendency to be out of phase when the overheads are mixed in, due to the wavelength of low end frequencies. When checking for phasing problems on drums assign all mics to a mono listening position. It is hard to detect phasing problems with mics panned to different positions in the stereo image. If you notice phasing problems just reverse the phase button on the input strip or move the mic positions. EQ overheads if you need a brighter sound and insert shelving curves in the high end try to avoid rolling off the low end, for this will make your snare and toms sound thin. E.g. +3dB @10Khz (shelf).



The Hi-hat

Use a small diaphragm condenser mic like an AKG 451 placed about 6” above the high-hats, pointed straight down at the center of the top hat. Sometimes high-hats have a tendency to produce unwanted midrange frequencies around 1.5 kHz which tend to make the high-hat sound trashy. Omitting some of this frequency range will allow the high-hat to sound more defined in the high end.

Room Mics

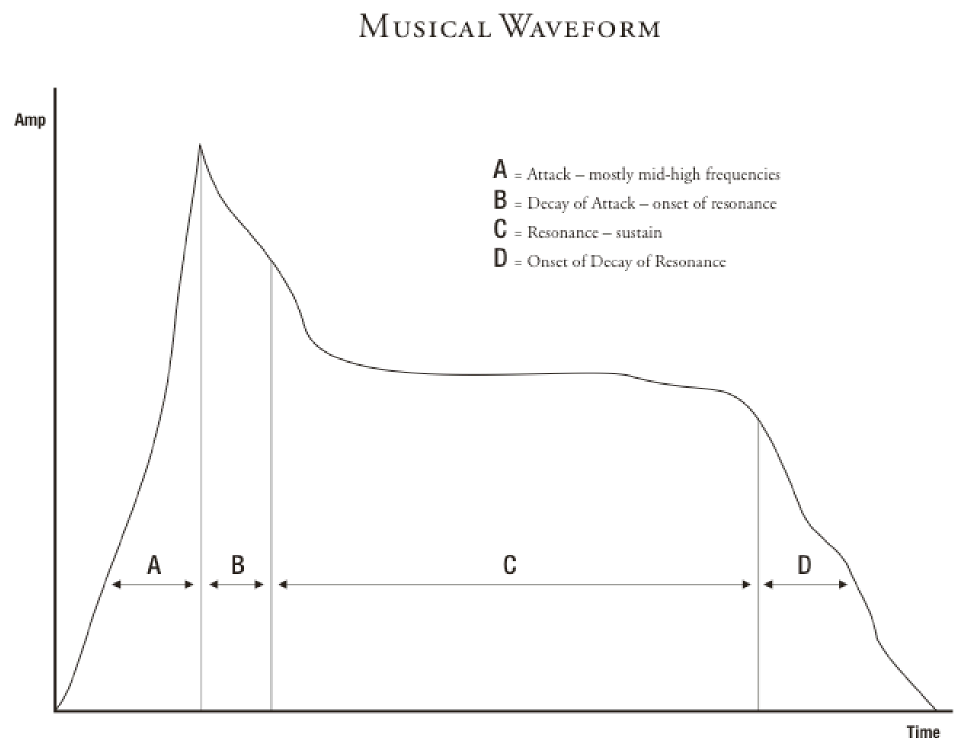
Use at least 2 omni mics of the same model. Place them in the centre of the room to get an even room sound. This often requires the use of hard surface baffles between the room mics and the drum kit. This removes the initial direct sound in the pickup allowing the engineer/producer to utilize more of the room resonance.

Drum Compression

Drum compressing and limiting is often used to control dynamic problems and/or create a desired effect. When using a live performance you tend to get excessive dynamics. For example: when the drummer hits a kick drum and crash cymbal on the downbeat of a chorus, even though the transient is of short time duration it will limit you into how much level you can translate to a CD in mastering. This transient causes the drums to separate themselves from the rest of the elements in the mix. Because the duration of the transient is so short it is hard to correct this dynamic problem through manual fader riding. A good solution for this is to bus all the drums to two tracks and bring this two track stereo sub-mix of the drums back into more inputs. At this stage you can insert limiting to control the extension of the transient. You will need to incorporate an attack time of less then 1millisecond due to the transient nature of the drums. The release time should also be very fast 5-10ms so the only transient is affected and the rest of the performance is left untouched. A limiting ratio of 10:1 or higher will suffice. Remember to allow headroom so some amount of the transient will pass through, rather than being hard limited. This is accomplished by first setting a limiting ratio, with a fast attack time and a fast release time. Next, set the threshold to a setting where the limited audio information is approximately 2-3ms in duration for the nature of drum transients is a very fast attack and a very fast release with little duration in between. The goal here is to limit only this fast transient without affecting the resonance of the drum sound.

Another advantageous use in dynamic control is getting your drums to sound punchier. This is achieved by first eliminating the random transients and then inserting compression with a ratio 4:1 to 8:1. The attack time should be any where from 20-50 ms which allows the louder attacks of sound to pass through unaffected. Once the attack is cleared the compressor will kick in, lowering the sustain part of the drum signal. Next set the release time (1-200ms) so the sustain part of the signal is compressed and decays until the approach of the next transient comp/limit. When you are sub-mixing drums to a stereo bus remember to insert the stereo link function on the comp/limiter.

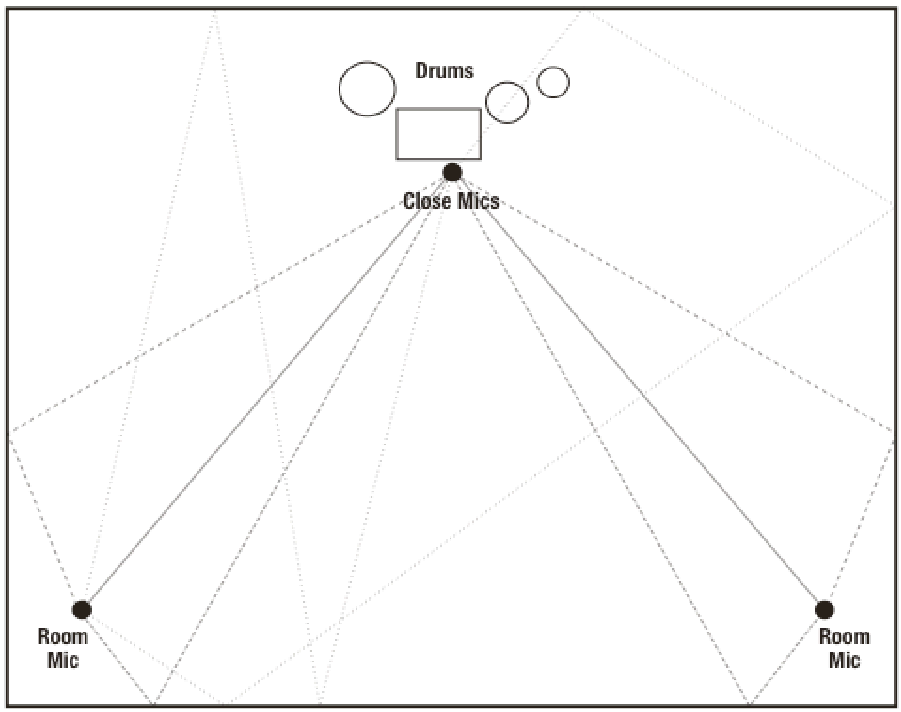
In dealing with dynamic control on separate drums allow yourself to create a certain characteristics to achieve great sounds. With snare drum a common problem is getting a good attack but with no sustain which causes the drum to sound inconsistent and weak. The problem here is that even though the attack of the drum is heard on a consistent basis the length and level of the sustain changes randomly. In dealing with this problem split the snare drum over two input channels. Over the first input try to maximize the transient quality of the snare drum by utilizing transparent limiting and EQ in the mid range and high end. On the other channel first gate the signal so all you hear is the snare drum. Next insert a limiter with a very fast attack and very fast release time. The goal here is to limit the attack of the signal heavily. This allows the sustain to be consistent in level and adds more length in duration. To add more body to the sound, EQ in the low mid range and low end. Now mix in this signal with the more transient snare drum signal which will allow you to add in more body to the snare drum that will make it sound bigger and more consistent. In effect, you are decreasing the dynamic range between the level of the transient nature of the drum and the sustain properties of the drum.



Room Microphones

Use two large diaphragm condensers of the same model placed at an even distance from each other and closest walls. If the room is 36' wide place the mics 12' from the walls to get maximum diffusion. If the room is 48' deep place the mics 18' from the walls. The biggest problem with room miking is the noticeable delay between the audio from the close mics and the original audio arriving to the room mics (diagram A). The solution is to remove the direct signal from the drums from entering the room mics (diagram B). This will allow the room mics to only pick up diffused early reflections and the room reverb, allowing you to mix it in at a higher level without any noticeable flam.

Room Mic P/U – A



Original Direct Sound

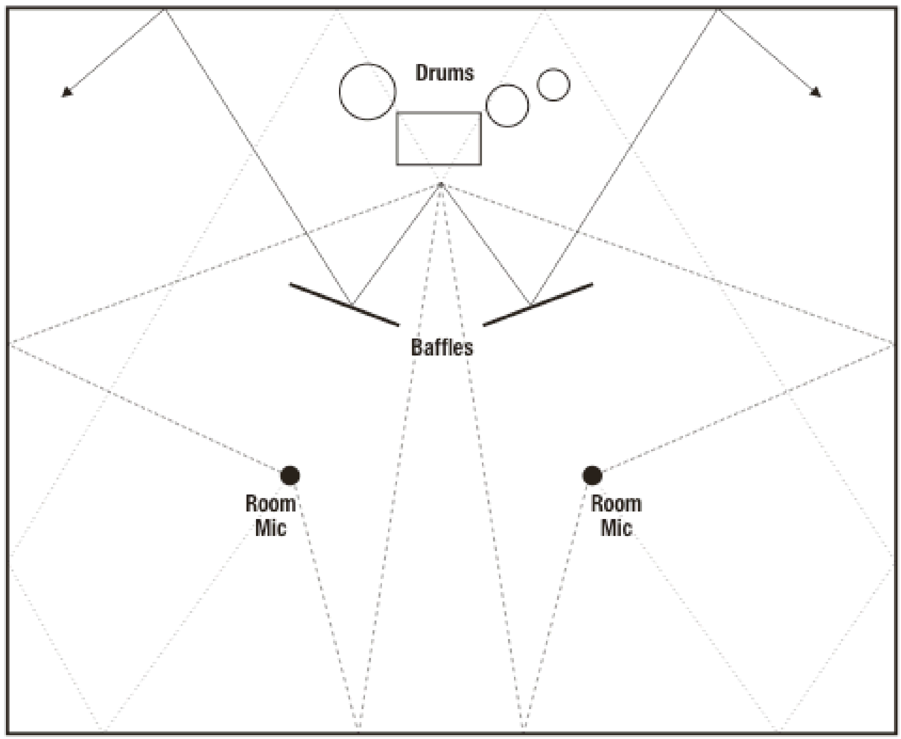
First and Early Reflections

Room Ambience

Close Mics – Cardoid

Room Mics – Omni

Room Mic P/U – B



Original Direct Sound (N/A)

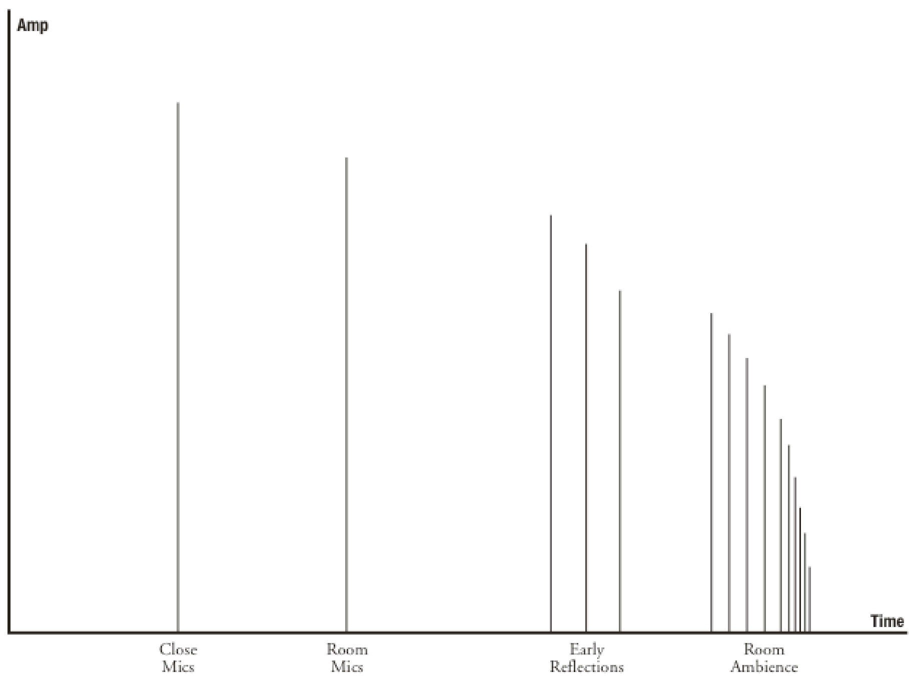
First and Early Reflections

Room Ambience

Close Mics – Cardoid

Room Mics – Omni

Drums


Amp
Time
Close
Mics
Room
Mics
Early
Reflections
Room
Ambience

Producing Bass

One important element of a good production is capturing a good bottom-low end in the recording to help drive the rhythm by complimenting the drum (bass drum) pattern and also serve as the musical foundation for the production. With the great variety of music styles today, there is a myriad of desired musical ideas for the bass sound and its musical contribution. From heavy metal, pop to jazz; all require different approaches. Often synths are used instead of the bass guitar in a track, which requires s a totally different approach and much more flexibility in attaining a different desirable sound. It is important to note that the bass serves two functions. One is for a "*feel*" effect focusing on the 30-60Hz range and the other is to hear the actual musical notation of the bass performance 60-200Hz. In dance and hip-hop music the desired approach is to feel the bass. In pop music it is important to hear the bass, and in rock it is a combination of both. Another contributing factor in bass is the duration of the note which if longer will give you the illusion that the bass is louder in the mix, as in hip-hop music.

Basic Requirements:

1) With Bass you need to know that larger gauge strings will produce more volume and sustain. With thinner strings you might not get enough basic level for the pick-up to grab without producing a lot of noise. Hard rock players often desire thin strings for it allows them to play faster and basically "Show-off". When in the early stages of pre-production have the bass player adapt to heavier gauge strings so the player will be well prepared when it comes time for recording. If the bass sound is squeaky, use a little oil on the strings.

2) Pick-ups: There are 2 different types of pick-ups, active and passive. Active pick-ups require dc power, usually from a 9-volt battery or phantom power. They also step–up the gain and produce a cleaner signal, for you don't need a lot of gain from the mic-pre, which might produce a lot of noise. You need to be careful though, for the amount of level coming from an active pick-up might overload the DI box and/or the mic-pre. With passive pick-ups there is no DC power required and no need to insert a pad on the DI. Passive pick-ups do not color the sound and are less expensive than active pick-ups.

3) Half-Cycle Distortion is common to bass. It occurs when the release time setting on the comp/limit is too short. Since low frequencies have long wavelengths the release time needs to be long enough so it does not release on the half cycle where it would produce distortion. If you need to have a quick release time, start with a long one and slowly shorten its duration until the onset of distortion and then lengthen it slightly.

Hard Rock/Metal Bass

With HR the bass's role is mainly to be felt, rather than hear what the player is playing. A good HR player will use medium to heavy gauge strings and active pick-ups. Most of the time the bass DI sound will not be used for it is too clean and dynamic. However, it is a good idea to record a DI bass for at a later stage you might want to send the clean signal back into an amp for a different sound. A lot of bass players will use a speaker cabinet that has a mid-range speaker (6"-8") and/or a bottom-end speaker (12"-15").

If using the mid-range speaker that supports the aggressive part of the sound, use a Senn 421 for it's good mid-range response and it's ability to handle high SPL.

For the bottom end you might want to use a D-112 or any large diaphragm mic to capture the low frequencies. When using LD condenser mics be prepared to insert the mic pad to prevent distortion. The condenser mic will allow you to pick up all of the speaker's sound due to the fact that cardioid condensers have a wider pickup range than a dynamic cardioid.

With compression and limiting, the desired effect will be to have the bottom end level remain a constant with very little dynamic range, for the bass will be mainly supporting the production as a sonic element. If the player is using a pick you will need to treat the sound so the attack will be noticed, with the sustain part of the sound. If the player uses their fingers the effect will be for support and sustain. A hard rock player will drive their amp hard for its compression effect and edge which helps the bass to sound aggressive and cut through the mix.

With EQ you usually look at the bottom end (30Hz-60Hz), a shelf or wide bell curve will do. Make sure it's giving you what you need for an even bottom on all 4 strings when played opened. Next, a lot of players like to

have some of the low-mids sucked out (300Hz-500Hz) to get rid of the musical part of the signal. For edge, try EQ around 2khz-4khz.
The most common trouble in recording bass is its dynamics. A bass often has uncontrollable dynamics especially from string to string and from a fretted note to an open string note. Usually a medium attack time and medium release time are required. For more of a sustain effect use a long release time. Do not be afraid to overdrive the amp and heavily compres the sound.

Pop Rock Bass

Pop Rock bass requires the most work of all bass sounds. Listeners tend to want to feel the bass and also hear what it is playing musically. A careful balance between the two is sought. One of the elements will be a good clean DI sound usually from an active pickup and active DI box. You must be careful about distortion especially when the player plays quite loud. At the DI box you will split the signal and go into a bass amp usually with a 12” speaker for good low end and a little edge.

Watch out for grounding problems that will produce a buzz/hum and/or potential live voltage. You will have at your disposal a ground lift switch on the power bar and a ground lift switch on the DI box to get rid of the ground problem. Once you have both desired elements from the DI and the amp you will then balance the level of the 2 to achieve the desired sound.

When recording make sure you try to keep the 2 signals separate, for when mixing you might prefer a clean bass sound for the verses and an edgier sound for the chorus.
With EQ you will add low end around the 80Hz-150Hz area and use a bell curve. Often I roll-off the lower part of the bottom end for it is not needed and might cause problems later.
With the musical part of the sound, EQ from the 400Hz-800Hz range with a bell curve with a medium Q (2 – 3:1 ratio). As for the attack part of the sound 2khz –4khz should do.
When getting your balance play with the phase reversal on one of the signals, you might be pleasantly surprised.

For mics I would suggest a good large diaphragm condenser or LD dynamic or both. I have often had success putting the condenser right on axis with the speaker to get presence or to the side of the speaker to get more of the warmth. When compressing, use 2 comp/limit that are the same model and link them together. By linking them together, the bass dynamics get treated the same and the mix balance between the DI and the amp remain the same. Use a medium to fast attack time and a medium to slow release time. EQ before you compress, especially if you are treating the low and bottom end. You don't want to drive the compressor crazy with a lot of low end and unwanted frequencies.

Jazz Bass

Jazz bass is recorded with a large diaphragm condenser mic. A dynamic mic doesn't have the high-end response you need to capture the finger playing of the bass. Be careful on the mic distance, for if the mic is too close you don't get an overall approximation of the sound of the bass and you might run into proximity effect problems. A lot of players prefer not to EQ the bass and use a minimal amount of dynamic control; instead opting to work on finding a good mic and placement. Sometimes players will have a pick-up on the bridge of their bass for helping with isolation. I tend to not use this for I have not heard a good sounding pick-up yet

Synth Bass

Synth bass is used a lot these days for it offers the most flexibility. You can expand the musical range of the bass and also have greater control over it's attack sound, its sustain and the ability to control the differences in dynamics. If a hard rock band decides to use the key of "D" or "C" you can use a bass sample to play in that range with good sonic ability.

In house music and rap music the bass is usually programmed to be in perfect time. I often like to put the synth bass through an amp to get a little more edge. Often you will only need a little EQ and little dynamic control.

Keying The Bass

When the Bass is playing the same pattern as the kick drum and the 2 of them are not together rhythmically, you can key the bass so it only plays when the kick drum is playing. This cleans up the problem when the bass payer gets ahead of the drums. You do this by sending a trigger signal from the kick and sending it to the key input of a gate over the bass and using a medium attack time.

Producing Electric Guitar

Producing good guitar sounds can be a real challenge for a prod/eng in the studio. There are so many variables that factor into the sound and the process of getting a great sound might require a lot of work and experimenting.

First start with a good guitarist, with a good instrument and a good sound at the amplifier. Otherwise, there's nothing you can do in the control room to get a good sound. You can improve upon it, but you can't get a really great guitar sound if you don't have the basic ingredients. The first thing that you should do, whether it's with a guitar or another instrument, is go out and listen to the source (you might need ear plugs). You don't want to hear something for the first time after it has passed through a microphone, a mic cable, a fader and a pair of speakers. If there's a really killer sound coming out of the amplifier, try to capture that sound as you hear it.

With electric guitar you must first look at the guitar and its set up. The string gauge will greatly influence the final sound. Within gauge strings, for example less than 10mm high E string will present problems in getting a full sound. With thinner gauge strings, less sound will be produced. You will often find when using thin-gauged strings with a lot of amplification the noise vs. music factor will be enhanced somewhat sounding like you're listening to a 747. Once the string gauge starts to get thicker, the more tonality is produced. If your guitar player is used to playing thin gauged strings, get them to move up to mediums and practice with this new set up for a couple of days before going into the studio. This will allow the guitar player to adjust to the new speed of playing that will be required. Also check to make sure the action is set up properly on the guitar, which will produce an accurate balance of the strings.

With amplifiers, most guitar players tend to prefer tube amplification over transistor amplification. The tube amp will have a warmer and fuller sound where as the transistor amp will sound thin. Tube amps, also, when over driven, will bring out harmonic distortion that is more pleasing to the ear. Make sure that the guitar chord is the proper length. If it is too short it will require the player to perform too close to their amp causing feedback and inability to hear other performers. If the chord is too long it will affect the quality of the signal due to the increased impedance caused by the length of the chord. Often, guitar players like to overdub in the control room. In this case, have the guitar player get his sound with the amp on the floor and then move the amplifier into the control room and have a thick-gauged speaker cable running from the amp to the speakers in the studio. Beware of microphonic pickup when playing in the control room.

With amplifiers, the speaker size will influence the over all sound especially in the low end. If you discover that you are adding a lot of bass to a cabinet that contains 8”-10” speakers, try moving up to a 12” speaker cabinet to process the low end. If you start to get up to a 15” speaker, you will soon discover that the sound will be sluggish with no punch, due to the larger mass of the speaker and the damping factor of the amp. It’s just too difficult for that size of speaker to move efficiently. Remember to match the impedance from the amp to the speaker, especially if using multiple cabinets. If you decide to use more than one amp head by splitting the guitar signal, be aware of signal loss and degradation caused by impedance mis-matching.

Grounding and ground lifting your amp should also be options that will easily rectify buzz and hum. It also will prevent you from getting a large shock across the heart if you are double grounded through your amp and the studio ground. Most good studios have a separate ground for all their equipment independent of the circuitry that is used for lights, heating…etc. This also prevents double grounding that may cause RF.

When using effects in a guitarist’s performance such as DDL, chorus and other effects pedals, integrate the processing in the recording. Sometimes it is difficult to regenerate the same effect with a piece of outboard gear in the control room rather than the personal effects of the guitar player. Remember when using effects you need to ask yourself “Is the effect important to the creation of the sound or is it just being used as a detached sounding effect?”

When miking a guitar amp, there are basically two approaches. If you are working with a hard rock sound, it is best to go with one or more dynamic mics such as an SM57 and or a 421. This type of microphone will be able to handle the high SPLs coming from the amplifiers. If using more than one mic, make sure their distances from the speakers are the same to avoid phasing problems. These mics like to be placed almost on axis with the speaker to achieve that "in-your-face" sound. Bands like Linkin Park, Green Day and Tea Party utilize this approach today…. The other approach is to use a condenser with a pad. The condenser microphone has a larger pickup pattern and wider frequency response. Microphones like a U87 or 414 are good for this set up, especially if the melodic component is very important to the sound of the guitar. Try to keep the mic at least 6" from the speaker and experiment with the angle of the diaphragm to the speaker. This type of miking is used to capture the whole speaker sound and with the angle you can effectively control the mix between the music and presence in the mic pickup position. Bryan Adams, John Mayer and English rock bands use this approach. These guitar players will often play chords with three or more notes rather than the root and 5th of the heavy rockers.

With getting room sounds, I find that placing a mic in front of the cabinet at least 6' away will give you an additional sound that will contain more resonance to the sound, but you need to be aware of how the acoustics of the room are contributing to the overall sound. If you are trying to get a room sound for the guitar, this miking technique has a major drawback with the mic further away still picking up the original direct sound from the amp. Try placing a reflective baffle on an angle in between the cabinet and the distant mic. This will allow you to remove the direct sound from the room mic, thus allowing you to bring up the more dispersed reflections of the room. Try using 2 mics like this for a stereo ambient effect; you will be amazed how much dimension you will be able to create in the sound.

Taking a DI send from a guitar can be beneficial if you need to change the sound in the mix. I produced a band called Harem Scarem where the guitar player recorded his performances at home using a processing box that produced half decent overdriven guitar sounds. The guitar was also recorded as a DI to another track. When I was mixing I already had a half dozen guitar amplifiers and speaker arrangements set up for great and different guitar sounds. So when I mixed, I just took the DI performance, plugged it into one of the amplifiers on the studio floor and recorded the new amplified guitar sound. With this approach I was able to change the guitar sounds dramatically for mixing while always using a great performance.

In EQing guitars you have the low end (80-200hz) the music range (200-800hz), mid range and high end. With hard rock it is important to get a consistent low end and mid range. You will often get low-end volume excursions that will drive compressors too hard and upset the overall harmonic content of the guitar. When the guitar player goes from a G6 chord to an open E chord you might get a surge in level of the low frequencies. If you notice this happening you will need to insert a multi-band compressor whereby you will just compress the low end of the guitar and the remaining frequency response will remain unchanged. Try to avoid the high end where the vocals might reside in the frequency range. If you are not getting enough of the music out of the guitar sound, add in a little bit of the musical range. With pop rock guitar sounds it is important to hear the actual musical characteristics of the performance. It will also need to be present sounding in the mid range. As always remember to not over EQ the mid range and separate the midrange from the musical characteristics.

With comp/limit hard rock guitars will sound very compressed already but beware of quick transients sneaking through. If you get rid of these transients it will allow you to turn up the overall level of the guitar. With pop rock guitar you might need to use a bit of compression to keep it at a consistent level in the mix. With solo guitars limiting/compression will often be required.

If your guitar will be the main musical component in a song behind the lead vocal try recording it with perspective as stated above. This can be achieved by placing two room mics that can be panned hard left and right to add dimension and more resonant to the guitar sound. Also try adding in delay to the sound. Quarter and eight note regenerated delays will provide a lot of body and sustain to the guitar. Make sure your delay settings are a fundamental of the rhythm of the track so the attack of the delays will be masked by the rhythm, allowing you to use more of the effect. Also try using different time signatures of the delays to create different feels to the track. Edge of U2 uses a quarter or eight-note delay with a quarter note triplet or eight-note triplet delay at the same time, “Where The Streets Have No Name”. Ex; 120 Bpm; 250ms and 333ms. A dotted quarter note delay to a signal can make the signal feel like it’s pushing the track, giving it a shuffle feel. In mixing remember that with guitars you’re trying to slot them in the overall frequency range and perspective. Make sure that its frequency response does not interfere with the bass or the top end of a track.

Producing Acoustic Guitar

The acoustic guitar is very much in style today. Crossing between folk, pop and rock genres. While the acoustic guitar remains one of the most simple instruments, it also remains one of the hardest to get a great sound on in the studio. It's really not that difficult though, if you follow a few basic rules.

The sound you get has a great deal to do with the quality of the player. Choose an appropriate type and gauge of string for the instrument and for the kind of sound you're after and make sure that the guitar's action is set up correctly so that it plays without buzzing. There are many different types of steel-cored wound string, all of which have subtly different properties. The most commonly used types on acoustic guitars are bronze, phosphor bronze and nickel wound. An instrument with lighter gauge strings (perhaps an 11 to 50 set) will generally be easier to play, but the sound will be thinner and low in volume. On the other hand, very heavy strings (perhaps a set beginning with a 15-gauge top E) can sometimes sound tubby and lacking in overtones on the wound strings. The best compromise is usually the heaviest set of strings that are still comfortable enough for the guitarist to play. Usually starting with medium gauge strings will give you a decent sound.

The size of the acoustic guitar has a lot to do with the frequency range that it projects. The bigger the guitar, the more low end it'll provide. These guitars are most effective with strumming chords in the open position. These "jumbo" guitars are normally strum with medium to heavy gauge strings that are capable of producing more resonance due to the larger amount of wood that will resonate sympathetically. A medium size guitar will sound tighter and project the sound quickly, which makes it great for soloing.

There is also the nylon-string guitar or better known as the classical guitar where the top three strings are nylon. This type of guitar produces a mellow and a very harmonically even sound. It obviously does not contain the same amount of mid-range and high frequencies that steel-string guitars have. Nylon guitars are becoming more popular in pop music due to their capability to produce harmonic content in a frequency range that will not affect the lead vocal. A great example of this is in the music of Sting. In a song like Fragile the nylon guitar can be mixed tighter to the lead vocal for it is not encroaching in the presence frequency range of the lead vocal. If Sting were to use a steel-string instead, he would have to lower the overall level of the guitar because of the high frequency encroachment produced by the steel-string guitar in comparison to the lead vocal. That would lower the musical harmonic content of the guitar whereby it would separate the vocal melody from the harmonic accompaniment provided by the guitar.

The 12-string guitar is the grand piano of the guitar family. Usually played in a strumming fashion with a pick and chords in open positions. The 12-string guitar works most effectively by itself or with little accompaniment for it takes up a lot of the frequency and musical range. If you already have a basic 6-string performance and you feel you need a brighter guitar in addition try changing the 3 low strings with lighter gauge and tune them up an octave (Nashville tuning). Try to avoid capos', because they tend to choke the sound of the guitar. If the guitarist is using a pick, it is always worth trying one of a different thickness. With strumming you will tend to get a more even sound with a medium to light gauge pick. With soloing a thick or medium gauge pick works best for incorporating dynamics.

Another thing to bear in mind is that the sound of acoustic guitar recordings can depend a great deal on the environment in which the instrument is played. Acoustic guitars thrive on live acoustics, and insufficient natural reverb is a common problem when recording them in small home studios. While artificial reverb can be used to liven up the sound of a dead room, getting a sympathetic natural acoustic always produces better results, even if you want to add more artificial reverb later.
To get a more live sound out of your room, try to position the guitarist so that the instrument is played close to some reflective surfaces like hard floors, doors and solid furniture. If there is carpeting on the floor of your recording room try placing a sheet of plywood on the floor and get the guitar player the take off his/her shoes. Be prepared to have an additional pair of socks in case of gross air pollution.

Most studios will have a broad range of different mics to choose from, there are few dynamic mics capable of doing justice to the acoustic guitar. It is best to use a small-diaphragm condenser mic for its greater high-frequency accuracy, and one with an omni polar pattern for a more transparent sound and removing any proximity effect. If the room has bad acoustics you will need to use a cardioid to minimize the influencing characteristics of the room.

Capturing a natural sonic balance from the guitar is very important. There are different sounds coming from different places on the guitar that are important in contributing to an overall natural sound. If a mic is used too close to the guitar, the direct sound from that part of the guitar the mic is nearest to will dominate the sound from other parts of the instrument and from the room. You risk miking up only a part of the instrument when what you're really after is the bigger picture. Opposite if your mic is too far away from the guitar. You can end up with a lot of room ambience, leaving the original sound distant and unfocused. As for the specifics of mic placement, position your ear as if it were the microphone while somebody else is playing the guitar. Move your ear around to find the "sweet spot". A common approach is to set up the mic around 6-8" from the guitar, with the capsule aimed between the sound hole and where the neck joins the body. This will usually produce a well-integrated sound. The levels of direct and reflected sound will be about right and the sound hole's contribution will be controlled because the mic doesn't point directly at it. If you need more low-frequency content move the mic position closer to the sound hole. If you need a brighter sound move the mic closer to the 12^{th} fret. This is where the first series of harmonic overtones originate that contribute more high-frequency content to he overall guitar sound. If you have a pair of enclosed headphones that are very accurate to a reference point that you have established, you can easily experiment with tweaking this mic placement while listening for the best sound. If you find a promising sound in this way, remember to check it out on your monitors before committing yourself. Headphones can sometimes be rather misleading. If you find a good position but feel the sound is too dead try switching the pattern to omni and if the opposite occurs switch the omni pattern to cardioid. Be careful to not get too close for this will create an unnatural balance from the guitar. If you are working with a studio musician they will most likely have a custom-made guitar. Ask them where the "sweet-spot" is on their guitar for the performance they are playing. If the guitar player is soloing and moving up the neck try placing the mic closer to the sound hole to give you a fuller sound of the guitar. This will obviously compensate for the lack of low-end that the guitar can produce when used in a soloing fashion.

Selecting a microphone depends on the size of the guitar, if the player is playing open chords or soloing. If the player is strumming with open chords use a pencil condenser. If they are soloing, move to a large diaphragm condenser. Dynamic mics simply don't cut it.

A guitar with a built-in pick up and a microphone will undoubtedly create some phase problems. Experiment with moving the mic closer and further away from the guitar. That will affect the phase relationship of the two sound sources. Phase, he can be a tricky bugger. This will work effectively if the guitar player is also singing whereby minimizing the vocal leakage into the guitar microphone. If you are cutting a track in a studio with drums try using the direct pickup only and replacing it with an acoustic pickup in an overdubbing stage. Even though direct pickups on acoustic guitars have come a long way I have yet to discover one that sounds as good as a microphone pickup.

Stereo miking works well for solo applications. The XY technique is good but still falls short due to its lack of direct sound access. It will give you more of a big cardioid pickup but with less high-end than a single mic. Placing a mic over the 14^{th} fret and another just slightly off-center from the sound hole provides a good starting point for stereo pickup. Make sure both mics are pencil condensers, the same model and miked with the same distance from the guitar. Also incorporate a slight off-axis pickup.

The main challenge when using a stereo technique is to make sure that all the different signals are in time with each other when mixed; if there are delays between signals this could cause phasing problems. Some prod/eng's get around this problem by placing all the different mics at exactly the same distance from the guitar's sound hole and this can be successful.

As with any studio recording, the composition of the cue mix you feed to the guitarist will be extremely important, so be prepared to take a little time in preparing it given the sensitivity of the mics traditionally used in acoustic guitar recording, it's easy to pick up unwanted leakage from the headphones. Solo the recorded track to check for this and if there's a lot of leakage coming through (from a click track, in particular) then consider turning down the overall headphone mix level or using a different pair of headphones. Closed-back models are obviously best in this application and reduce the possibility of feedback.

Recorded acoustic guitar sounds will usually benefit from a little processing. This should be kept to a minimum while recording, so that you leave your options open for the mix. In recording roll-off any problems in the low-end such as rumble by inserting a high-pass filter. As always stated, it's always safer to leave EQ and dynamic processing until the mixing stage.

Equalization of the acoustic guitar is very common but used very subtly. The first thing to try is just rolling off some boominess bass, if there is some, using a high-pass or shelving equalizer at 60-80Hz. This can prevent the compressor from working too hard and maintaining an even harmonic balance. It can make a big difference, for example, if other sounds in the mix have strong low mid-range components and if you listen carefully to rock or pop mixes that include acoustic guitar, you'll notice that the low end is quite even. Most acoustic guitars performing in a strumming or fingerpicking style have a mid-range and/or high frequency boost. With the mid-range use a wide Q centered anywhere between 3-7K. If high-end is needed, try a shelving EQ from 8-12K which will produce a silky top-end sound. Be aware of making the acoustic guitar brighter than the lead vocal, if it is mixed at a loud level. If you need musical body, boost in the 600Hz-1.5K range with a medium Q. With acoustic solos you might need to enhance the low end between 100-200Hz to add more body to the performance especially if the guitarist is soloing high up on the neck.

With compression for strumming a ratio of 2:1 – 4:1 with a medium attack and medium release should be used, if required. Remember that the transient sound of using a pick identifies the rhythmic component of the performance. If the attack time was too quick it would create the illusion that the guitar player is playing behind the beat. For soloing you might need to limit the transients slightly, then EQ and then compress. With processing on an acoustic guitar it should be done with transparency in mind.

If the guitar performance is continuous strumming, there will most likely be no need for reverb. Reverb may be needed if the recording was made in a small room or studio. Mono recording can also be given a sense of space and width by adding a little stereo reverb. Ambience settings with pronounced early reflections are particularly effective in adding life and realism to the acoustic guitar. With strumming use a short pre-delay of 30-50ms and a bright reverb with a 1-2 sec decay time. With a guitar solo use a pre-delay of 100-150ms with a warm reverb with a decay time of 2-4 seconds. De-ess the send to the reverb if there is a lot of high frequency finger noise.

Producing Grand Piano

The grand piano is the most acoustically complex instrument to record, with its great dynamic range and wide musical range. From classical, jazz to pop music it lends itself very well to recording. There are numerous miking and processing techniques you may utilize depending on the desired effect you are looking for. Grand pianos vary in size from 7'-9'6" with the larger pianos sounding bigger due to the size of the resonating sound board. Achieving the precise tonal characteristics can be challenging yet will prove to be very satisfying when achieved.

One thing to acknowledge is that the same grand piano with the same miking set-up will most likely sound very different with another player even if they are playing the same musical piece. How a player strikes the keys and uses the sustain pedal are just some of the personal performing characteristics that define many different styles and sounds. With hard hammers and close miking you may get transients that meter too slowly to read and you'll have to use your ears to identify them. The mechanics of the piano can inhibit a good pick-up with the extraneous noises from the pedal, hammers and resonating buzzes. The acoustic ambient characteristics associated with the recording environment also influence the sound you are striving for.

With pop piano (Alicia Keys, Elton John), we tend to prefer a close pick-up. This allows for good clarity, minimal ambient influences. As we move into jazz-pop (Norah Jones, Dianna Krall) we discover that the grand piano sound starts to play a bigger role in a production and needs to be treated accordingly and isolation from the live singing is a factor. With jazz improvisation (Oscar Peterson, Keith Jarrett) the use of the piano harmonically and sonically are greater and miking set-ups are more challenging and need to be very accurate. Last but not least is classical piano where certain rules are applied for achieving an excellent pick-up that requires the ambient acoustics to play a major role in the overall sound.

Pop Piano

Pop piano is a situation where the piano plays a contributing role in the production by defining the chord changes. It needs to be able to be heard amongst various other instruments yet not overpowering or attracting too much focus especially if the artist is not known as a piano player. Close miking is the preferred way to go which allows for clarity and isolation.

I first ask the player where in the range on the piano they will be playing. A well disciplined player will play between 3-4 octaves, shying away from playing too many bass octaves with the left hand; respecting the bass player, avoid getting in the way of the lead vocals both harmonically and rhythmically and using the sustain pedal only when needed. Too much sustain pedal makes the piano sound too reverberant and muddy.

For mics I prefer pencil condensers for there ability to pick up an accurate mid-range and high-end. Because the diaphragms are small in mass, they tend to react faster than large
Diaphragm condensers and therefore pick up higher frequencies more efficiently. I don't really need a lot of low end from the piano for with a good arrangement a bass player will cover the fundamentals of the chord changes. If the player is playing a lot around or slightly below middle "C" and not to dynamically I'll use large diaphragm condensers to capture the low end.

I'll position the 2 mics approx 8"-12" above the strings and approx 12'-18" apart. Try to position them over the same harmonic points of the strings and slightly angle them between 35 - 45 ◦. If the mics are positioned on axis (0*), some of the notes will sound brighter than other notes due to the cardiod pick-up patterns of the mics. I'll also watch that the mics are not too far apart to avoid getting "the hole in the middle" sound. If the mics are too close to the strings the balance of the different notes both musically and sonically will be affected and I'll not get much of an even resonance from the soundboard. I'll try to leave the lid open but if isolation is required I'll slightly lower the mic position and lower the lid to the half way position and use some type of blanketing to prevent leakage.

With EQ, pop piano requires harmonic clarity. The music of the piano needs to come through clearly. I might add a little mid range 3khz-5khz and/or a little top end 10khz shelving (Wide "Q"). For some situations I'll just use a top end shelving curve and lower the activating frequency point anywhere down to 3khz. You need to remember that if you boost the mids and the highs you will get more clarity but eventually start to separate the brightness from the music of the piano. When we listen to a grand piano we tend to prefer to hear the left hand or low part in the left speaker and the right hand high part in the right speaker. When EQ is required it must be done to both tracks equally. As to not create individual sonic characteristics between the low end (left hand) and the high end (right hand). For example; if the piano needs to be brighter at 10khz, then boost the left and right channel the same amount. If you treat each channel differently in the sonic ranges the ear will tend to focus on the speaker with the brighter sound source. Applying the same EQ to both channels keeps the sonic characteristics the same and helps to retain good stereo imaging. If you need to EQ in the low-mids (music range) to create some separation, do it subtly if the EQ points are somewhat different. With low end add a little if you need to get warmth but be aware of clouding the production. Avoid bottom end boosting and if anything roll off the bottom end if the piano sounds boomy.

If the player hits the occasional chord too loud you might need to limit random transients to prevent distortion and/or the piano audibly jumping out in level. The best way to do this is to treat the transients in a sonic transparent fashion. That is to have some dynamic control without affecting the sound of the piano. The goal is to manage the transient problem without noticing its effect. With transients you need to limit only the upper part of the transient by using a very fast attack time; 1msec-10msec and fast release time; 10msec-20msec(link the channels together). The idea is to get in and out as quickly as possible, to manage the transient without affecting the intended dynamics.

If the piano does not define the chord changes enough and the attack sounds soft you can make it sound punchier and livelier by compression.

The idea here is to get the attack of the sound to come through more. You do this by using a med-slow attack time 50msec-200msec, and a slow release time 250msec or greater. This compresses the sustain part of the chord playing which gives you the sensation that the piano was played more aggressively.

I would suggest limiting first if necessary, then EQ (high-end) and then compress. A ratio of 3:1 to 6:1 should get you started. Bruce Hornsby uses this technique for getting his piano sound.

At certain times you might want to get a sustain sound from the piano. This is achieved by high comp/limit ratios, fast attack times and very slow release times. A good example of this can be found at the end of The Beatles song “A Day In The Life”. Here the release times on the compressor were modified to release very, very slowly.

Jazz Pop Piano

These days’ female jazz/pop singers and pianists dominate the charts. Norah Jones and Dianna Krall are 2 of the best. Here we acknowledge that the lead vocal is the most important element but is immediately followed by the grand piano. Everything else plays a supportive role. One thing in common is that these types of artists are highly skilled pianist and will use the entire piano both musically and dynamically. These types of artists sometimes record their lead vocals while they are playing piano, which makes capturing the best performance very challenging and trade offs are sometimes required. When performing place a piece of 4’ x 4’ foam, a foot thick right on top of where the sheet music tray sits. I actually remove the tray for it usually produces sympathetic noises. This gives me great isolation between the vocal mic and the piano mics. Because there playing at times maximizing the full range of the piano I’ll use large diaphragm condensers for their ability to capture low end. I’ll place them about 12”-16” above the strings so I can get more of the sound board resonance and 16”-24” apart for the range of playing will be wider. Because the higher strings on a piano do not have dampers they will sustain. If the top end mic is not positioned to capture this range it will sound distant and reverb like (basically a little higher and wider pick-up than pop piano). I will leave the lid fully open for I don’t want to choke the sound. If I need more low end from the piano I’ll introduce a third mic over the lower range in an effort to capture the sonic fullness of the piano. I will add this mic in to both left and right channels and use it more for sonic purposes than musical purposes. I will always rely on the full stereo imaging coming from the other 2 mics. If I have to roll off the mid to high frequency range of the 3rd mic to achieve accurate stereo imaging I’ll do it.

With EQ you will need to make sure the top end of the piano does not interfere with the presence of the lead vocal. If the piano is to bright you will invariably have to bring down its overall level. When you do this you also lower the music element of the piano. All of a sudden your vocalist sounds barren for they are musically out there exposed on their own and even though the piano can be heard clearly it will not contain enough of the harmonic information from the low mid-range to support the lead vocal.

Even though the levels of the piano and vocalist are close they are quite detached musically. "Be Aware"

With Compression/limiting and EQ I tend to use it minimally. As with all piano EQ and dynamic control what you do to one channel you do exactly to the other.

Jazz Piano

With traditional Jazz pianists like Oscar Peterson and Keith Jarrett you will get performances that are highly complex in dynamics and musical content. These types of
pianists are always improvising on the spot where they are literally are all over the vast range of the keyboard with incredible speed with dramatic dynamic changes. These random changes in performing are happening all the time and you need to prepare yourself to capture this type of performing. Experiment with mic positions and different condenser mics. Of all pianists, jazz players are the most articulate when it comes to meeting their needs. They "speak our technical language". I usually use pencil condensers like the B&K (DAP) 4000 series. They can handle a lot of level without distorting and can translate the percussive nature of jazz playing. As I said previously small diaphragm condensers capture very fast transients more accurately than large diaphragm. I'll often place the 2 mics slightly higher in the range and factor in a 3rd low end mic assigned to both left and right. If the room has good ambient characteristics I'll pull the mics further back and get an overall sound from the piano. With Jazz it's nice to get the articulation and the resonance of the piano. For dynamic control I'll get the player to play very loud and I'll still back off the level to allow for more headroom. With traditional jazz playing you do not want to be in a position to have to reset levels to prevent distortion. If any EQ is required it's usually in the high end just too add a little shimmer to the sound. With dynamic control, it will be used rarely for the dynamics are often exaggerated to highlight the performance. If any limiting is required it will be for getting more level on a CD, but only if it is not that noticeable.

Classical Piano

In recording classical piano factoring in a good recording ambience is very important to the overall sound. Recordings by the greatest classical pianist were mostly done in good concert halls and large studios.

The conventional and traditional way to record piano is to set up 2-3 large diaphragm condensers at different angles approx 8ft-12ft away facing the piano. The mics are angled similar to the angle of the piano lid opening and are usually set up high (6ft-12ft). Large diaphragm condensers are used to capture the low end of the piano and are often used in an omni pattern to allow the acoustics of the space to be used and mics used in an omni have a flatter frequency response than mics in cardioid patterns.

The distance between the mics and the piano dictates the ratio between the direct sound and the ambient sound. The goal of this type of recording is to place the piano and the mics in a strategic place for optimum clarity and room ambience. However I find this type of pick up limiting.

If the tempo changes dramatically from adagio (slow) to allegro (fast) the piano sound can vary. If you found a mic position that suited an overall good pick up, you might discover that the piano sounds detached and dry with the slower moving pieces and quite muddy at faster tempos.

What is perceived to be happening is that this type of pick up has dead spots in it, where once the ambience completely decays you hear dead air between the notes. The opposite happens when the tempo picks up and is quick. The piano begins to sound muddy and reverberant, for the decay is hanging over too much into the next note. This can be very apparent in there are sudden dynamic changes where the piano goes from a loud dynamic and quick tempo to a softer dynamic and slower tempo in a short period of time. The piano sound appears to have too much reverb in amplitude and decay time. Other than taking the time to find suitable mic positions and piano placement and risking losing a good basic mic position and piano placement most often the people involved will settle for a basic good all round position. I find this limiting.

I was very fortunate to work with Glenn Gould the greatest classical pianist of the 20th century who was very much into sound innovation. With his recordings I would find a good position for the placement of the piano, usually in the center of the room away from any close walls. Next I would place 3 large diaphragm condensers approx 8ft-12ft from the piano in cardioid pick up patterns. With this placement I would strive for a sound that would be clear and balanced if the music was at a quick tempo (Allegro). Next I would place a stereo mic or 2 matching condensers between 12ft-18ft from the piano in omni to capture a medium reverb time that would include early reflections. Next, I would set up another stereo mic or 2 matched condensers approx 20ft-30ft from the piano for a reverb effect. This would allow for smooth decay times if the tempo was slow (adagio) and avoid any dead air.

With this type of pick up I have maximum control over the recording situation. With 3 different mic setups at various distances I can change the piano sound from a clear distinct sound to a very warm and reverberant sound without changing the mic positions and having to constantly go back into the studio to alter mic and piano positions. With Glenn I would preview and mark the score where changes would be required if I was recording to a 2 track final mix or record to a multi-track and have control when I was mixing.

Microphones

B&K 4000 series, Neumann M149, 87, U-67, U-47, Akg stereo C-24
Excellent transient response, quiet, flat frequency response

Pre-amps

GML, Millenia, Neve, --any high quality pre that is quiet and good transient response

EQ

Neve, GML, API, Manly; --4 band, quiet, no colouration effect; more edge around 3khz-5khz bell curve wide “Q”; presence 10khz and up shelving; low end fullness 80hz-150hz

Limiting

Very little, not perceivable when inserted, fast attack and fast release times

Compression
Usually for pop; level control, creating more attack to the sound
Med attack-medium release

Producing Lead Vocals

There are many types of singing and various methods of recording vocals from classical, crooning, rock etc. You will discover that you need to develop personal styles and techniques of capturing and enhancing their performance.

As a producer or engineer you need to know what is required of your talents to effectively fulfill your role in capturing a good take and sound.

Listening to various successful recordings will provide you with not only a reference point but also with a framework to further enhance your goals and objectives in capturing a solid quality performance. Audition some CDs of vocalist's sounds that can be related to what you require. On a reference monitoring system, this should give you a starting point in where to go with equalization, processing, and balance in a mix.

You will require a good quality **microphone** (condenser), a **preamp** (that can amplify a very dynamic performance and maintain a quiet noise floor), a versatile **equalizer** and a transparent **compressor/limiter**.

Because recording vocals is often a sensitive and emotional issue for singers, it is a good idea to consistently give the singer positive feedback of words of accomplishment and encouragement.

Microphones

For most vocalists a high quality large diaphragm condenser microphone is often the choice. If you have access to a tube mic, even better. The tube mic will sound warmer and if there is any distortion, it will be less offensive to the ear. U-47, AT3035, C-12, M-49, C-4000B are often found in better studios and work remarkably well. U-87, AKG 414 and Rhode mics will be found in about every studio and often work quite well. The U-87 will have an even frequency response, where the 414 will accentuate the high end. If recording a rock vocalist try a Shure SM-57. You will get an enhanced mid-range sound with no distortion.Ribbon mics like the RCA models are very good but most have a high noise floor.

When recording bed tracks, change the mic from song to song to get a general idea of which mic sounds the best. Also when ready to record final vocals, line up 3-4 mics and quickly have the vocalist go from one to the other to see which mic is the most desirable. Remember to check all the dynamic parts of the song; certain mics sound good in verses but might be to thin sounding in the choruses.

<u>Microphone placement</u>

For a lead vocal place the mic around 3”-6” from the singer. A pop filter may be required. In choosing a pop filter, make sure it stops a lot of wind transmission (blow at the filter and place your hand on the other side to check) and does not affect the frequency response too much. (Place the pop filter between your ear and a speaker and move it in and out of the way and listen for any sound degradation). Place the pop filter as close to microphone as possible for vocalist’s do not like singing close to a pop filter. If the singer is too bassy from the proximity effect, either change the pattern from cardiod to omni, insert a high pass filter or simply have the singer stand a couple of inches further back from the mic. Take note that when a vocalist is moving back and forth from the mic in an area from 1”-3”, the low end will drastically change and become very hard to control. Make sure the microphone is suspended in a cradle to remove or to prevent unwanted rumble coming through the mic stand. Make sure the acoustics of the room do not influence the desired vocal sound, which occurs when the vocalist stands too far back from the microphone. If the room is too live try to have the singer move in closer to the mic or dampen the room with blankets or baffles usually close to the singer. If there is a music stand involved for the singer to read lyrics make sure it is dampened down and the stand doesn’t ring sympathetically with the vocal performance.

<u>Creating The Right Environment</u>

Before recording vocals, ask the singer what they need to feel comfortable in the studio when recording. Remember singing is an emotional and mental experience, so having the singer feeling relaxed is very critical. Try to set up baffles covered in quilts and blankets close to the vocalist, this makes the studio seem more comfortable and helps reduce the room acoustics in the sound of the singer. Keep the lighting tapered with a lamp or candles. You might need a small lamp to place on the music stand so the lyrics will be seen easily. Have a comfortable chair and table to place things on and a pitcher of water and a glass for vocalist's throats dry up quickly. Make sure there are pencils on the music stand for singers have a habit of changing lyrics at the last minute. Also place them in an area of the studio that they will be in a position to not have to look at the control room all the time. Standing in the middle of a big studio with bright lighting and people staring at you can be very intimidating for a vocalist, so creating a very comfortable and relaxed environment is very important.

Equalization

Male Vocalist*:*
High pass filter at 50hz
Low end 100hz-200hz
Low mids 400hz-800hz; med "Q"
Mid range 3khz-5khz
Top end   10khz and up

Female Vocalist:
High Pass filter at 80hz
Low end 200hz-300hz
Lo mids   400hz-800hz
Mid range 3khz-5khz
Top end   10khz and up

Limiting and Compression

A good vocalist will work with mic distance in relationship to dynamics. During soft and loud passages they will intuitively move back and forth from the mic. This will lower the effect of the dynamic control function and maintain a high quality sound. However, when starting out as an engineer or producer you will most likely not have this luxury or feel intimidated to solicit advice to the vocalist. Even with a good microphone and good mic preamp, recording vocalists can be a major problem if various processing is inserted in the wrong sequence. For example: if you insert a compressor or limiter with too slow of an attack time what ends up happening is the dynamics of the vocal performance expands. This is caused by too slow of an attack time on your comp/limiter whereby the initial transient passes through the comp/limiter unaffected and the remaining vocal dynamic is affected. If inserting EQ that enhances the mid range or high end, before this type of setting on the comp limiter it will exacerbate the problem even further. You could also introduce sibilance problems into the sound. To play it safe I would suggest this technique:

First limit the vocal with a quick attack and quick release time - this will allow you to manage the transients of the vocal. This will make the vocal more suitable for compression, if desired. Do not EQ the vocal before limiting. Insert the EQ directly after limiting but before compressing. With compression the limited vocal will allow you to use a medium to slow attack time and medium to slow release time. This affectively compresses the tonality or vowel sounds of the vocal, which often require level management. A ratio of 2:1 to 4:1 should suffice. An attack time between 25-100 ms and a release time of 200-500 ms, or better yet use your ears to get the right attack and release settings. Make sure the release time is slow enough to prevent pumping and breathing yet fast enough to not affect the next part of the signal that might not need to be compressed. When dynamically processing a vocal try to have the vocal go back to unity gain as often as you can for example: with a 4:1 ratio the meter should be moving from 0VU to -4VU. If you see the meter moving from -8VU to -4VU you are over-compressing and corrupting the quality of the vocal. Remember, the more you dynamically process a signal, the thinner the sound will get. Dynamic processing does not process evenly over the frequency range, especially in low-priced compressor/limiters. EQ before compresssing. For example if the vocal has too much low-end and is not EQed the compression will be triggered by the low end which will only thin out the sound leaving dynamic problems untouched. Also, if you are EQing mid-range into the vocal the compression will factor in the EQ and compress effectively.

Sibilance

Sibilance is a problem that can destroy the fidelity of a production – a singer who's every S and T is accompanied by a burst of high-frequency noise. This isn't anybody's fault, it's all down to how an individual's mouth works, but it seems that the better the microphone the more sibilance is captured. This is especially true of some condenser mics, but unfortunately some people tend to equate a very bright vocal sound as being more refined or better produced. What's more, adding effects such as reverb or using heavy compression can make sibilance noticeably worse.

As sibilance is a high-frequency problem, and equalizers are designed to emphasize high-frequency detail, it's hardly surprising that using an enhancer tends to exaggerate sibilance even more. The best place to tackle this problem is back at the source, and if you have a mic that's less susceptible to the offending frequencies, try this first. Don't worry if it's not as bright as the original mic; you can use equalization to help compensate for that. Be aware of high frequency distortion that might sound similar to sibilance. The high frequency distortion will most likely be coming from the mic or mic pre-amp. Changing the position of the singer relative to the mic may help in decreasing the sibilance, but in serious cases, you may need to resort to using a de-esser.

If you need to de-ess, do it before you EQ and comp/limit. Any high end equalization before the de-esser will make it work harder. Also focus on the problem frequency range of the sibilance. If you notice a “shzzz” sound, the problem area will be in the 3kHz-7kHz range. If it sounds “ssss” it is in the 8kHz-12kHz range. Most de-essers have a mode where you can listen to what is being removed from the signal called a side-chain monitor. This will effectively let you target the problem frequencies accurately and also indicate how much of the sibilance you are removing. When de-essing try to avoid looking at the reduction meter and use your ears. De-ess as much as necessary without creating a lisp problem.

De-essing is “frequency select limiting”. It uses very fast attack and release times due to the short waves lengths common to sibilance. Most DAWs have plug-ins that will de-ess but actual analog multi-band compressors work best. You can vary the Q, the ratio, the attack and release times and the amount of gain reduction. The Brook Sirens unit is one of the best de-essers out there.

De-essing the reverb send from vocals will greatly reduce the level and duration in the reverb. Remember that sibilance is just noise; there is no musical component to it. In most natural reverb settings you will rarely hear a sibilance problem in the decay of a sound. By de-essing the sibilance the reverb will still produce high frequency reverb content that might be desired when mixing especially if there is a lot of EQ in the 12-15K range used for creating a breathy intimate effect.

If you are adding mid-range and high-frequency to a vocal always de-ess before you EQ. This will prevent the compressor from creating more of a sibilance problem, keeping in mind that high frequencies contribute a small amount to the overall lead vocal level.

For example: in a word like SPARK the S content will meter -20VU and the PARK will meter 0VU. If I compress the signal without de-essing before hand the S will remain at -20VU and the PARK will drop to -6VU. What you have done is taken the original 20db difference between the S and the PARK and now made it 14db effectively creating more of a sibilance problem. If you were to EQ the high frequency range this will exacerbate the problem even further. The trick is to get the PARK sounding as compressed and EQed as you like and then with the de-esser inserted before the compressor and the EQ, take away the amount of sibilance you want.

<u>Headphone Mix</u>

It is very important that you take the time to provide an excellent headphone monitor mix to the vocalist for singing. Most vocalists will need to hear a clear band mix with sufficient harmonic and rhythm content. If the vocalist is getting ahead or behind the beat you will need to send more drums or instruments with a rhythmic component. If the song when finished will have only a lead vocal and a solo instrument for the intro and first verse, you might suggest to the drummer to keep time by playing the hi-hat softly so it can be used to keep everybody in time and then can be removed for the final mix. Note that most singers do sing ahead of the beat.

If the vocalist's pitch is a problem then you might need to send more harmonic instrumentation to the headphone mix. If there is not enough there you might put down a synth pad guide track for the vocalist may reference their pitch too and then not use it in the final mix. If the vocalist has to come in before the downbeat insert a pitch reference a couple of seconds before the song starts. This works especially well if there are key changes in the song and you always have to back to the beginning. This is also a good time to experiment with reverb settings; compression, EQ and effects for singers love to hear an enhanced sound in their headphones. If you find the singer projecting too much or singing too softly then they are not hearing themselves properly in the headphones and this will cause numerous technical and performance problems. Try to set up to record at least 4 tracks so you can have 4 takes to choose from to make a master take.

Producing Backup Singers

Backup vocals are mainly used to provide harmonies to the lead vocalist. To produce them properly you need to relate to them, how much they need to express in volume and timbre for them to blend well with the lead vocalist. There are many ways to record backup singers. One is for them to split parts amongst themselves on the studio floor and double track their performance. Often they will perform another harmonic blend to contribute harmonically to the lead vocal which usually occurs in the choruses. By doubling or tripling the backup vocals it will allow you to mix them in at a level where the musical component will stand out without them having to sound too present. Good singers will sing without vibrato and will either close off their S's or not sing them while performing, which keeps the performance sounding clean. Often, lead singers make poor backup vocalists due to the fact that they can't control their dynamics or sing without vibrato. It is important to give them a headphone mix that focuses on the lead vocalist, harmonic content and rhythm. Just supplying them with four or five elements that are indicative of accurate pitch and rhythm will be enough to enable them to sing well. If the lead singer is also doing the backup vocals you will obviously need to use a lot of tracks (Queen), the singer will have to focus on matching the phrasing on their previous vocal tracks. When recording, it is advisable to use a good quality condenser microphone in a cardioid pickup. If you have a stereo condenser try it, for it will allow you to achieve a stereo perspective and widen the pickup pattern. Try to avoid giving one mic to one singer if using more than two backup singers. This will deteriorate the effect of proper blending. Always give the singers at least 2 bars before they start singing for pitch and time reference. Make sure to clean up all the extraneous vocal sounds when you are finished recording. If recording to more than one track set up all additional tracks with the same processing and levels so you do not have to continually set up tracks on the go, whereby you're slowing down the recording process. Try to always set up more tracks than you need. Bus assign to all tracks and make sure all signal routing is clean. If the band recorded to an automated rhythm you will be allowed to use one chorus for all the remaining of the choruses (Lose My Breath). I myself like to have songs build in dynamics and prefer to have the backup vocal performances a little more expressive towards the end of the song. With compression, use a ratio of 3:1 - 4:1 for just tightening up the dynamics. If you’re using 3 or more tracks, try bussing to one stereo set of tracks. This will allow you to bus compress, EQ and process all the backup performances uniformly. With EQ make sure that the backup vocal performance does not encroach in on the lead vocal performance. There are exceptions to this rule due to the nature of the production. In panning backup vocals try to place them on equal sides of the centre image (9 o’clock and 3 o’clock). Panning them hard left and hard right tends to draw too much focus to their performance. In mixing try grouping all your tracks to one channel input for one master send for all performances. Remember this channel is to serve just as a master send so all backup vocal tracks get the same amount of processing. Remember to take the master send track’s main signal out of the 2 mix. When mixing in your backup vocal tracks to the final mix keep in mind their priority of their importance to the song.

Equalization Key Frequencies

30Hz - 60Hz

1. Increase to add more fullness to lowest frequency instruments like foot and the bass. Will give you a feel bottom end for monitoring systems with 12” or larger woofers.

2. Reduce to decrease the "boom" of the bass and will increase overtones and the recognition of bass line in the mix. This is most often used on loud bass lines like rock.

3. 60Hz – 30Hz Hpf to remove extraneous noise or undesirable low frequency from audio signals.

60Hz - 200Hz

1. Increase to add a fuller bass sound to lowest frequency instruments.

2. Increase to add warmth to guitars, toms, bass, snare and piano.

3. Reduce to remove boom instruments & increase clarity.

200Hz - 800Hz

1. Increase to add musical content instruments and vocals.

2. Varying bandwidth or “Q” to focus in on the sweet spot on instruments.

3. Reduce to decrease the unwanted overtones from drums.

800Hz -1.5KHz

1. Increase for clarity and "punch" of bass.

2. Hi-lights the first set of overtones from instruments.

3. Reduce 1.5KHz to remove harshness from hi-hat.

4. Reduce 1.2KHz – 2KHz removes brittle sound from loud singing vocals.

1.5KHz - 3KHz

1. Increase for more "pluck" of bass.

2. Increase for more attack of electric / acoustic guitar.

3. Increase for more attack on low piano parts.

4. Increase for more clarity / hardness on voice.

5. Reduce to increase breathy, soft sound on background vocals.

6. Reduce to disguise out-of-tune vocals / guitars.

3KHz - 5KHz

1. Increase for vocal presence.

2. Increase drum attack (snare / toms).

3. Increase for more "finger sound" on bass.

4. Increase attack of piano, acoustic guitar and brightness on guitars (especially rock guitars).

5. Reduce to decrease the “shhh” sound from singers.

5 - 10KHz

1. Increase to add presence to harmony vocals and secondary focus instruments.

2. Increase to add attack to percussion instruments.

3. Increase on dull singer.

4. Increase for more "finger sound" on acoustic bass.

5. Reduce to decrease "S" sound on singers.

6. Increase to add sharpness to synthesizers, rock guitars, acoustic guitar and piano.

10KHz - 12KHz

1. Increase to brighten vocals.

2. Increase for "light brightness" in acoustic guitar and piano.

3. Increase for hardness on cymbals.

4. Reduce to decrease "S" sound on singers.

12KHz - 16KHz (shelving)

1. Increase to brighten vocals (breath sound).

2. Increase to brighten cymbals, string instruments and flutes.

3. Increase to make sampled synthesizer sound more real.

4. Increase to add silkiness to acoustic instruments

Compression and Limiting

What is Compressing and Limiting?

The main job of a comp/limit is to reduce the dynamic range of a signal. The comp/limit reduces the loud part of the signal it does not bring up the lower parts. That function is done by the prod/eng. Under what circumstances is this a good idea, and what, if any, are the trade-offs? One of the side effects of compression is that the sound being processed changes: the more you comp/limit the more you alter the fidelity of the signal. Comp/limit is used for two purposes as in all processing 1) to manage the signal and remove any sonic problems 2) to be creative. In managing the signal comp/limit should sound transparent. That means it is doing its job without any noticeable degradation to the signal. In creative situations you will utilize the comp/limit to alter signals to achieve desired effects.

Most vocalists (and some musicians) need comp/limit to keep their levels even; bringing down the peaks means that you can make the average signal level higher (Manually bringing up the output) and more audible in the mix and this results in a tighter, more confident sound that sits well in a mix. However, don't just rush in and apply maximum comp/limit. Listen to the singer run through the song, watch the record level meters, and try to decide for yourself whether the voice needs a lot of control or simply a light touch. In any event, it's better to under-comp/limit or just use subtle limiting during recording, to prevent digital distortion from occurring in digitally storing the signal. Remember that digital distortion is very unpleasant as compared to analog distortion. Some of the better A-D converters incorporate hard limiting to prevent distortions. Other cheaper converters will just simply shut you down and not process at all. 24-bit digital storage is very quiet when used properly allowing you to always apply more comp/limit when you come to the mix stage, whereas over-comp/limit is impossible to reverse when recorded that way. With today's DAW's (24 bit) there is virtually no need to dynamically process a signal for level control that can be done in a mixing stage without introducing noise problems. It is best to use comp/limit in the monitor stage when recording and then accurately processing the dynamics in the final mixing stage.

Limiting

Limiting is designated as a high compression ratio, any ration from 8:1 – infinity. Limiting incorporates very fast attack and release times. It is used to manage fast acting transients in a signal. Broadcast transmission always incorporates limiting to prevent their transmitters from blowing up. However, they commonly use very slow release times to avoid any pumping and breathing effects. Any audio signal that contains transient's work well when limited rather than compressed such as drums, lead vocals and guitars.

Compression

Compression uses ratios from 1.5:1 – 7:1. The goal is to tighten up the dynamic range of a signal so it can be heard easily throughout a mix such as a lead vocal. Compression usually incorporates medium to slow attack and release times.

Input

The input level of the signal to the comp/limit should be set to around 0VU with occasional excursions going up to +3VU. Too much gain in the input stage will introduce distortion. Not enough gain will cause increases in the amplification stage of the comp/limit increasing additional unnecessary noise to the signal. When setting up a comp/limit, use an oscillator from the board set at 0VU and set the input and output of the compressor to 0VU. This is known as Unity Gain.

Setting the Ratio

If you have a ratio of 4:1 that indicates that for every 4db over the threshold only 1db will be added to the output of the signal below the threshold. For example: if you have a signal that is peaking at 88db and the threshold is set at 80db the final output will now be 82db.

Threshold

It determines the upper area of the dynamic range that needs to be comp/limit. The lower the threshold the more of the signal will be comp/limit. The Threshold is preset manually by the prod/eng. Make sure that the comp/limit continually returns to 0db of gain reduction. If you notice that the gain reduction meter is continually going back and forth between -8VU and -4VU that means the signal is being comp/limit too much thus degrading the quality of the signal. Reset the threshold so that the meter reads between -4VU and 0VU to minimize any degradation to the signal. Remember that the more you comp/limit a signal, the more the quality of the signal is affected, especially dealing with harmonic content. If you need to really comp/limit a signal try to use a tube comp/limit, which will produce distortion that is more palatable to the ear.

Attack Time

This is the time it takes for the comp/limit to start processing once the threshold has been crossed on the increasing amplitude side of the wavelength of the signal. Attack times vary from less than 1ms to almost a second. Some comp/limit have Auto-Attack (VCA).

Release Time

This is the time it takes for the signal to return to its original level after the signal has crossed the threshold of the downward part of the envelope. The release time's length is determined by the prod/eng. Some comp/limits have an Auto-Release function (VCA) which does the job for you and works well for lead vocalists. To prevent low frequency distortion (half cycle-length) a long release time is required. Short release times are often desired but may produce pumping and breathing effects.

Hard Knee/Soft Knee Compression

If the comp/limit ratio is set to 6:1 with a soft knee, once the threshold has been crossed the comp/limit, over time (ms) will go from 1:1 and rise to 6:1 compression. This results in a smooth transition from the on-set of compression to full pre-set compression. With a hard knee this transition is immediate. Soft-knee compression usually provides the smoothest compression, but hard-knee models give tighter control and may be better for creating hard compression effects.

Noise

Comp/limit are not inherently noisy devices, but because they reduce the dynamic range of the signal being treated, some make-up gain is needed to bring the signal level back to Unity Gain. Although comp/limit really just turn down loud sounds, once you've adjusted the make-up gain control, the loud sounds are back where they were and the quieter sounds are much louder. If you're compressing to achieve 10dB of gain reduction, any noise that happens to be part of the signal will also be increased by 10dB for all input levels that are below the threshold. Periods of silence between words or phrases are most vulnerable, as it's here that the comp/limit gain is highest. As well as starting off with the cleanest signal you can, it might be wise to gate the signal immediately before it enters the comp/limit. Some comp/limit have built-in expander gates for this purpose: used properly, they really can make a difference. If there is a lot of output noise like that is found in some tube comp/limit, you might need to insert an expander to minimize the noise.

Multi-Band Compression

When a certain frequency range becomes a problem, the user can help to target the problem area and compress the undesirable frequencies. If an acoustic guitar has too much low frequency build up you can compress only the low frequencies and leave the upper frequencies alone where the guitar remains harmonically intact. This also works well when you need to remove sibilance from vocals.

Getting a Loud Sounding Mix

This process is still very misunderstood with today's productions. I've discovered that in the rock genre there were a lot of productions that sound awful. When it came to finalizing or mastering their mix they all seemed to have employed a lot of hard wall limiting and a lot of equalization in the mid-range and high frequency range.

If you desire this effect and your mix already sounds balanced in level throughout, first limit with a super fast attack (less than 1ms) and fast release time (10 – 20ms). With most mixes today the number of transients and their duration contribute little to the overall length of the song. If you noticed that there are 50 (approx) transients in your production, the transient duration will be 2ms (approx) in duration. If you wanted to get this production sounding 4x louder you would need to limit the top 6db of the production. For example: if the maximum head room is 90db and these 50 transients are reaching this level you can limit all these transients down to a level of 84db and then bring up the entire production back to 90db, thus increasing the overall level by 6db. The amount of signal that has been affected is only 1/10th of a second. What you have effectively done is limited 1/10th of one second of a production that is 5 minutes in length. In other words you have only limited 1/3000th of the time of the 5-minute production. Be aware of half-cycle distortion due to short release times. The best way to target the optimum release time is to solo the bass and bass drum in the mix and listen for any distortion or a grainy sounding effect. If this occurs lengthen the release time until it disappears. It is easier to solve this problem by soloing the low frequency content of the mix rather than listening to the entire mix.

Next, add in the EQ needed. This will prevent the limiter from working harder than necessary. The transients have now been flattened out slightly so if any high end is required it will not be that much because the duration of limited transients are now longer and more noticeable to the ear. If you put on more high end before you limit you would need to add more to hear the effect on the transient percussive parts of the production, whereby making the limiter work harder.

After EQ you can now compress to create an overall loudness effect to the mix. Use ratios of 3:1 – 6:1 with medium attack (40-80ms) and release times (100-250ms).

If your mix has major level discrepancies between the verses and choruses, try this set-up compress-EQ-Limit

Getting A Punchy Mix

When you have finished the mix and feel it is soft and not rhythmically defined as you like you can create a little punch and make it kick a bit more. First limit just a little to get the odd transient out. That should not be hard, for you have already declared that the mix needs to sound more transient throughout. Once you have done this, clean up the extraneous low bottom end and focus on getting low end to speak around 100hz. Add in a little mid-range from 2-4khz and the compress with the following settings. A ratio of 4-6:1, with an attack time from 50-100ms and a release time of 150-300ms.

Getting a Radio/TV friendly mix

The way broadcast works is that they insert settings of infinity : 1 as the limiting ratio, super fast attack times (micro-sec) and very long release times (3-6 sec). This means that if you have a high transient in your mix, the station’s limiter will grab it and keep your mix down in level for at least a couple of seconds, even though the transient lasted for only a very short time. To get around this, incorporate your own limiting that resembles the broadcasters, except shorten the release time to as fast as possible. This will allow your mix to get back to its normal level almost immediately. The goal here is to get in and out as fast as possible. Be aware of half-cycle distortion!

Linking 2 Comp/limit

If a situation calls for comp/limit on a stereo source like a grand piano, overheads, room mics… you will need to link your 2 mono comp/limit's together so that their attack and release functions work in sync. This prevents random imaging problems. If a piano player hits a low note with their left hand at a very loud volume, the stereo comp/limit will bring down the output of the other comp/limit (right hand) the same amount and at the same time, keeping the stereo image intact. When setting up make sure that one of the comp/limit is set as the master by using the link button.

Expanders and Gates

Basically an ex/gate is the opposite of a compressor, where a compressor reduces the dynamic range, an ex/gate increases it. It is used mainly for getting rid of undesired noise, leakage problems and creating special effects. For example; hi-hat leakage on a snare track, gated reverb, keying additional sounds… An ex/gate appears to make the louder parts of the signal louder but doesn't, it actually lowers the level of the softer parts of the signal making the louder parts more audible.

How it works

When a signal falls below a preset threshold, the signal is eliminated or lowered in amplitude. When the signal moves above the threshold the gate then opens and doesn't affect the signal anymore.

Attack Time

Once a threshold has been set, the signal must pass that threshold for the gate to open. How fast it will open is determined by the attack time. In order for drums to work you will require a very fast attack time in order to allow the transient to fully pass without any deterioration of the initial part of the waveform. With instruments that do not contain initial transients, a medium attack time will be fine. However, signals that contain a lot of low frequency information will require a med to slow attack time because of half cycle distortion. For background ambient noise problems, a slow attack will be needed so it doesn't sound like an on/off switch.

Release Time

Once the threshold has been crossed on the decreasing amplitude side, the release time determines how the long the ex/gate will take to close again. Extending the release time may be used for creating special effects. For example gated reverb.

The Range

The range is similar to the ratio control on a comp/limiter. It determines the level of the gated signal to the original sound. For instance, if you are trying to eliminate hi-hat leakage from a snare track and still wanted a certain level of hi-hat leakage, the range control will allow you to control the level of hi-hat leakage you still wanted in with your snare track. This processing is called "expanding the dynamic range". If you decided to completely eliminate the hi-hat leakage from the snare track, this processing is called "gating".

When trying to remove unwanted background noise such as amplifier buzz, room and ambient noise it is critical that you not gate the entire signal. This will draw less focus to the gate opening and closing. The ear will always notice audio that is switched on and off immediately more than audio faded in and out quickly, for example, a guitar amplifier with a noise problem. If you set the gate to fully closed and the guitar then triggers it to open the ear will notice two things: 1) the sound of the guitar and 2) the sound of the noise turning on. This confuses the ear for it does not know which signal to focus on. If you set the gate to allow a low amount of noise to come through all the time the ear will not notice the noise shift in level when the gate opens. For it is now perceived as a level change rather than an on/off change. This might also require you to extend the release time slightly for a smoother fade into the gating mode. Try to keep all gating and expanding in the monitor section when recording.

Key Input

This is a function that allows you to ex/gate a signal by using an external source as the trigger to open the gate when desired. When using room mics for drums and you want to favor the snare drum in the room mics, you will need to set up gates on each mic in the expander mode. To accomplish this take an auxiliary send from the snare mic, gate and compress the send to provide a clean and even balanced trigger. Take this send and insert it on the key inputs of the room mic channels. In the gates set the range to maximum whereby all audio information below the threshold is gated. Then set the attack time to as fast as possible and the release time to medium. This will enable you to find the optimum threshold point. Then set the release time to however long you want the gate to remain open. Then set the range to arrive at the desired level between gated and non-gated audio information. For example if it is set to 50% you will have the snare drum content at 100% in the room mics and the rest of the kit will be 50% of the level.

Applications

It is obvious that an ex/gate can solve leakage problems but it is also a great tool for creating certain desirable effects. Here are a couple of examples:

1) If you have a bass player and a drummer playing the same rhythmic pattern you may use the kick to trigger the bass to open only when the kick is heard (downbeat). This allows the low-end part of the performances to have accurate rhythm. It suggests to the listener that the bass player and drummer are listening very closely to each other.

2) You could have the snare drum trigger different musical sounds to create a harmonic effect on the snare rhythm pattern.

3) Using the gate function to alter a reverb decay sound to create what is better known as “gated reverb”. This is accomplished by a predetermined threshold and release time.

4) Triggering drum ambient mics to only open on a snare to create a bigger snare sound. This is accomplished by using the original snare track as a gated send to the ambient mic’s trigger input.

Digital Delay

When a sound source excites in a reflective acoustic environment, it will produce duplicates (delays) of the original signal. These duplicates will never be the exact in frequency response, and dynamic structure will always differ from the original. The overall nature of the sound will vary because of; the time difference between the original sound and delayed sound, the dynamic nature of the delay, the number of audible delays over time and the frequency response of the delays.

If the sound source is a vocal and the first delay is heard within 15ms of the original, it will produce an overall flanging and phasing effect; for the delay is so close in time to the original that it corrupts the frequency response of the original. If the delayed sound is between 15ms and 100ms, the overall combined effect will produce a distant sound. If the delayed signal is over 100ms it will sound discrete from the original. Depending on the reflective properties of the acoustic environment, the frequency response of the delay will vary. If the surface is glass or concrete the delay will contain a substantial amount of midrange and high frequencies. If the surface is carpet, the higher frequencies will be absorbed by the surface and the delayed sound will appear duller than the original.
If there is more than one surface you will hear a cluster of delays with their differing sound properties. If the cluster gets dense enough, it will appear to start to sound like reverb.

In a recording studio you can create this delay effect electronically. Most DDLs will delay a sound from 1ms to 4 sec. Most DDLs will have more than one output so you can create more than one discrete delay from a single sound source. When using multiple delays the longer the delay time, the less bright to the original it should be.

Low Pass Filter

This feature rolls off high frequencies at selected points so the delay sounds duller than the original. In a real situation the returning delay will always have less high frequency response than the original.

Regeneration

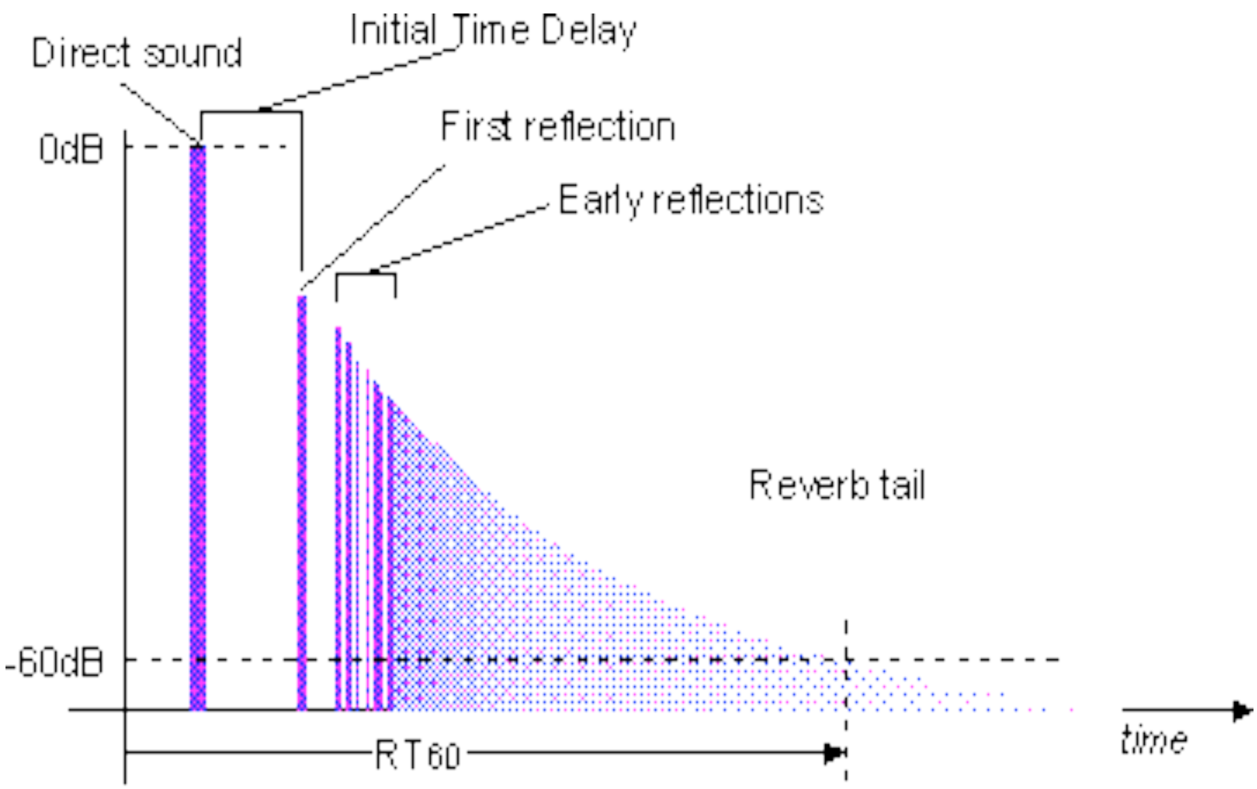
This function allows the delay to repeat itself at a predetermined delay setting. If the DDL is set to 200ms than delays of 400ms, 600ms, 800ms and on will be heard but at diminishing amplitude levels as their length increases. Using this function in conjunction with a low pass filter will create a very realistic delay.

Rhythmic Delay

When you want the delay to sound like it is in time with the music track, you will need to find the BPM (beats per minute) of the song and its corresponding delay setting. For instance, if the BPM is 120 than the quarter note equivalent is 500msec. If the BPM shifts, a rhythmic delay will not work. The faster the BPM the quicker the delay will be. The delay time is figured out by dividing 60 by the BPM.

Creating Dimension In Mixing

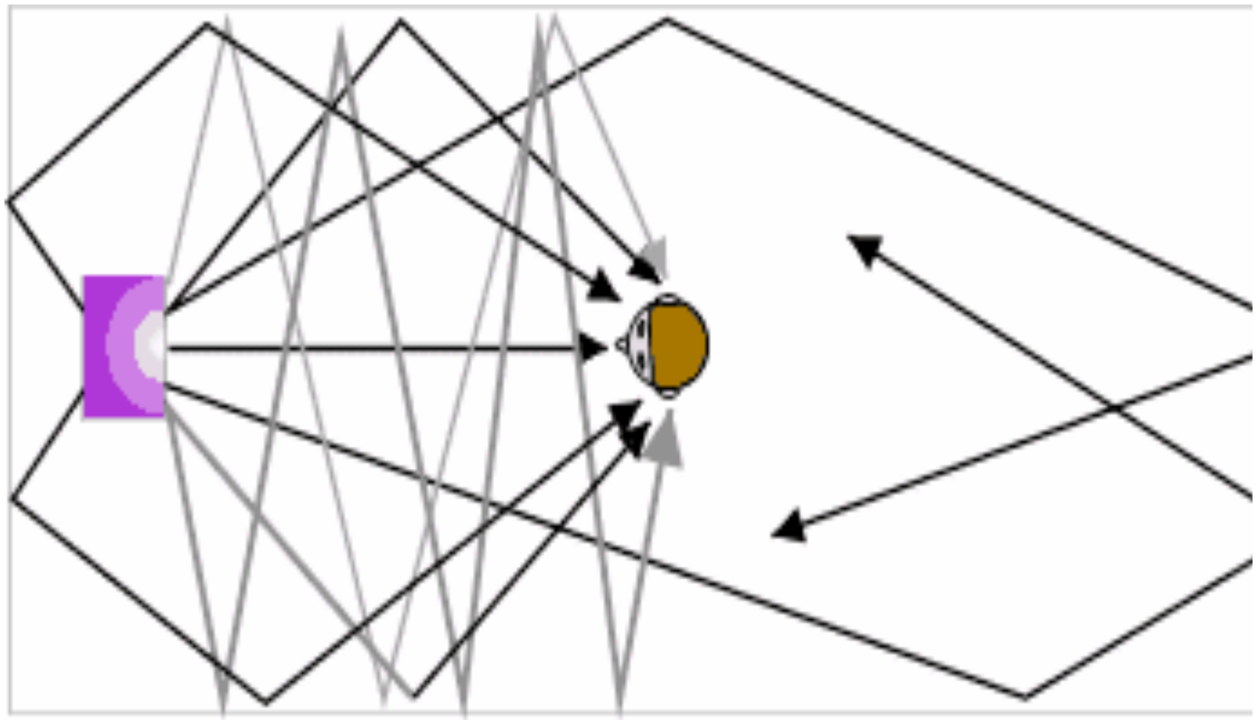
The best sounding mixes contain dimension and perspective where you can actually visualize depth in a performance. To achieve this you need to understand how direct sound, reflected sound and reverb work and how to create this depth with the original sound, DDL, reverb and EQ. Dimension is a combination of a series of multiple delays (reflections) and original sound. Once reflections get dense enough that we can no longer distinguish these as separate individual sounds over a short period of time, it turns into reverb effect. Reverb can generate sounds that are smooth and appealing to the ear. To use depth effectively you need to look at music sounding 3-dimensional rather than a 2–dimensional. To create this you need to incorporate an audio signal that is made up of original direct sound, early reflections and highly diffused reflections (reverb). With the proper structure of these elements, level, frequency response and time duration you will have the basic knowledge on how to create dimension in mixing. In this age of digital technology, artificial reverberation is not only more affordable than ever before but can also be stunningly realistic and very controllable. With a good understanding of the physics of natural reverberation and the fundamental operational principles of reverb processors, it is possible to quickly create the illusion of any acoustic environment you can imagine.



Breakdown of Reverb Waveform

How Are Reverb Times Determined?

How are reverb times determined? A reverb time or "RT-60" is the time it takes for a sound burst to decay 60 dB from its original level in an acoustic setting. Since 30 dB of ambient room noise is common, I like to say "the time it takes for a sound burst to decay from 90dB to 30dB". Time consuming and costly tests can be performed to see exactly what frequencies are reverberating and at times what times, but with a study of the interior finishes we can closely predict this now.



Relationship between listener and direct sound, 1st reflections, early reflections and highly diffused delays (reverb).

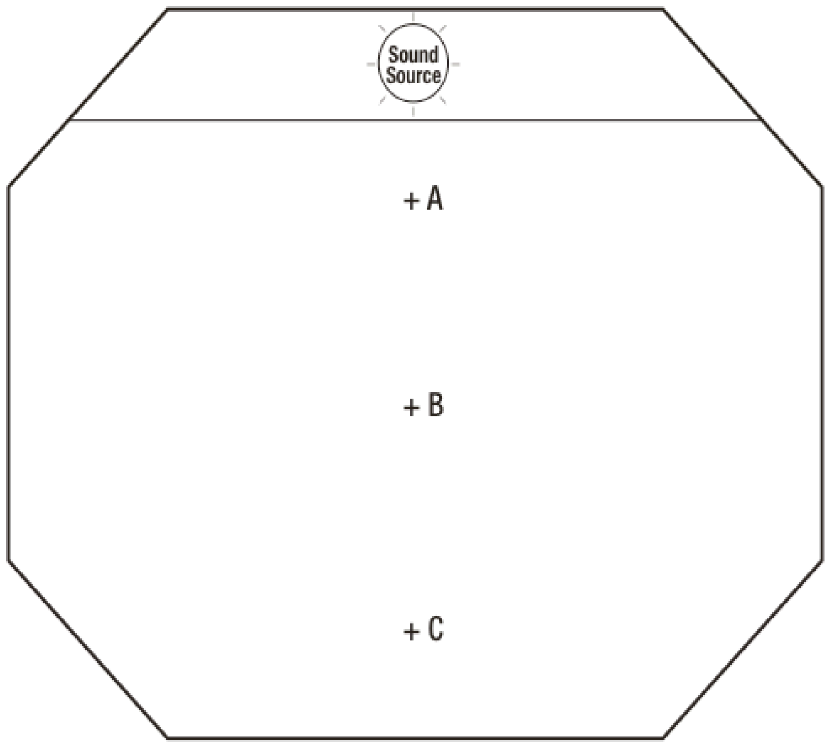
<u>When it comes to creating the impression of a believable reverb environment, what are the factors that matter most?</u>

There are three areas of reverb perception. Firstly, there's the whole issue of intelligibility and appealing reverb and what influences them; what makes something hard to understand and appreciate as opposed to easy to understand and appreciated. You have the sense of distance, which is influenced by nearly any time range. Direct sound energy coming in nearly any time period will cause a feeling that you are at some distance from the originating sound.

This distance effect will be made up of original direct sound, and its relationship to duplicate delays. If a delay arrives within 15ms of the original sound it will create imaging problems. For example, if you have a sound panned in the center and a delay of 1-15ms on the right, what you will hear is the image in the center shifting to the left. This is caused by the innate characteristics of human hearing in its relationship to localization. The ear perceives localization because a sound wave will arrive at one ear slightly later than the other ear. This is an innate survival mechanism for human behavior. It is otherwise known as the Haas effect. If a delay of 1-15ms is brought back and panned to the same position as the original you will create phasing effects.

If a delay signal arrives later than 15ms but before 100ms (approx) from the original sound it will create dimension, for what you have done is alerted your psycho-aural response, which tells you that you are listening to the sound in a reflective environment. Whereas, if you just heard the original sound only the psycho-aural response would create the effect that you are standing in a field. If you had a signal panned centered and delays of 40ms (left) and 60ms (right) it would sound like you were sitting at a distance with the left reflective surface slightly closer to you than the right. If these delayed signals are dull sounding it will imply that the reflective surfaces are absorbing the high frequency content and placing you in an environment of wooden walls rather than glass. If a reflection is heard after 100mS, you will perceive it as a separate form of sound energy, perhaps as a discrete delay depending on the nature of the source. It will also be easy to localize it in the stereo image, so if you have a reflection coming in at 200ms and it's panned to the left side, you'll hear it coming from the left. This is not true if it arrives between 15-100ms. If the delay occurs between 15-100ms it will not be perceived as a discrete delay, it will only create depth. If the delay comes from the left it will be difficult to localize the delay in the panning image, for it will not be perceived as a separate sound event although if you have very short percussive sounds such as rim shots, clicks or handclaps, you will perceive the reflection separately because of its transient nature and short duration time. If this occurs, you will need to shorten the delay times to maintain clarity. Generally, any reflections arriving between 15-100ms will not affect clarity. Adding reverb with delays will create a natural sounding acoustic environment (more on that later).

Listening Positions



In figure, you will see the setting of a concert hall with three different seating positions situated at fixed distances from the original sound source. In position “A” you will mostly hear the original sound source (80%), little early reflections (5%) and reverb (15%). In position “B” you will hear the original sound (60%) early reflections (20%) and reverb (20%). In position "C" you will hear original sound (40%), early reflections (30%) and reverb (30%).

In position "A" the original sound source (80%) will be full in frequency response and arrive to the listening position in 5ms. The early reflections will barely be audible because the listener is sitting very close to the sound source and the walls of the hall are almost all-equal distance from the listening position. The reverb will be delayed when it arrives back to the listening position. The actual time of this delay will be measured by the distance from the original sound source to the walls and then back to the ear (100ms pre delay). The reverb will not be bright. It will be warm sounding for the high frequency content of the reflections have been absorbed by the walls. Because the "A" listening position is not close to a wall and at a distance to the original sound source it will barely hear any early reflections. The delay of the onset of the reverb will indicate how far the walls are from the listener. The length of the reverb will indicate how live the environment is. The overall sound will be intimate, clear and pleasing to the ear, especially if it is a great singer performing a ballad. To create this in mixing you will need to add in a reverb that rolls off more high frequency content over the decay of the reverb. This means, as a reverb gets longer it also gets duller. A pre delay of 100-150ms is needed to create the effect that the singer is close but in a live acoustical environment. Watch out for sibilance in the reverb. Sibilance is just noise and will effect clarity in the mix. The way to get rid of this is to heavily de-ess the reverb send not the reverb return. Try to remove all sibilant information above 3kHz. Roll off the reverb return in the high frequency and low frequency area and maybe slightly boost around 2-2.5K to add a little presence for clarity in the reverb. If using a long reverb time that tends to thin out over time add in a stereo delay based on the rhythmic value of the song. If the song has a tempo of 120bpm, a quarter note will equal 500ms. It is important that when you add in a delay to your reverb that it be a fundamental of the rhythm for the landing of the beginning of the delay will also land on an instrument playing on the same beat. This will allow you to increase the delay to your reverb without noticing it as a discrete delay. Obviously, if the delay was 400 or 600ms you would hear the delay sounding discrete for it is landing in awkward places in the rhythm of the song. If you add in a de-essed, slightly regenerated quarter note delay to your reverb sound, you will add musical body to the sound of the reverb. Instead of adding a mono 500ms delay, add a stereo delay of 490ms (left) and 510ms (right). Make sure the 2 delays are at least 15ms apart to prevent the Haas effect. This stereo delay will enhance the effect of the reverb and still sound in time with the song. Also insert a low pass filter on the delay, so when it regenerates it sounds less bright on each additional delay and more believable to the listener for this is what truly happens in a natural acoustic environment. In total the original sound will be 80%, early reflections 5% and reverb 15%.

In listening position "B", the original sound source (60%) will have less high end and low end due to the increased distance between the sound source and the listener and arrive 35ms at the listening position. The early reflections (20%) will be arriving mainly from the left and right walls. The longer the early reflections, the greater the distance between the sound source and the listening position. Especially if the early reflections get less bright with the longer delay time. The early reflections inform the psycho-aural response by informing the listener that they are in an acoustic environment with at least two reflective surfaces. A delay of 60 and 80ms will indicate that the listener is sitting further away from the sound source than with a delay of 20 and 40ms. It is important that the delays reside between 20 and 100ms and be at least 15ms in difference to prevent the Haas effect and that the longer delay be slightly lower in level. The reverb will arrive to the "B" listening position sooner than the "A" listening position. This is because in the "B" position the time it takes for the original sound to arrive to the listening position is 35ms and the onset of reverb begins at 100ms. The difference is 65ms, which is what your pre-delay should be set to. The reverb frequency response will sound better due to the slight deterioration of the original sound caused by the time it takes for sound to arrive to the "B" listening position from the sound source. So to create this dimensional effect, make sure that the original sound source does not have an extremely wide frequency response. The depth will be created by at least two or more delays arriving between 20-100ms. The sooner the delay the closer you will be to the sound source. Make sure the delay has some high frequency roll off so the ear will not confuse the delayed signal with the original signal as being the focus. The reverb will have a smaller pre-delay and sound slightly brighter than the reverb in the "A" listening position. Be careful that the reverb time does not corrupt the harmonic content of the original sound source. Most instruments that require this effect will be playing in a harmonic content rather than a melodic content, like a lead vocal or a solo. A good rule of thumb is to make sure that the reverb time of an instrument playing harmonically is not too long where the mix becomes harmonically confusing.

In listening position “C” the original sound (40%) will arrive to the listening position (65ms) at a lower level than positions “A” and “B” and its frequency response will be even less than listening position “A” and “B”. That is not to say you should go out of your way to deteriorate the sonic quality of the sound. It is more like “Do not put too much efforts to make it sound full”. It should contain low end and presence in keeping with the character of the instrument. The early reflections will be longer than the reflections of listening position “B” (60-80ms). They will still sound less bright than the original sound but will be more prominent in level in the overall sound (30%). It might also be good to create more additional delays (120 and 160ms) beyond 100ms, but these delays should only add depth and not necessarily be heard as discrete delays. The reverb will also contribute more to the overall sound and its pre-delay will be even shorter. In the “C” listening position, the original sound arrives to the listening position 65ms later. Because the acoustics of the environment are fixed, the reverb should not change dramatically. With the onset of reverb occurring 100ms after the original sound source, the difference between the reverb arriving to the listening position and the original sound is only 35ms, which is now your pre-delay setting.

Overall as we move further back from the sound source the frequency response of the original sound source gets smaller and early reflections and reverb add to the overall sound. As you move further away from the sound source the reflections and the reverb increase in content to the overall sound. The distance in time between the original sound source and the early reflections and reverb will decrease. The overall sound source should always be louder than the reflections and the reverb for this is a fundamental rule in creating depth in your mix. If you chose reverb as a pre-send, the reverb that is generated will still contribute qualities of the original sound source.

The frequency response of the reverb dictates the acoustic properties of the reflected materials. If it is hard like concrete, the reverb will contain a lot of mid-range and high end. If the reflective material is wood it will mainly absorb high and mid-range frequencies. Many musicians prefer older concert halls because of their warm acoustical properties that tend to just reflect the musical content of the sound source. Remember that reverb works best when it is treated in a musical context. It can elongate the duration of beautiful melodies; it can create more resonance to drums and add perspective to various harmonic instruments in the mix.

If you have a sound source like singing and want the vocal to sound like a ballad performance, you'll find that you can create a recording where the singer sounds close or far away, at the same time. This is a very easy thing to do if you have no 1st reflections, early reflections and late reflections (highly diffused). You get this by close-miking the original sound and then adding delays, pre-delay to the late energy (reverb), and reverb with hall or a plate setting. If, for example, you use a reverb setting of say, 3 seconds with 100ms pre-delay on the onset of later highly diffused reflections (reverb), you'll find that the vocalist sounds very intimate, a sort of "in-your-face" sound but in a hall environment. If you like this reverb setup but wish to create more distance between the singer and the listener and don't want to change the reverb decay time, you will have to introduce a series of 1st reflections so you can slightly recess the singer. Just using a standard 3sec reverb setting with no pre-delay and no delays (reflections) will just give you a basic 2 dimensional hall environment without any sense of listener-to-vocalist distance, no matter how much reverb or length of reverb time you assign to the original signal. By adding the earlier energy and over a wider range we can create what type of a 3 dimensional sound we desire in our mix.

If you add in delays from 50ms-250msec you might create problems with the sound remaining intelligible and clear. However, you might want to utilize this for creating a slap back type of effect for lead vocalist. If used effectively, it will create distance between the original sound source and the listener. It is also important to decrease the high frequency response to keep the original vocal more present and clear and also allowing you to use more of the delay signal.

Decay Time

The length of time from the onset of sound after the initial sound has been established until it has dropped in level by 60db.

Pre-delay

The distance in time between the onset of the original sound and the beginning of the reverb sound expressed in milliseconds

Diffusion

If the diffusion is set too high (reflections very close in time) it will make the reverb sound very smooth. If it is low you might start to hear discrete delays that might clutter the sound.

Room size

The larger the number, the bigger the size of the reverb space. Certain programs will introduce more early reflections into the reverb algorithm.

Modulation Rate and Depth

Randomly shifts the times and its related intensity of the early reflections, creating a more authentic effect. If using a lot of this function you need to be aware of any pitch variances of signals with a lot of harmonic content.

Density

The amount of 1^{st} reflections and early reflections and the time difference between them. You also have control over the amount of this effect in the reverb mix. Often used for creating good room sounds for drums.

Frequency Controls

All reverb loses high frequency content over time. If you EQ a lot of high-end over the diffused part of the reverb, it tends to sound very unrealistic. In most "*plate*" and "*hall*" algorithms the high frequency response gradually tapers off over time. There are also frequency level controls at various low frequencies to keep the reverb from sounding muddy.

Gated Reverb

A setting where the reverb stays at one level over time and then suddenly shuts off. Often heard in snare drum sounds in the 80's.

MIXING

Most good mixers these days can start their mix process at any desired point because of their years of experience and their relationship with their monitors. When starting out as a mixer you do not have this experience and need to start at a reference point that will produce desired results for your mix. I have designed this mixing segment for those with little experience or are new to the mixing process.

Before starting a mix you need to have a vision of how you want your mix to sound. Refer to CDs with examples of what you are trying to achieve, for creative and tactical purposes this will give you guidance on where you would like to take your mix sonically and musically.

Near Field Monitors

Good near field monitors play an essential role in consistent referencing. The monitors should be capable of reproducing frequencies from 60hz to 17Khz and be able to handle high SPL, and set up in a triangular fashion 3-4 feet apart. Make sure the monitors are not too close to the plane of the console so to minimize high frequency reflections that will corrupt proper imaging. If you're using monitors that are not true in frequency response, equalize the monitors in the monitor stage (post fade) to allow for discrepancies. This will alleviate you from incorrectly EQing your mix to compensate for inaccurate monitors. Also the distance from your ears to the monitors should be set up so the room acoustics do not play a significant role in the sound of your mix. For example, if the monitors are too far away and the room is reflective your mix will sound too dry.

Outboard Gear

I like to start off my mixing sessions with at least three different reverbs, three DDL's, a stereo chorus effect and two extra stereo effects processors with many assorted stereo effects like phasing, flanging, etc. as well as enough analog comp/limit for processing acoustic audio. One good stereo EQ and stereo compressor are necessary for mastering my final mix. Two audio storage mediums, one for master and one for safety purposes for your final mix, such as a hard drive, DAT machine, analog 2-track etc.

Storing audio to digital should be done in the best sounding formats e.g. 24Bit/96Khz.

Setting up the console

1) Grouping – assign all tracks of similar instruments close to each other. For instance put all drum and percussion channels side by side. All guitars side-by-side etc. Mark all different instruments with different colors on the console strip. This will make it easy to recognize and locate certain instruments easily. Try to group all hard drive returns to the center part of console. Things like solos and lead vocals that require a lot of fader moves should be placed in the center of the console for optimum monitoring purposes. Patch all outboard gear to the outside channels, e.g. 1-8 and 29-36 for they only need to be set to one optimum level. If you have the time and will be mixing for more than a couple of days, insert 1Khz tone at 0VU into each input strip placing the fader at 0VU position to check line cleanliness and continuity.

2) Setting up Line Amps - First bring up all channels to a basic rough balance with the priority music tracks such as lead vocal to a position where the lead vocal sounds cleanly audible with another 10dB of fader headroom. Now fine tune all line level amps (-5dB to –20dB) so all faders are in maximum working range. It is very hard to make detailed level changes when the fader is close to the bottom. Allow 10dB of headroom on all faders.

3) With a priority track such as a lead vocal, bring the lead vocal up on one channel and buss it to another input. This will allow you to control the level of the vocal before any processing. In the first vocal channel you can roll off low frequencies such as rumble (60hz), proximity effect, etc. In the second vocal channel insert limiting, equalization and compression and any de-essing, if necessary. If a vocal needs to be compressed whereby the choruses are recorded significantly louder than the verses, what will happen is that the vocal in the verses will not be compressed at all. Or, if you set compression on the verse vocal, the chorus vocal will be overly compressed and very thin sounding. Remember the more you compress the signal's

quality tends to be reduced. If all verses are similar in level and all choruses similar in level but a lot louder designate one channel for verse and other channel for choruses. This same approach can be used for solo instruments or anything that will be a priority in the mix.

Starting the mix

At this stage you should have a basic idea of where the focus of the mix resides. If it's Norah Jones, it will be the lead vocal and the piano, for hip-hop it will be the groove, the bass and vocal, for rock it will be guitars and vocal. Whatever the focus is, it should get the best treatment such as good analog equalization and compression. I have yet to hear any digital equalization and compression that sounds as good as analog. If dealing with someone like Norah Jones, listen to similar sounding albums in that genre of music. Try to approximate the equalization, compression and reverb of the sound that you desire. Remember that you will most likely be processing it further and the object here is not to emulate totally, but to start you in the right direction. Next, you would bring in the piano, respecting the fact that the vocal will take precedence in the high frequency range (presence). The piano should sound clear but not override the high frequency of the vocal. A good way to test this is to listen to the piano without the lead vocal and if you feel it is a little dull you are on the right track. As soon as you start to make the piano sound like the focus you will have to EQ more high frequencies into the lead vocal. This will obviously make the vocal sound too bright and thin where you're actually separating the sonic qualities from the musical qualities of the vocal. From there you can then turn the lead vocal off and build the sound of your rhythm section.

Also in this stage you need to assign your instrument breakdowns to group to fader masters. This will allow you to make level changes and mutes on groups of instruments as a whole. If using a moving fader system, assign your lead vocal channel to a group master even though it is only one channel. If you have made a lot of fader moves with the vocal channel in a verse and now realize you need to bring up the lead vocal for the entire verse, having a group master will make it easy for you.

Drums

You need to decide where the drums should fit into your mix. Should the bass drum be tight with the bass by introducing the rhythmic or attack part of the bottom end. This will allow the bass guitar to be warm and full in the bottom end that tends to work for a lot of pop tracks. A common mistake is to EQ too much low end on the bass drum and not enough on the bass guitar. This will give you the illusion that your mix is bottom light for what you are doing is shortening the duration of the low frequency envelope in your mix. Also, the bass drum tends to be more transitory than the bass guitar, giving you the idea that the low frequency content of your mix is inconsistent. Should the bass drum need more resonance and depth to it, adding in ambient mics or short reverb programs will suffice. One thing to make sure in your mix is "do you want the bass drum to be felt or heard"? EQing in the 30-60hz range will produce a "feel" bass drum but will sound very thin on smaller speakers. If you EQ the bass drum between 60-120hz it will allow the bass drum to be heard on smaller speakers. With it you want to get a lot of "hear" low end and attack sound between 2-4khz and also dipping between 300-600hz range which contains a lot of unnecessary overtones. If the track has enough space in it, you can factor in a tight verb or a tight ambient room for you will be able to hear it. If the track is dense, don't bother to try and create one for it will just take up space and clutter the bottom end of the track.

What sound should the snare drum have? Should the snare have a lot of reverb to make the backbeat sound longer in duration or short and percussive? Do you want to mix in a lot of room ambience that is triggered by the snare to make the snare drum sound bigger? (see Gating). Do you want to compress the snare drum to get more sustain? If you desire this effect you will need to bring up the snare on 2 tracks, one for the attack sound and another channel to first gate the snare and then compress the snare with a fast attack and fast release time. You might want to gate the snare and compress the overhead mics (keyed by the snare) to remove snare leakage from the overheads without making the hi-hat sound too ambient. You might also want to gate the toms for cymbal leakage especially if you use condenser microphones on the toms. Also, gating the snare reverb send will minimize the hi-hat from washing out the reverb. If the transients of the drums are random and excessive you might try to buss comp/limit the drums to control the transient excursions and minimize the dynamics in the performance to maintain a consistent level from the drums. Adding rhythmic delays to the snare might make the groove more interesting.

Bass

Once you have finished the drums, you can add in the bass. For pop music it is best to have the bass drum provide the percussive nature of the bottom while the bass fills out the sustain and musical parts. With the bass you will want to find a balance between the amp and the direct sound. The amp sound will give you an edgier quality where the direct sound will give you a fuller sound. With EQing the bass for low end should be between 80-120hz for you will want to hear the bass on smaller monitors. Remember to check phasing between the DI and the amp signal. Compression is a good idea with the ratio of 2:1 - 4:1 with a medium attack time and medium-slow release. With a medium attack time you will allow the percussive nature of the bass to be heard. With the slow release time you will have the low end sustain. The release time should be long enough to avoid half cycle distortion. If you need the bass to sound more musical you will need to EQ in the 400-800hz range, and for getting an edgier sound EQ between the 2-3Khz range should suffice. Remember to EQ before you compress.

With hip-hop music, the bass tends to be a feel bass with a lot of information in the 30-60hz range. Also minimizing sonic information in the musical range and the mid range will remove any actual music information and the attack of the bass. Synth bass is very popular because you can create an even balance between 30-60hz and elongate the duration of the note to create the illusion that you have more bottom end. On some of the better hip-hop records they will raise the low frequency target area slightly higher to the 70-100hz range and elongate the duration to create the illusion that there is a lot of bass information so that it can sound full on smaller monitors. Be careful not to over-EQ the bottom end so it will sound good in clubs or in cars with huge bass drivers. These kind of audio systems already hype the feel frequency range of the bottom end. In compressing hip-hop bass do not be afraid to use a lot with even higher ratios. The goal is to have the bass loud and as even as possible.

With rock bass the idea is to create an aggressive in your face bass sound. For this you will focus mainly on the amp sound. Trying to mix in the DI sound with the amp sound might cause phasing problems in the mid range that will be detrimental to what you want for your bass sound. With your sound you need to get a consistent bottom end and a lot of mid range. Boost anywhere between 50-100hz for the bottom end, dip between 400-

800hz (this will allow the guitars and vocal to have more room to speak musically) and boost between 1.5-2.5Khz for mid range. Be aware if the bass player is using a pick instead of his fingers for it can create uncontrollable audio transients in the mid range. With compression, you need to use a lot (4:1 - 8:1). If the player is using a pick you might need to limit the transients before you compress. The attack release times will have to be fast (listen for half cycle distortion) in limiting and medium to slow for compression. Sometimes it's a good idea to put in multi band compressor over the bass to target specific frequency areas. If you also recorded the bass direct and you needed a more aggressive sound for your mix, try sending the direct signal out to an amplifier in the studio that can be miked. This will allow you to modify on the spot your bass guitar sound to your needs.

Piano

In a situation like Norah Jones, the piano will be second in priority behind the lead vocal. The piano will be spread fully across the stereo image. When getting the piano to be present you will need to EQ the mid range and high end. When starting the mix you will have already ball parked the lead vocal EQ and have approximately EQed the piano in relation to the lead vocal. So when you add in the piano to the bass and drums and if it sounds dull, EQ the piano slightly brighter and you will most likely be okay, for when you started out, you allowed yourself a certain amount of head room in the mid and high frequency range for the lead vocal. If you find that the piano needs a lot of high frequencies you have obviously over EQed the bass and drums in the mid range and high frequency. If this has occurred pull back the boosts in mid range and high frequencies on the bass and drums. The problem will most likely be with the overheads and snare. Remember, in dealing with the snare your dealing with a lot of high frequency information over short time duration. So instead of adding more high end EQ over the snare's transient, try limiting the snare which will allow you to elongate the high frequency content of the snare drum's duration and create the illusion that it is brighter. Here's another solution, if the snare is sounding the way you would like in the high end and you do not want to reduce the level of the snare try compressing the snare with a medium attack time. This will shorten the duration of the snare but will not sacrifice the rhythmic transient of the snare drum that is integral to the overall drum performance. This gives the illusion that the performance has not been sacrificed rhythmically or musically in the mix but the snare drum still sounds bright.

Guitar

In a situation like Norah Jones the guitar performance on the bed track was tailored to support the piano and vocal in a musical and rhythmic fashion. Just bringing the guitar track up to balance it in the track should be easy to do. For the guitar player has designed his performance rhythmically and harmonically around the vocal and piano phrasing. The only potential problems that might occur is if the guitar is not present enough and/or loud enough throughout the performance. A solution is to add presence in the 3-5khz area factoring in the fact that you do not want to have a build in the frequency range between the guitar and the piano. If you notice the guitar is getting lost in places, try compressing in the 2:1 - 4:1 range with a medium attack and release times. This will allow the rhythmic transients to go through unobstructed while raising the sustain resonance of the guitar. If the guitar is soloing in an expressive manner you might require a bit of limiting first. Also add in processing to create depth perception of the guitar remembering that the piano should be forefront to the guitar. A quick solution is to add a stereo delay with setting of 40ms hard left and 60ms hard right with a short reverb. Remember to roll off some of the high frequency content on your delay returns. This will create the illusion that the guitar will be sitting further back in the mix than the piano without creating noticeable level discrepancies between the piano and the guitar. With a pop track where the guitar is not the main focus, but is there to add rhythm and harmony, EQ it in a range that is not as wide as the main instrument. Avoid EQing in the very low and very high frequency ranges. Balance its level against the piano so it sits comfortably. If you feel it needs to sound further back in the mix and you do not want to lower its level, try an assortment of these effects: add in short delays (15-100ms), unnoticeable rhythmic delays (eighth note or quarter note), chorusing and reverbs with little pre-delays.

Mixing the Bed Track

(Norah Jones) Once you have EQ'd the drums, bass and guitar and have placed them in their proper perspective get a balance on the drums, bass, piano, guitar and lead vocal. Start factoring in processing such as reverb, chorusing and delays to create depth perception in your mix, allowing yourself a little more headroom for further enhancement. Remember mixing is a building process that requires constant sonic evaluation throughout the process. It is important that you incorporate mutes or level changes at this stage though automation. Once finished this basic mix of all bed track components with the lead vocal you should have a mix that should be able to stand out on its own for these are the basic elements of the song. If you have not achieved a satisfactory product by then keep working on it and do not expect that adding in any additional musical elements will make it better, it won't! All you will do is create a confusing and unprofessional mix. A good idea is to refer back to the monitor mix you did on the date you recorded for in a lot of cases there are certain things about the monitor mix which will sound better then where you are at now with your mix. You will easily discover if you have over EQed or over processed any elements that might separate the sonic components from the musical components of the song. Remember that you might need to continually reference your lead vocal sound against other outstanding albums. Then prioritize what is important to the lead vocal. In a case like John Mayer it will be the guitar and the vocal. In hip-hop music it will be the drums, bass and vocal. If you maintain this philosophy mixing will always have a creative rather than a redundant approach. One critical component of creative mixing is remaining in a creative headspace. If you get your bed track balanced with your vocal, automate it to sound like a final mix. This will remove repetitive redundant moves that the brain should not be focusing on. It is hard to be creative when you are preoccupied with making level changes that you know could be automated. The strategy here and until the end of the mix is to keep the creative process alive.

Backup Vocals

Recording backup vocals is fairly easy if the vocalist understands their objective how to work with the lead vocal performance. In the case of the lead vocalist adding a double track in unison, you should record with the identical set up that was used for the lead vocal. When adding in the double track, mix it at a level below the lead vocal and be prepared to not make it as present as the lead vocal. The goal here is to add more musical body to the vocal performance. If both vocals have the same presence it might confuse the listener to which vocal is the lead. When adding in the vocal double you will lose presence to the lead vocal but will achieve a vocal performance that will be more forgiving in pitch.

If the lead vocalist is adding a harmony to their lead vocal melody it will usually be the 3^{rd} and or the 5^{th} and sometimes the 7^{th}. Record the vocalist with the same set up used for recording the lead vocal. When adding in the harmony it will always be at a slightly lower level to the lead vocal.

With two or more singers singing harmony to the lead vocal they can perform in two ways. One is for the backup singers to sing the same harmony part at one time. The other method is for the singers to split the harmonies amongst themselves at the same time. Double or even triple tracking harmony parts is very popular and can best be heard by groups like The Bee Gees and The Eagles. If the backup vocals are singing counter point to the lead vocal you will want to have them as present as the lead vocal. When recording three or more tracks of backup vocals it is best to submix the parts to a stereo bus and bring up the stereo bus into two additional channels. This will allow you to put exactly the right amount of processing on all backup vocal parts rather than guessing at sends and EQ levels on each individual track. Remember to clean your backup vocal tracks before mixing for backup vocalists like to sing a pitch reference before they sing their part.

Solos

When an instrumentalist is soloing they should have the same perspective as the lead vocalist. In other words, when they are performing their solo they should stand forefront in the mix. The only exception to this is when you want the soloist to sound like they are soloing in a band performance. This usually happens when their bed track performance is replaced by soloing. This can be heard in punk and rock music. If the soloist is a lead guitar, saxophone or another instrument make sure all parts of their performance can be heard. This usually requires a bit of limiting, EQ and compression. For effects, I usually will use delays, reverbs with pre delays and other forms of processing. If the soloist is performing in a call and answer style you will need to make sure that they are slightly less present than the lead vocalist but more present than the rest of the instruments.

Adding-in additional instruments

Before embarking on the next step, review the status of your mix and make sure it sounds finished. If for example you have made the decision that the vocal performance in the second verse needs to be louder than the first verse and you don't make that level adjustment then, how will you know what levels to set for any additional instruments coming in at the beginning of the second verse? For example, if you added congas in at the second verse and you have not made the lead vocal level change you will most likely mix the congas in at a level relating to the drums and the lead vocal. When you start automating the mix and increase the vocal level in the second verse what happens is that the congas will be lower in level than where they should be and in most cases you won't even notice. By the end of the mix, the conga performance will be at a level where they are just taking up space instead of lifting the rhythm at the second verse. Automate all moves and mutes when ready. This will make it easier to place additional instruments in the proper perspective.

When adding in strings be careful not to put too much reverb on them. This will prevent their performance from creating harmonic confusion and keep them articulate sounding. If you need to recess the perspective of the strings use a short reverb or even a DDL. You will most likely need to ride the level of the strings especially with the violas and cellos due to their harmonic placement in lower registers. If you need to compress use 2:1 to 3:1 ratio with slow attack and release times.

If you're adding in horn sections be careful to watch for transients especially from trumpets. Due to the complex frequencies of horns it is best as with all additional instruments to try and ride the levels before using any dynamic processing. In the case of horns where transients are very fast you will often have to use fast limiting. If adding reverb use short reverbs (1-2 seconds) that are bright sounding.

With percussion the idea is to make sure that the attack part of their performance comes through cleanly and relatively even. With parts like congas percussionists will perform with a dynamic range that often cannot be translated in a mix. If the performance is 16^{th} note in nature and perform on 2 or more congas you will most likely have level discrepancies between the congas. If you solo the congas on their own they will sound fine but hearing them in the mix you will not hear an even balance between the two.

To solve this, use compression with fast attack and fast release times to even out the dynamics.

Woodwinds such as flutes, oboes and clarinets are very warm sounding in nature. They often don't need any dynamic processing and if they do it is very subtle. When a flute plays in a high register you might need to compress. Piccolos on the other hand should be burned at first sight. With perspective medium to long reverbs with pre delay work quite well in keeping woodwinds sounding warm and natural.

Finalizing the mix

When you have finished your mix make copies of the mix and audition them on other monitoring systems like a ghetto blaster, a car stereo and home speakers. If you have the time, give your ears a rest. I like to leave the mix set up over night and come in the next morning with fresh ears to do final adjustments, which I tend to always do. Do not belabor your mix, which means no endeavors to seek perfection. Believe me, you'll most likely be the only one to notice. Early in my career I would present a mix to the client for their comments which would often be "sounds great" and then inform them I only have a couple of minor adjustments to make. After spending four hours on the mix I would spend another eight hours making my minor adjustments and present the updated mix to the client who would comment, "we can't tell the difference". Perfection, I have learned, is the ability to present something in its simplest form that can be appreciated to its fullest extent. Listening to some of my favorite recordings I have noticed mistakes but who am I to remix Sgt. Pepper's. I might mix Sgt. Pepper's perfectly but I know for certain it will sound nowhere near as good as the original mix.

Try to play your mix to normal people who buy CDs because they like the music, which means avoid your techy friends who might steer you in a direction of technical merit that might not make any musical sense. If you are having problems with your mix by all means reach out for advice to your trusted peers for their subjective and constructive feedback. This is not the time to be a sensitive new age drama queen worrying about your feelings getting hurt. This is a time to be honest and open minded and welcome suggestions that you're willing to put into action.

Rock mixing

With rock mixing the goal is to get your song sounding big and powerful, by incorporating the full frequency range and limiting the dynamic range. To achieve this you will need to dynamically process each element on it's own. Try using the limit-EQ-compress process, which will allow you to basically just set levels and keep them there. With drums, subgroup into 2 stereo pairs including all original and perspective elements. On one stereo subgroup limit all the transients and do not be afraid to do a lot of limiting. You will need to incorporate a very fast attack time and a release time that will allow the signal to return to unity gain before the onset of the next transient. This process should sound as transparent as possible. On the other stereo subgroup use massive limiting with a very fast attack time and very fast release time with the goal of elongating the duration of the drum sound. The goal here is to limit so you can master as much level on a CD and create a bigger drum sound by sustaining the sound of the drums that do not add any more level to the transient. When you're adding the sustain limiting to the transparent limiting, you will notice that the overall peak level of the drums does not get any higher but the drum sounds gets bigger.

With rock guitars, the idea is to have them big and "in-your-face". This is accomplished by first limiting the transients out of the signal especially if it has been recorded to a hard drive. Recording to analog tape solves this problem through tape compression. Try limiting with ratios 10:1 or higher and use a lot. Be careful to make sure that the sustain parts of the signal return to unity gain. Next, EQ the guitar in the 3-5khz range for presence, for the low end 80-120hz. With EQing the low end, listen for out-of-control bass levels of the guitar, which are caused by turning up the bass control on the amp to 11. When this happens certain low frequencies jump out at loud levels while others remain unaffected. If you notice this occurring you need to roll off the low bottom end first before any other processing, which will allow you to manage the dynamics of the guitar. If you do not do this before compression you will most likely corrupt the harmonic content of the guitars performance. If the guitar player goes from a G6 (an open E on top) chord to an open E chord the low end might increase due to the fact that you are playing open low E string. When striking the open low E chord you are also playing an open E and an open B remember that when you played the G6 chord you also had an open B and an open E that were harmonically balanced against the low G. Due to the fact that the low open E is much louder than the low G the compression will bring down the open E and the open B. So what is occurring is that even though there is an even balance change between the G and E low notes, the open E and open B notes of the chords are much different in levels. Rock guitars tend to not require a lot of perspective processing, if any processing is desired it will be effects like chorusing, phasing, etc. These days some of the actual distortion processing found in effects boxes sound pretty decent. Mixing in this type of processing with an amp sound on a performance can produce huge guitar sounds. What the processing can bring is that "in-your-face" component to the sound with the amp adding the resonance of the sound. A major problem with this is phasing. The return of the processed signal and the amp sound are not exactly in phase over the entire frequency spectrum. A solution for this is to double track your guitars, have the processing tracks panned hard left and hard right, while the amps sounds reversed in panning. This will allow the processed sound of the performance to be panned to one side and the amp sound from the performance to be panned to the other side, ultimately removing any phase discrepancies. This is great if you are working at home recording processed guitar sounds and taking a direct signal at the same time allowing you to record your performance through various amps while mixing. With leads you might need to limit the transients first. With processing subtle stereo chorus, rhythmic delays and reverbs (with pre-delays) will enhance the sound significantly.

With guitars and bass the limit-EQ-comp works quite effectively even with guitar sounds that sound very compressed from amps like Marshalls. With solos you will tend to limit a lot due to their transient nature. If you stand in front of a Fender Twin Reverb while a guitar player is soloing on a Strat you will hear what I mean.

With lead vocals processing with the limit-EQ-comp works quite effectively when used extensively especially if the singer has recorded with a dynamic mic. When rock singers sing out, their throat tightens and when recorded with a dynamic mic it can produce transients between 1.2 - 2K. What might help here is to use a multi band dynamic processor. This will allow you to turn your mix up to a level that will rival a 747 without the vocal tearing your head off.

When processing for perspective in rock music, reverbs should be short if used at all. A common effect for lead vocal is a rhythmic digital delay that enhances the rhythm of the performance and adds depth to the vocal so it can sit further back in the mix. With bass and drums subtle use of DDL and short reverbs will aid in placing them in the right perspective. Be careful of over EQing the mid range and the high end especially if there is a lack of 3rds played on the guitars. At some point you will start to separate the sonic elements from the musical elements. A good example of this is to compare a song by Billy Talent, Tea Party, and Green Day with a song from Led Zeppelin, Tool and Dredg. When you're finished remember to compare your mix with successful mixes.

Hip-hop mixing

Hip-hop music is comprised basically of grooves, bass, vocals and little harmonic content. The goal in hip-hop is to get the rhythm to be the focus point, a good working relationship between the groove and the vocals. With the groove a lot of the EQ is spread over the entire frequency spectrum, from 30Hz to 17K. The bass and bass drum are designed more as feel than to be heard, with little presence on the bass drum and the bass. The duration of the bass drum is quite long in comparison to other genres of music creating the illusion that the track has a lot of bottom end. There is a lot of dynamic processing on the bass and the groove to keep it at one consistent level throughout the song. When starting a hip-hop mix begin with the bass, drums and vocal. You should achieve a balance between these elements that can make the mix stand out on it's own. Next mix in the harmonic elements of the song as in the case of Destiny's Child's new song "Lose My Breath" there is an orchestra pad that plays only two chords and is used periodically throughout the mix. I believe if you add in a lot of harmonic information it will require the vocalist to sing in tune. A lot of hip-hop music these days is sung with one note in a rhythmic pattern based on the bpm of the song. It seems fortunate that anyone with a sense of rhythm but tone deaf can be a hip-hop singer. With the vocalist there is no perspective processing and if any EQing is used it is in the mid range and high end. A lot of hip-hop singers like to hand hold dynamic mics while rapping which slots the sonic nature of their vocal in the mid range area because of the frequency response of a hand held dynamic mic. In the mastering of hip-hop a lot of dynamic processing and EQing is done. If you follow this basic formula you will not be surprised to discover that you can mix hip-hop as well as any body out there.

Mastering

In these times mastering engineers are a dime a dozen. It seems anybody with a DAW can call themselves a mastering engineer. The good mastering engineers with excellent equipment are very expensive. Personally I know only one excellent mastering engineer in Canada and probably one of the best in the world who is Peter Moore of the E Room in Toronto. He is an excellent mastering engineer because he understands music. Mastering engineers like Bob Ludwig, Steve Marcussen and Bernie Grudmen will charge you $7-$10,000 to master a CD. However if you follow a few simple rules you can closely approximate their value in the mastering process.

These days the goal of most mastering is to get as much signal on a CD as possible. If you look at the waveform of this type of mastering, it looks like a square wave due to the hard/wall limiting. This type of limiting gives you the "in-your-face" type of sound. I find that most of this type of mastering sounds very thin, very squashed and just plain awful. There are better ways to achieve this effect that can be utilized in mixing.

If you want your mix to sound loud and good try to limit the individual source material rather than as a whole mix. For example, if you notice that the drums are the first instruments that are being limited consistently try to achieve this effect in the mixing process. If you don't what happens is that every time the limiter kicks in by being triggered by the drums everything else in the mix at that point goes down in level in relationship to the drums. If you bus limited the drums in the mix you will discover that your guitar sounds will be consistently even for they are not being consistently lowered in mastering due to the limiting process of the drum's dynamics. Remember the more you comp/limit your mix, the more you corrupt the fidelity.

If your mix is dynamically all over the place try re-recording the mix as a new mix. I have heard mixes sound even up until the bridge/breakdown where instruments drop out and then when it gets to the end chorus the level has been turned down so the chorus does not sound jarring to the ear when it comes in, due to the drop in level in the breakdown section. A good way to avoid this is to spot check the levels of your mix throughout. In most situations the mix should gradually get louder from the beginning to the end. If the mix is even but still has dynamic problems with transients and overall levels, limit the mix first with a very fast attack and fast-medium release times. Watch out for half cycle distortion. If you are doing this while you are mixing the best way to spot this is to solo all instruments that are high in low frequency content. If you notice a grainy sound extend the release time to where you no longer notice this effect. If you have the luxury of a DAW try re-drawing the transients of your mix so no limiting is necessary. This is a process that requires a certain amount of expertise that comes with a lot of practice.

After limiting make any EQ adjustments, you do not want to EQ before limiting for this require the limiter to work harder. When EQing make sure you reference your mix to your target audience. Be careful not to make it too bright for you will begin to separate the sonic nature from the musical nature of the mix. I feel it is unfortunate today that most music tends to sound too bright and in the years to come I feel the artist and producers will regret what they did in mastering. I believe there are still great sounding records out there like Norah Jones, Alicia Keys, and Linkin Park, etc. that are mastered with sonic quality in mind. After EQing insert any compression if necessary. When compressing, use medium-slow attack and release times for it will allow you to get level consistencies that will make your mix sound louder and transparent.