Recording and Mixing Drums in the Modern DAW

Introduction

I'm pretty sure I'm not alone when I say that my drum mixing techniques have evolved through many incarnations over the years and pretty much change every few projects or so. This is probably due to the fact that I'm still learning, have no qualms about that, but as time goes on I, just like many others, focus myself to refine and rework my techniques so as to strive for better sounding mixes.

Most of sound engineering/music production is a long journey of trial and error where we turn, push and twiddle our way towards a better way of tackling problems and taking our craft to more creative heights. It becomes quite clear very early on that the frustrating truth is that most of this stuff cannot actually be taught through a book and even if it could, it doesn't make a difference until our ears catch up and can actually hear it. No matter how technology progresses and no matter how many people get home studios there will always be the barrier of training your ear through constant hard work and careful objective listening. These two concepts along with humility coupled with a keen willingness to learn are essential ingredients to achieving excellence in the field.

Of course, knowing the fundamentals and even a knowing them a little more is crucial to a better understanding because, at the end of the day, audio theory and all the technology that surrounds it is a bona fide science that can all be communicated with mathematics. Fortunately, you don't need a Ph. D in mathematics to become a competent mix engineer because we were born with our ears to decode these pressure waves and help us to be creative instead of worrying about the minutiae of the little picture. For in mixing, it is the *big* picture that counts.

In the past, being an audio engineer was an esoteric endeavor where a carefully selected aspiring applicant would learn under a mentorship in a professional environment and eventually, after making lots and lots of coffee, succeed his master into being a fully fledged sound professional. In our particular country mentorships are few and far between which results in a lot of us learning this stuff on our own or going to audio college, but as my experience was later to reveal to me, many of the industry secrets that you do learn along the way as a career audio engineer are simply well thought out and carefully executed fundamental concepts that are easily overlooked or sometimes entirely forgotten. One piece of advice? Always jump in the deep end and do the most challenging or intimidating projects you can, and always take opportunities to learn from professionals with more experience than yourself. And remember:

Professionalism is an attitude, and a skill set - not a platform.– Tim

Halligan, thewombforums.com

Everything counts

The sound of the drums has always drawn my attention more than any other aspect of recording. It is indeed my favorite part of the album production process and nothing quite satisfies me like the sound of well recorded and well mixed drum sound. I'm sure from your own experiments that you've all discovered that getting that particular drum sound is a lot harder than you'd expect and requires a huge amount of trial and error to achieve even ‘acceptable’ results. In the case of drums, once you get in there, dealing with multiple microphones and phase issues, controlling the broad dynamic range onto disk/tape at the risk of distortion or clipping, you have the undesirable task of meeting the expectations of yourself and your clients by manipulating them during the mix phase into sounding larger than life or at least suitably so to serve the material.

Now, not all drums are required to sound larger than life. Some drum sounds are best left natural, or at least mixed to be perceived as such, provided they were recorded sufficiently enough to do so. Something tells me that an improvisational jazz record would not benefit from extreme EQing and heavy compression. Or will it? But what about those instances when the drums need that larger than life sound and bold and captivating energy? Is there one single idea or secret to it, or is it a combination of things that gives the mixed representation of the drums that certain energy and magic? In the most sober of senses, the answer is simple:

Everything counts.

Your Sound Is Only As Good As Your Source

Most of the substance of a good recorded sound (and this goes for just about anything) begins at the point you place the microphone in front of the source. Finished and klaar. There is no such thing as an ‘acceptable’ sound that you are willing to commit to tape so it can be manipulated into a ‘better’ sound later. Take my word for it. Set your goals high. Unfortunately, the sound you capture at the source is what you are stuck with and should not be viewed as a ‘starting point’ but the first shot at achieving whatever sound you’re going for. This is achieved by having the best microphones possible, placed in a well thought out manner to capture the drum kit in it’s entirety, coupled to good microphone preamps and finally (if you’re using a digital system) the best possible digital audio converters and speakers you can afford. This may seem daunting as not all of us have access to the likes of Neumann, DPA, Neve, Lavry or Klein and Hummel, but there are ways to achieve great results with modest gear. Will it sound better than Steely Dan? Probably not. Will it get your project across effectively? Yes, with the right considerations.

Hands down, the best way to start improving your sound is by buying the finest matched pair of condenser microphones you can afford. The reason for this is that while upgrading your hardware (computer, mic pres, outboard gear, converters, etc) may occur at regular intervals a good pair of microphones will serve you for a lifetime and supply you with the quickest way to a better sound you can rely on. Of course, good quality microphone preamps and digital audio converters are essential to achieving excellence as well but the range of quality is narrower than that of microphones and most decent specimens that are available now in your local pro-audio shop are entirely usable.   After all, the top shelf specimens are usually *way* out of most aspiring engineers' budget. That being said, it's probably a good idea to invest in at least a top quality DA converter (like the Benchmark DAC-1, +/- $1000) and some decent monitors because they are the last stop before your ears and therefore it is critical that these are representing the audio faithfully.

Don't underestimate the room!

Drums hugely benefit from room ambience. It is for this reason that recording the drums in the best sounding room you can muster is a good idea.   Generally any obvious ambience considerations are best left for the producer but suffice it to say that the room shouldn't have many resonances or dips in the frequency spectrum, particularly in the low end. Typically drums in modern pop/rock are recorded in a large, bright, wooden floor space, on a carpet, with baffles placed to control the amount of room sound that enters the close microphones. Room mics are then placed in such a way that the drum kit is picked up as a whole and the ambience of the room is captured to separate tracks. If space and time permits, I generally start with the room mics when getting sounds because I like to first hear the drum set's sound as a whole.

Often, finding the right place to put the drummer in the room is a listening exercise and the only way to do this is by the process of elimination unless you are familiar with the room. You might have to move the entire drum kit a few times to different portions of the room before you come to a conclusion that a specific spot sounds the best. All that is required is to listen to the drummer play each time and make a call based on what you hear. It's simple concept, really. You are basically listening for what you envision the mics are going to hear, place them there, and then *actually* listen to them. It's the only way to ever know that you are capturing what you want before you press record.

Finding the right spot

Finding the right spot to put the close mikes is a similar exercise which sometimes requires an assistant. The fact of the matter is that there is no way that you can eyeball where and how to place your close drum mics. It's just not humanly possible. You may get lucky. It's entirely possible. But the only way to truly know is to move them around the relevant drum until you capture the right tone and the least amount of bleed. Every room and drum set setup is different so they will come with their own set of variables. These can not be circumvented by speculation so it is wise to always find that sweet spot before pressing record.

So, once we have assembled the best possible components we can in our recording chain, placed the drummer in a comfortable position in the room that sounds best to our ears, it is time to start miking. These are generally the types of mics I use for each drum:

Kick/Inside – AKG D112: I use the AKG D112 because it is a large diaphragm dynamic microphone that can handle the low end and high SPL of a kick drum whilst allowing for a clear attack and chunky mids. I also like the Sennheiser e602 or the Audix D6 for a more tailored sound. Nowadays I would say the D112 and the Shure Beta 52 are the most common mics that I see for this application. The D112 is known for being more on the 'thuddy' side whilst the Beta 52 has more of a tailored, 'pillowy' sound. Other notable mentions to handle this duty are the Audix D6, the Audio Technica AE2500, the Sennheiser E602, or the Beyerdynamic M88.

Kick/Outside – Rode NTK: Or any other quality large diaphragm tube condenser. This is a large diaphragm tube condenser from the Australian microphone company, Rode. I use this mic to give a little extra low end and a more air to the kick drum to round out the isolated and sterile sound of the inside mic. Most LDCs will handle this duty well because most of them can handle high SPL very well. The Neumann U47 was the professional standard for this application for many years and if you have one available, use it!

Snare/Top – Shure SM57 (or beta 57 for a brighter sound): An industry standard. If you're looking for more attack out of your snare but want to attain that naturally (without EQ), try the Shure Beta 57 because it has a more pronounced upper mid-range presence peak than the SM57. Dynamic mics are mostly selected for drums because they handle the transients better as a result of their rugged and slower moving elements and can also take high doses of sound pressure. There are millions of other options out there that would work just as well on snare drum but what I would suggest is start with the SM57 to gain a reference and move on from there as it has been used on countless albums and is tried and tested.

Snare/Bottom – (or beta 57 for a brighter sound): The main purpose of this mic is to pick up the sound of the strainer on the bottom of the snare drum. I like to use the same mic as the top as I find that when you flip the polarity, a more even sound is produced. This is a matter of taste so you may find you need either a brighter or a darker sound and will have to choose a mic accordingly that will provide that sound. This is one of those things that can not be taught and comes by experiencing different microphones.

NB. As far as the snare mics go, generally the bottom mic's polarity is flipped in relation to the top mic and is blended accordingly to mix in the sound of the strainers. This is because the bottom mic, being that it's pointed up at the bottom head underneath the snare drum, will naturally be opposite in polarity to the top mic. This occurs as a result of the diaphragm's movement being directly analogous to the movement of the drum skin when struck. When the top head is caused to depress by being struck by the drumstick, the bottom head reacts accordingly and is captured as such by the bottom microphone. The only difference is that the bottom microphone is capturing the opposite motion of the bottom head in relation to the top head, i.e. the opposite phase of the pressure wave created by the striking the top head. This will usually cause certan phase cancellations when combined with the top mic so initating a polarity reversal by way of your mixers' channel polarity switch will bring both signals back into phase. However, this does not mean that it will sound best for the material so check to see if the polarity reversal will supply the sound you are looking for. Once again, let your ears decide.

Hi Hat – AKG c418 mini-condensor: This is a mini condenser that was originally created by AKG for percussion instruments. Being as such, the microphones' response is bright with a presence peak in the high upper midrange and a gentle bass roll-off starting at around 500Hz at a 6dB/oct slope. This, to me, makes it a perfect candidate for miking hi hat as it's nice and bright with an unobtrusive midrange and a nicely rolled off low-end. This supplies me with a good sound before it hits the tape that never needs to be EQed. Alternatively, any other decent small diaphragm condenser will work.

Toms – Sennheiser MKII 421: These microphones have been the industry standard for miking toms for decades although any good quality cardioid dynamic microphone will work as well. SM57s will handle these duties just fine and so will some diaphragm condensers (some engineers use U87's!), which is fine, but I prefer dynamic mics for rock/pop because they are more directional and therefore tamer when it comes to cymbal bleed. What makes the 421's so nice is that they are clean, punchy, and focused while supplying a great deal of rejection from the outward extremes which makes them perfect in the battle of minimizing signal bleed when recording drums. This is because at the upper frequencies the polar response of the microphone becomes narrower so cymbal bleed is minimized while the on-axis sound, i.e. the tom, is maximized.

Overheads – Neumann TLM 103 matched pair: I prefer large diaphragm condensers for the overheads but any matched pair of small diaphragm condensers (as best quality as possible, of course) will work fine as well, especially if you're using an X/Y configuration, which is the method of aligning the capsules of two microphones vertically at around 90 degrees. This results in a focused stereo image that collapses well into mono. To me, the overheads are the most important mics in the drum kit miking process, so careful attention must be paid to their position and the quality of their reproduction.

A quick note on overhead mics is that, for me, this is where the sound of the drumset starts. Early on, it was a major revelation to me when I realized that the overheads are not there to merely capture the cymbals. In fact, I would say that starting with a great overhead sound that captures a balanced image of the entire kit (as always, to match your production goals) is probably the best approach. This forces you to, firstly, get the drums themselves sounding as best as possible and, secondly, to get a good sound with just two mics. Finally, when it comes to choosing overhead microphones, the options are myriad and varied. Would I recommend dynamic mics as overheads or even a single mic, i,e, a mono overhead? Sure, if it fits the vision of the production. I do think it's safe to say that the options are myriad and varied. In my experience, the most common choice of microphone when it comes to overhead miking is the condenser or capacitor variety followed in close second by ribbon mics. Notable mentions for overhead applications are: AKG C12V, AKG 414, Royer Ribbons, Neumann U67/87, RCA 77 and many others, including worry free stereo microphones.

The initial placement of the microphones all comes down to knowing the polar responses of your microphones and placing them in order to capture what you want and to reject what you don't. This can be a tricky exercise, especially when dealing with a drummer who sets his kit up very tightly so getting a mic in there proves to be almost impossible. In my opinion, an experienced studio drummer should know the basics of studio recording and should be able to adapt to setting his kit up to accommodate microphones. All drummers should know that failure to compromise means a compromise in sound quality if the engineer can not capture what is necessary. So, after you get the guy to move his splash over a few inches, the tracking can begin.

Tracking

Tracking, to me, is *the* most important stage in the recording process and requires the majority of your attention. I can not stress this point enough. *The only secret to great recordings is that **your sound is only as good as your source**.* A well overused cliché, I know, but there is a reason for that and a reason why it is the golden rule of recording. True, when you listen to your favorite album there is a good chance that you will be listening to something that has a fair amount of processing and enhancement. Sometimes it sounds that way but in reality it was really just a well crafted recording. Be that as it may, the tools we have to manipulate audio have purpose and would not be here if we didn't need them. This is for two main reasons:

1. To correct an inconstancy
2. To supply an effect

What my experience has taught me is that the better the quality of firstly the performance and secondly of the recording during the tracking phase, the less of a degree of processing is needed and the easier it is to reach excellent results. An even, consistent drummer is a lifesaver to a mix engineer and a pleasure for workload, not to mention the mix itself. It is just another aspect of the *'everything counts'* ideal that needs to be great in order to sound great.

The next aspect of a great drum sound is ambience, as touched on previously. Back in days of old when major commercial studios were the only gateways to professional audio a lot of attention was paid to the construction of the performance rooms, or stages, as they were called, where the acoustics and therefore the ambience was finely tuned to supply a luscious and natural reverberation to recordings. Great sounding ambience can add immense depth and space to a drum kit and is almost essential if not because of the advent of digital reverb and now high precision convolution reverbs which make it possible to record drums in a relatively 'dead' environment and add convincing ambience later.

A major consideration here is that the shape of a room can highly influence the response and quality of a recording, even whilst being 'dead'. If you are recording in a room with parallel walls the room will have certain frequency resonances called *standing waves* that will accentuate certain frequencies in an unnatural way at certain parts of the room called *nodes*. Because of the average dimensions required to accomplish a simple recording, these frequencies are often most prominent in the low or bass end of the spectrum and will cause the recording to become 'boomy' or 'muddy'. Simple alterations to room shape by way of corner functional bass traps, baffles and diffusers can greatly relieve this problem.

Tracking Levels

The first thing I would say when tracking drums is to track them dry, without EQ (unless you know *exactly* what you're going for), and keep your levels relatively low (NB. good mics and good mic pres are a must here. I can't stress the front end enough, especially if you're working in a DAW). Peaks topping out at -12dB should do the trick. If you think it's too soft, simply turn up your control room volume. Modern analogue to digital converters may seem relatively the same but they're not and some perform better than others. It's also somewhat true that low to mid level converters can perform better at higher sample rates. The old adage of 'keep your levels as hot as you can before clipping' is now advice from a bygone era and should be avoided completely.

Back when 16-bit audio and therefore narrower dynamic range was the norm, getting the most out of your bits by way of maximized levels was preached as gospel. However, with the advent of 24bit audio it is no longer required (or advisable) to drive your levels anywhere near the 0dBfs (digital clipping) region. The reasons for this are simple. The first is that it's not necessary, so why do it? In the 24-bit domain you have a theoretical dynamic range of 144dB (closer to 110dB in reality) and that's more than enough considering the signal to noise ratio of even the quietest mic pres don't usually exceed the 110dB mark.
The second is that as the voltage reaching your AD converter rises it has a harder and harder time tracking it because of a phenomenon called "inter-sample peaks", so the closer it approaches 0dBfs, the more it misrepresents and distorts the audio. The main advantage of keeping your levels low is that you preserve the full dynamic range of the recording whilst making sure your converters are operating at their optimal level. If there is no obvious noise at these levels, then there's no problem.

One more handy thing to know when it comes to levels in general is that on most AD/DA converters, 0VU (Volume Units, the industry standard metering scale for decades) is -18dBfs. This is not to be confused with the dBu or dBv scale found on most analog equipment, such as mixing consoles. Just so you can correlate how these levels relate, this is the breakdown:

0VU = -18dBfs =+4dBu

These are generally regarded as standard nominal levels in a professional audio system. This means that when the input level of your mixer's microphone preamp is hitting +4dBu on the output PPM meter, it should be seen as -18dBfs on your AD converter's input meter[1], and 0VU on a VU meter. Note that an average mixer's meter tops out at anywhere from +18dBu anywhere up to +30dBu. This is an indication of where the mixer's internal circuitry will clip, or in laymen's terms, distort. So, if you subtract +4dBu from, say, +18dBu, the resulting number – in this case 14dB – is the amount of headroom you have to work with. Therefore, if your highest peaks are hitting -18dBfs at your AD converter, you have 14dB of headroom before the audio will experience clipping.

Simple, right?

Internal Levels

I don't recommend going above -6dBfs on ANY channel at any time. It's interesting to note that after talking to a plug-in engineer I was informed that even the best plug-ins perform poorly above the -6dBfs mark because their internal calculation becomes unstable, again related to inter-sample peak distortion.

[1] Depending on the calibration of your AD converter, 0VU can be seen by the input meter as anywhere from -12dBfs to -22dBfs but the general value is regarded as -18dBfs.

Keeping your levels conservative ITB (In The Box, i.e. inside the DAW software) is the best approach when mixing, and processing should be auditioned in relation to the original source at a nominal level, i.e. the same level. Sometimes it's easy to think you've done something really great to a track when in actual fact you've only made it louder and therefore appealed to the ears' equal loudness curves. Reference is the key word here and all processing you attempt should always be in relation to the original sound. In other words, when you affect a sound with a processor, you should always A/B the processed signal at *the same perceived volume.* How else are you going to know if what you've just done *actually* sounds better? Louder is definitely *not* better so trust your ears or use a trusted RMS meter to match the average levels. Both used together can be very powerful tools in processing and correcting inconsistencies.

Bussing and Panorama

This is a typical track list for a drum kit as alluded to in the section on microphones:

1. Kick Inside
2. Kick Outside
3. Snare Top
4. Snare Bottom
5. Hi Hat
6. Tom 1
7. Tom 2
8. Tom 3
9. Overhead L
10. Overhead R

I've left the ambience tracks out purposefully as that will be explained in detail at the end of the guide.

Now, there are many ways that you could possibly bus and pan these tracks and it is of my opinion that, while taste has a lot to do with it, the bussing and panorama that you choose should first serve the song, and second serve the processing that you decide is required to serve the song. Since we're discussing modern pop/rock production, the following are a few of my basic bussing and panorama techniques for that particular genre.

The first thing I would do is bus the two kick channels to a mono group, unifying them into one channel and panning it dead center. I would do the same to the snare. Doing this gives you control in balancing the kick and snare as whole instruments and whilst also allowing you to control the balance of the two kick and snare tracks (in and out, top and bottom, respectively) in relation to one another. Bussing them this way also leaves you the option of inserting processors such as compressors and EQ's to affect the kick or snare as a whole instead of noodling around with each individual track. The same applies to fact that you can send the unified snare signal down one FX send instead of having to send from two separate channels. Bussing them both center serves to even out the low end in the stereo field called “the phantom image” between the two speakers, maintaining punch and clarity. Some guys pan the snare off to the left or right a bit, as it appears in the overheads. This a matter of taste and is ultimately the decision of you, the engineer and the producer.

Sidenote Here it is important to discuss the relationship between the overheads and the snare. The simplest way I’ve found to get a coherent stereo image in your overheads whilst keeping the snare focused at the center is to use an X/Y configuration (a small diaphragm condenser pair works best because you can really get the capsules clost to one another) whereby the two capsules, almost touching, are aligned at 90 degrees to one another and aimed above the drum kit. Because every scenario is different, where it’s placed is up to you but generally 3 to 6 feet above the drum kit is sufficient. Your ears will tell you when it’s right. This allows for the sound coming from the snare (and everything else, for that matter) to reach the mics at relatively the same time. This can really help with clarity and focus in the mixing process, especially for something as important as the snare. As far as the spaced pair goes, the main keyword here is ‘equidistant’ and both overheads should be equal distance away from the snare drum. If you tape a piece of string to the center of the snare drum you can mark the required distance, eg. the distance to the left overhead, and apply the radius to the right overhead. This will give you phase coherence in the overheads in relation to the snare but will not take care of any phase issues in relation to each other so it’s a good idea to follow the “3:1 rule”. This is a rule that states that in order to avoid phasing issues between two microphones, they should be placed three times the distance apart than they are from the source.

Right, moving on.

The next course of action would be to create another group, this time stereo, and bus the all tom tracks to it. The general consensus is that panning toms hard left or right is to be avoided due to the fact that it can make them sound unnaturally wide but despite that, this topic has still been debated endlessly. The truth is that it’s a matter of taste and what fits subjectively into the material. The LCR [2](Left, Center, Right) technique, for one, embraces placing the toms at either the extremes or all dead center, and many engineers employ this philosophy. Accordingly, it has also been supposed that the toms can also be placed on the stereo field by their position in the overheads. In other words, you would first listen to the stereo overheads and pin point the position of the toms within the

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[2] LCR (Left, Center, Right) is a mix philosophy that follows three panorama positions: L, C, and R. The reasoning behind this is that the ear has a difficult time distinguishing much between those extremes and thus may be avoided. It also stresses proper stereo miking techniques and the placement of mono sounds either at the left or right extremes or in the center. The claimed result is a clear and more defined mid sound contrasting with carefully placed side sounds. Better ono compatibility is also said to be achieved.

stereo field, and then attempt to match that by panning the tom tracks to those positions.

At this point it is helpful to note two mixing concepts related directly to drums whereby a “perspective” is chosen in the approach to panorama. These concepts are called “drummer perspective”, or “audience perspective”. All this basically means is that, in the case of drummer perspective, the elements of the drum kit are placed in the stereo field as you would hear them sitting behind the drum set. For example, high hat to the left, overheads stereo flipped, and toms panned from left to right. Obviously, in audience perspective the opposite would be achieved by placing the high hat to the right, toms panned right to left, and so on. For a right handed drummer, that is. Throw a left hander in the mix and everything flips once again! Of course, these are two inter-related concepts that might not work for every scenario as you may only require mono overheads or mono drums altogether but for a general scenario, it’s safe to say you would be alright by choosing any one of the two techniques described above. As always, it ultimately boils down to a matter of taste.

Of course, the overheads get their own stereo group as well. I tend to keep my overheads hard left, hard right these days otherwise it’s just not stereo miking, is it? I’ve heard a lot about panning the overheads in; say 80L - 80R to make them ‘narrower’, but I personally see no advantage in this technique. Sure, the correlation meter may tell you that they’re more phase coherent but what you’re actually doing is merely minimizing the effects, or should I say the *minimizing the perceived difference in phase cancellation* when you switch to mono. This is for the obvious reason that you’re bringing both the L and R signals closer to the center, giving the illusion of better mono compatibility. A good stereo recording should be phase coherent to begin with so it is my inclination to always leave stereo signals where they should be: hard left and right.

Now that you have all your main components of a drum mix bussed the final step is to create a master group to handle overall drum processing. I generally bus the hi-hat to this group, panned off to the right somewhere percent. It is also not uncommon for me to pan the high hat hard right, depending on the circumstance.

So, this is how we’ve bussed everything so far:

1. Kick inside, Kick outside → Kick group (mono)
2. Snare Top, Snare Bottom → Snare group (mono)
3. Hi Hat → Master Drums Group (stereo)
4. Tom 1, Tom 2, Tom 3 → Toms group (stereo)
5. Overhead L, Overhead R → Overheads group (stereo)

6. All Drums → Master Drum Group

All the above groups should now be bussed to a stereo master drums group.

Now you have control over the individual track blend, i.e. snare top vs. snare bottom; control over the sub groups, i.e. stereo toms, overheads, etc; and control over the entire drum kit. Your drum tracks are now bussed and panned and ready for processing.

Processing

The main goal when mixing drums in pop/rock is usually to create a larger than life drum sound that is clear and defined whilst retaining warmth and punch. True, much of the end result is dependant on the performance of the drummer but there are many techniques you can employ to the individual tracks, the sub-groups, and finally the master group to sculpt and glue the sounds together into one powerful unit.

EQ

Equalization is most definitely the first tool engineers reach for when attempting to correct inconsistencies in the spectrum of a recording, and it is the oldest form of signal processing we have. For this reason its methods of use are myriad and varied, opening the concept up to extreme speculation and misinterpretation.

The main purpose of EQ is to out even (equalize) any spectral inconsistencies that may have made its way into a recording as a result of any number of factors including a troublesome room, a bad instrument, mic response, frequency build up, and so on. It is also used as a tone shaping tool to gently boost or cut certain areas to achieve an overall "sound". I have two words that I use to describe these two processes, the former being "surgical", and the latter being "topical". I generally apply surgical EQ at the track level and topical, tone-shaping EQ at the group and master bus level, although there are exceptions, such as mono tracks that are bussed straight to the master bus and are not grouped.

In an ideal world every recording would be perfect and require no EQ. Indeed, this is what we should be striving for when recording and why the idea of getting the best possible sound you can at the source is the number one amendment to the golden rule.

If you think about it, microphones, its position, the room, and even the instrument itself are forms of EQ. Each microphone has its own specific frequency response. Where you place it affects that response. How you play and setup the instrument (drum heads, beater material, dampening, etc) affects the resulting tone. And each room sounds different and emphasizes different parts of the spectrum, further affecting the recorded sound. Cleverly utilizing these aspects during the recording process will achieve better sounding results, *naturally,* than even the best EQ. That being said, it is still common, especially in modern pop/rock, to hear quite extreme EQ to tailor the sound. I am certainly not against it if that's what it takes to achieve your desired results. Heavy metal would certainly not be the same without it. However, I would still always stress to not EQ unnecessarily and to aim to achieve the desired sound from the onset.

When it comes to drums, I always start as a reference for EQ with the overheads. Many times EQ is not required so I do not use any because I use my best mics for the job in a well thought out position. If EQ is required I will generally reach for a UAD Pultec[3], inserted into the overheads group bus. I will then pay attention to the frequency balance between the kick drum and the rest of the kit and trim any low end boom that might be a nuisance. I will also pay attention to the upper mid range to make sure that it's not strident. The Pultec is great for this because you can make quite heavy-handed adjustments without affecting the sound negatively. It also has a few tricks to it whereby you can smooth out these areas by cutting and boosting within the same band. With the absence of an EQ plot graph, you'll have to use your ears!

Once I've got the overheads sounding how I want them to, I'll add the kick and blend it to taste. In modern pop/rock the kick drum is generally quite loud, punchy and present. This may be how your original recording already sounds when you bring up the fader so leaving it alone in that circumstance is the best option. However, typically a generous helping of EQ is often employed to achieve the desired tone. In that course, attention is given to the attack portion of the spectrum from 2.5 kHz up to about 8kHz, the lower mid-rage boxy zones from approx. 150 Hz to 1000 Hz, and the low end from 150 Hz downwards. These figures are always approximates because all processing decisions are subjective to the source. I generally find myself boosting the attack quite considerably (+6 dB's or more) and cutting the lower mid range to "scoop" out troublesome and boxy sounding mids. I generally leave the low end how it is at this point because later on it will have to be auditioned in relation to the bass guitar for the obvious reason that they occupy similar frequency ranges.

Next I'll add the snare drum to the mix and balance it with the rest of the kit until it blends the way I want it to. I personally like a really upfront kick and snare relationship. It really bothers me if I can't hear the snare drum at any point in the song so I take great effort to make sure the snare is tight, punchy and crisp with a well tailored attack. Generally I'll add some air at about 10kHz and attenuate any troublesome resonances and inconsistencies in the lower mid-range. In the past I would also boost the attack in the 2.5kHz region but these days I rather employ a plugin by Steinberg called the Envelope Shaper (originally created by SPL as the *transient designer*) to boost the attack portion of the waveform. I find its effects much smoother and more natural than excessive EQ and it keeps the transient from being buried when applying compression.

[3] The UAD Pultec is a great sounding plugin based on the Pultec EQP1a, a classic tube shelving EQ from the 50's. It has been used on countless recordings and continues to be used extensively. It is known for it's smooth, transparent sound and originals specimens are highly expensive and sought after.

Next I'll bring up the toms and examine each track for bleed because any change in EQ you make will affect the bleed as well. If there is excessive low end rumble, which is common on tom tracks, I will trim it out with a gradually sloped HPF[4] in the sub regions of the 1st and 2nd toms. The floor tom usually benefits from a full low end so I'll make a decision as to whether it will be detrimental to the sound to trim the rumble out and whether I can live with it. Tom mics often benefit from gentle mid range attenuation to filter out the ringing frequencies and to emphasize their fullness and attack. Once again, it is quite possible to achieve this in the recording phase using the console EQ so this is a decision you must take based on the material at hand.

Generally I am happy to use the track EQ's within my daw for individual track EQ but there are a number of plugins, commercial and free, that I employ to get the job done as well. These include the Waves Renaissance series, Bootsy's Nasty series (freeware), UAD Cambridge, UAD Neve bundle, Antress Modern series (freeware), and an EQ called Electri-Q by AiXcoustic Creations . There are a million other options from companies such as PSP, TC Powercore, URS, McDSP, Stillwell, DDMF, and so on, but when it comes to plugins, it is better to use a few good specimens sparingly than to plaster them all over every track.

Finally, for the sake of covering all the bases, here is a rough outline of frequencies related to drums:

1. **Kick –** HPF, 30 – 80Hz; fullness, 60Hz – 120Hz; boxiness, 300 – 600 Hz; attack, 2.5 – 5kHz; air, 8kHz +.
2. **Snare –** HPFLS, 75 – 100Hz (24dB/Oct or more if possible), fullness, 100 – 240 Hz; boxiness, 400 – 1000Hz; attack, 2.5 – 5kHz; air, 12kHz +
3. **Hi hat –** HPF, 200 – 600 Hz, harshness/nasal 1.6 – 2.4 kHz; sheen 6kHz+
4. **Toms –** HPF, 60 – 100 Hz; fullness 80 – 120 Hz; boxiness, 300 – 600 Hz; attack 5 – 8kHz
5. **Overheads –** HPF 30 – 200 Hz (depends); nasalness, 1.2 – 2.2kHz; sheen, 6kHz +; air, 12kHz +

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[4] The use of the high pass filter is a highly debated topic among professionals, the main argument being that if the track was recorded properly, the low end should not need such indiscriminate low end hacking. That is not to say that the high pass filter does not have its place, but rather that more restraint is to be employed when using one. My opinion is somewhere in the middle where I will usually grab for the low shelf, instead to gently attenuate the troublesome region, but leaving the very low part of the spectrum intact. A HPF can be helpful, however, when there is just too much sub-bass or rumble and the track will not be affected negatively by the use of one.

Compression

The real helper in processing drums for rock/pop is compression and is my one of my favorite processors. There's no doubt that most of the music we hear today would be hard pressed to succeed if it wasn't for this great device that is basically an automatic fader designed to control dynamic range and in some cases inject a certain tonal aesthetic and energy. The basic function of a compressor is outlined as a device that raises the average level of a given source but I have come to feel that a compressor is so much more. Not only does it raise the average level but it can be used as a tone shaping tool and to either suppress or enhance the transients of a waveform to bring out certain characteristics that weren't as present in the original. The attack and release sections of a compressor are important parameters that, if used correctly, can become your best friends in the fight against boring, lifeless drums. I will go into some settings in more detail as we work our way through this part of the tutorial.

Compressing Drums

Compression can be a nebulous beast compounded not only by the fact that all compressors are not created equal, but by the shear variety of their design and application. For me, there are four main reasons to reach for a compressor:

- To supply an effect
- To control dynamic range
- To suppress or enhance transient information
- For upwards compression in parallel

The Effect

Supplying an effect with compression is completely related to how a particular compressor sounds. You just can not achieve the same results with one compressor that you did with another. For this reason there are certain compressors that are well suited for this task and some that aren't. Most aren't. When it comes to drums, the effect is usually a well tailored attack and release to achieve a pumping sound to the drums and a crisp snap, all executed in a musical manner. Three notable mentions in this category are the DBX 160, the Empirical Labs Distressor, and the Universal Audio 1176 LN. All three of these compressors are known for their very musical sounding compression and certain mojo that they inject into the material. It is for this reason that you will often find me using the UAD 1176 plugin to smash rock drums into submission.

But inserted where? And how much gain reduction?

As time has gone on, I have found myself moving away from using compression at the track level for compression only at the master drums bus. I tend to mix "into" the compressor to supply the energy I am looking for while listening for the best gain reduction range for the desired sound as I'm adjusting my balance at the track level. This serves to not only supply a pumping, breathing effect, but also to smooth out the dynamic range in a musical way. It is important to note that the attack and release are quite important here for "sculpting" the transient to poke through for just long enough to be perceived as crisp and then the release set to engage in time with the material. I generally start with a medium attack (generally 10-35ms, but a lot faster on the 1176) and adjust the release until I get 60% recovery (back to 0dB) before the next transient. As with all audio equipment, there is a "sweet spot" operating range where the unit will sound it's best and all adjustments should be done with the ear, instead of by number.

Controlling Dynamic Range

There are times where you may encounter a drummer whose playing has a wide dynamic swing and requires attention. The caveat in a case like this is that compression is not in any way a remedy for this, the reason for reason being that while you may get each hit closer in amplitude (level) to one another, each hit will still sound different because the tonal variation of the drum is directly proportional to how hard or soft it's being played. However, if the desired change in dynamics is subtle and you can achieve it with a compressor, then that is completely acceptable to use one. Particular attention should be paid to the attack in this case and a compressor with more transparent characteristics should be chosen.

Transients

Transients are the first few milliseconds of the waveform and drums are full of them. In comparison with a flute, you will note that the flute's transient characteristic is slow and gradual, whilst a snare drum's will be short and sharp. It is in relation to these transients that a compressor works. Generally, a faster attack time will suppress the transient and a slower attack time will let more of the transient through. Subsequently, a fast release will let more decay of the note/instrument through, whilst a slow release will suppress it. It is by cleverly utilizing these two controls that you can achieve the desired relationship between transient information and sustain. Also please note that excessive [fast] compression can greatly affect the top end and it is how a particular compressor handles this problem that marks its quality.

Ratios and Settings

Generally the ratio is selected based on how much gain reduction you will need when the audio extends over a given threshold. Typically, higher ratios for drums are chosen from 4:1 all the way up to hard limiting, depending on the application.

For rock/pop drums I generally find myself using the UAD 1176 at an 8:1 ratio[5] with the attack set medium-slow and the release quite fast (50 – 150 ms). This is not set in stone, however, as I will adjust to the material for either less dramatic or more extreme settings, whatever the need may be. As far as other compressors go, the DBX160 or the Distressor will do fine as well, pending the exploration into their sounds. The DBX 160 is a VCA-based compressor and the Distressor is a digitally controlled compressor that employs a harmonic distortion component. Both have been used on many a hit record and will process drums in a very pleasing manner. As far as plugins go, UAD makes emulations of both of them and they sound very good.

EQ with Compression

Naturally, because a compressor increases the average level of a given source, it helps immensely to filter out the offending frequencies so that they are not amplified along with the frequencies that you want. Cleverly isolating and eliminating troublesome resonances and frequencies can aid in better sounding, smoother compression.

Ambience

Once I have got a good balance with my unprocessed drums, I focus on creating L and R ambience tracks that can be blended into the master drum group to broaden the spatial information of the drums and give them depth. It is also common to have an additional snare reverb but often room ambience is sufficient so it can be omitted.

There are many ways to add ambience to drum tracks. The first and most obvious is to record them in a good sounding room that has favorable acoustics and set up a stereo pair of room mics. Obviously this is a luxury for most so knowing how to emulate ambience artificially using plug-ins is essential.

In the interest of using only the highest quality artificial ambience, it is of my opinion that convolution reverbs that employ impulse responses are the best option for achieving this. I use SIR (Super Impulse Response processor) and Steinberg's Reverence, but there are many other (and expensive) alternatives out there such as Altiverb, Waves IR-1, Wizooverb, Nebula, etc. The thing I like about SIR is that it supports files up to 96kHz, is completely free, and is compatible with wave file impulses which means that if you're so inclined, you can make your own using Voxengo Deconvolver or a similar deconvolving utility.

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[5] The 1176 ratio buttons, when pushed in simultaneously, offer what is called the "all button mode". The result is a unique-sound that soon became widely copied in rock music mixes and has been imitated by other compressor manufacturers. The ratio goes to somewhere between 12:1 and 20:1, and the bias points change all over the circuit. As a result, the attack and release times change. This change in attack and release times creates a compression curve that results in an "overdriven" tone.

And it doesn't just stop with reverbs. There are many websites on the web where you can download all sorts of impulses made from all kinds of outboard units such as preamps, compressors, famous outboard reverb units and also impulses captured from real acoustic spaces.

The process of creating an ambience track is simple. What I do is I solo all the drums and insert an instance of SIR (you may use your favorite reverb plug-in) and set it to 100% wet. I usually turn the kick down quite a bit as I don't want it to cause the reverb to flutter and because I am aiming to recreate the sound of the drums in a real room as I would hear if it was actually miked up. In fact, you can attenuate whatever you like as now the channel faders essentially become FX sends. Just save your mixer level settings for the drum channels so you can go back to where they were when you're done.

Drums generally require short to medium room ambience so I scroll through my impulses/reverbs until I find one that I like. What's nice about SIR is that it has a powerful filter section that can be used to pre-EQ the reverb and eliminate mud and shape the reverb to taste. Once I've picked a reverb and set the filter I set my left and right locators to include the entire drum arrangement (be careful to leave a bit of time at the end to capture the reverb tails) and export/bounce the ambience down to split stereo mono tracks. I then bus those two tracks to their own group, which is routed to the master drum group track. You now have a stereo ambience that can either be mixed into the master drum group (for ambience blending via the compressors' release knob) or merely sent to the master bus as an open, unprocessed ambience. It is also common to compress room mics quite heavily and the UREI 1178 stereo compressor was a usual go-to compressor for this duty. Also, the SSL LMC-1 plugin was released for this purpose.   It is a digital emulation of an SSL reverse talkback microphone compressor that engineer Hugh Padgham found worked great on drum ambience mics during a recording with Phil Collins. It is now available in plugin format by SSL and it's free.

When using a dedicated reverb for snare, the mono snare sub-group can be fed via an FX send to an aux/FX channel where another instance of your favorite reverb plug-in is inserted. High quality reverbs are a must here as they too are not all created equal. The snare can be radically enhanced by a short roomverb, a gated reverb or anything else that tickles your fancy and serves the mix. The options really are endless and picking the right reverb for the song is just a matter of taste and experience. The more you get to know your available reverbs the more you'll know what reverb to use for what purpose.

Final Considerations

The techniques outlined above are only guidelines and demonstrate only a few of the countless techniques that are out there. These particular ones have worked for me and the clients that I most encounter and should be viewed as techniques to be used in a pop/rock production. I hope that some of you found helpful hints within this guide and that they will help you along your way. I believe that audio engineers should stick together, help one another and share experience so as to benefit the industry as a whole. Happy production.