Compiled By
-=The Exorcist=-

Welcome to Studio Buddy! This program was written to help home recording enthusiasts and studio junkies benefit from our years of experience in the trenches at some of America's top recording studios. The techniques described in Studio Buddy have been handed down through the ages, and are tried and true.

But as experience will teach you, no two studio situations are ever exactly alike. So view these techniques as rules of thumb, or starting points. You will quickly discover that like any recipe, our techniques are often best when seasoned to taste. Have a good session!

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Studio Buddy

How do I record a snare drum?

Recommended mics: Shure SM57, AKG 414, Sennheiser 421, Neumann KM 84, Neumann KM 184

For the snare drum, it's always a safe and highly effective choice to use the venerable Shure SM57. Bring it in from the audience side of the kit and give it a 45 to 60 degree angle with the capsule about an inch or two above the head. The farther away it is from the head, the roomier the sound, but the more potential you have for phase problems. The closer to the head you get, the more bottom end you'll get - it will give you that "goosh-y" sound. By the way, it's always a good idea to have the snare mic follow a line to the drummer's crotch - not that it's a particularly good sounding part of the anatomy, but because it's away from the hi-hat and any potential leakage problems.

Recommended eq for the snare is: +2@100Hz on the bottom if necessary; roll off 300 to 700Hz in the lower mids to eliminate the box-like sound; and +2 to +6 dbs @ 5, 8, or 10Khz to brighten up the top end. Tuning the snare is very important in getting the right sound. If you encounter undesirable ringing in the snare, try a small piece of gaffers' tape. You can also try taping a small piece of a feminine napkin to the outer edge of the top head to eliminate over ring.

Remember that a snare is full of transients, so keep your levels fairly low to avoid overloading your preamp, tape machine, or the tape itself. -2 or -3 VU or + 2 or +3 peak reading are typical levels.

Studio Buddy

How do I record tom toms?

Favorite mics: AKG 414, Sennheiser 421, most condenser mics work well.

If the mic has a pad switch, use it. It's always better to pad at the mic, rather than the console. If the mic doesn't have a pad switch, but you're seeing too much level coming into the console or hearing any break-up/distortion, try padding the mic input at the console.

Mic all three toms with the mics set at a 45 degree (or thereabouts) angle to the drum head with the end of the mic (the capsule end) pointing at an imaginary spot about 2" past the rim nearest you as you place the mic (this is assuming you're working from the audience side of the kit). The floor tom mic can be placed a little closer to the center of the head, but not too close. The distance of the mic from the actual head should range between one inch and six inches depending on how "roomy" you like your drums to sound. The further the mics are from the drums, the roomier the sound, but you'll have to pay more attention to possible phase cancellation problems.

Eq.: +2@100Hz, -4@300 to 700Hz, +2@5K or above.

Tips: Dampen the drums to reduce ringing using a little bit of gaffers' tape or tape a piece of feminine napkin to the outer edge of the drum head using gaffers' tape. Generally speaking, the more mid-range you roll out of the toms, the better they will sound, to a point. You can roll out too much, and the result will sound hollow and box-like.

Experienced engineers concerned with saving tracks will often combine the stereo overheads/cymbals with tom-toms, using just two tracks panned far left and far right for all the toms and cymbals.

Remember that a tom-tom is full of transients, so keep your levels fairly low to avoid overloading your preamp, tape machine or the tape itself. -2 or -3 VU or + 2 or +3 peak reading are typical levels.

Studio Buddy

How do I record a kick drum or bass drum?

Favorite mics: Sennheiser 421, AKG D-12 or D-112

If the mic you're using has a pad switch, use it. If not, pad the input at the console. Mic the kick drum from the audience side, but only after throwing a sandbag in the drum to weigh it down. Let the sandbag touch the head (that the beater hits) just enough to dampen out any obnoxious overtones, but not the good, natural sounding ones. The mic should be placed about half way in to the drum itself and pointing at the beater. If you bring the mic in from the right side of the drum and angle it at the beater you will be avoiding leakage from the snare drum, which is a good thing to do. You can experiment with the depth of the mic, but always keep the mic pointed at the beater for maximum attack. If you want a "poofier" kick sound, you can point the mic away from the beater, but again, try to avoid letting it point in the direction of the snare to minimize leakage. If you want a roomier sound, you can pull the mic out of the drum a little bit. The further out you pull it, the roomier it will get. Some engineers use a second mic a foot or two outside the kick. Be sure to check the phase relationship of the two kick mics if you try that technique. If you have phase cancellation problems, they will usually manifest themselves by canceling out the bottom end of the kick.

Eq: If you need more bottom end, try boosting @ 60 or 100Hz. Try rolling off lower mids (300-700Hz) to get rid of a box-like sound. To add more attack, try boosting in the 1K to 3K range.

Remember that a kick drum is full of transients, so keep your levels fairly low to avoid overloading your preamp, tape machine or the tape itself. -2 or -3 VU or + 2 or +3 peak reading are typical levels.

Tips: If you don't have gobos to block incoming and outgoing leakage, try placing a moving blanket in a tent-like fashion around the mic stand and kick drum opening. Tune the kick drum up or down according to the key the song is in, making sure that the tuning works well with the register the bass guitar is in.

Studio Buddy

How do I record a bass guitar?

Using a direct box is the most common way to record a bass. Coming out of the direct box into the console's preamp or an outboard preamp will give you the ability to get the appropriate amount of gain. For a cleaner, more direct route, try going into, then out of an outboard preamp, directly into your tape machine input on the appropriate track.

Most engineers use a limiter/compressor on the bass. It gives the bass a fatter sound by controlling peaks so that all the notes coming out of the bass have roughly the same level. Many engineers prefer to use a tube limiter such as a UREI LA-2A to get the fattest and warmest sound possible. It's an expensive piece of gear, but it works great. Most engineers set the threshold and ratio knobs so that the bass signal is always getting "squashed" by 2 to 5 dbs. A typical "ratio" setting is 3:1, meaning that for every 3 db of peak signal over the threshold, the limiter will only output 1 db.

For a more punchy bass sound, set the threshold so that the signal is getting hammered, and when soloed, it sounds obviously squashed. When in the context of the entire track, the squashed sound will tend to be less obvious, while making the bass much more apparent.

Other limiters that are often used include the UREI 1176 (also not cheap), and the DBX 160 (fairly inexpensive). The 1176 is famous for its wide range of control on the attack and release, as well as its "classic" sound. The DBX 160 is a favorite of engineers looking to get a snappy, poppy bass sound often used on dance records.

The DBX 160 X (notice the X) is also a good inexpensive limiter for bass recording. It combines the range of controls of the 1176 with the fast attack and release times of the DBX 160.

EQing a bass for recording is usually pretty straight ahead. Add a little bit @ 100Hz to make the bottom fatter. Try 60Hz if you want to go even lower and fatter, although most car radios won't do a great job of reproducing 60Hz. If you're recording in a digital environment, it's always a plus to use a tube equalizer such as a Pultec to warm up the sound. The natural distortion caused by the tubes tends to add desirable harmonics to the bass signal.

To get more "bite" from your bass, try adding a couple of dbs @ 2.5Khz. When recording a bass, it's always good to be aware of the octave that the part is being played in. The octave may dictate where your most effective eq points are.

Some people prefer to record a mixed signal that comes from a direct signal as well as using the sound coming from a bass amplifier. A more advanced engineer might typically combine or mult the signals to just one track of tape. For a less experienced engineer, it might be a good idea to record the two signals to two distinct tracks, then combine them at a latter time. It's extremely important to remember that when recording the same signal from two sources that you are likely to encounter phase anomalies, meaning that the two signals will arrive at slightly different times. The result will often be comb filtering which will make some frequencies less audible the others. The bottom end is usually the first thing to disappear. This can be fixed by engaging the phase switch on the console, moving the mic closer to the bass amp's speaker, or using a very short delay on the direct signal so that it hits the console or tape machine at the same time as the later, amp/mic signal. This is a pretty tricky endeavor, and not recommended for novices.

When using a mic on a bass cabinet, it is usually desirable to try a condenser mic that is well-known for its bottom end, and to place it a foot or two back from the amp's grill. The reason; bass notes have long waveform, and require some air to fully manifest themselves. Rule of thumb, the closer your mic is to the amp, the more attack and edge you will hear. Farther away will give you more bottom end.

Studio Buddy

How do I record cymbals using overheads?

Favorite mics: Neumann U-87, AKG 414, Neumann KM 84, Shure SM81

Recording overheads can be remarkably simple or incredibly difficult. The important rule of thumb is to watch out for phase cancellation. The overhead mics will often interact with individual drum mics causing phase anomalies which manifest themselves as dropouts at certain frequencies in any or all of the drums.

If your drum sounds get "cardboard-y" after you bring up the faders on your overheads, you probably need to adjust the overhead mics.

Typically, engineers will record the overhead mics in stereo, making sure to bus or assign the overhead on the left side of the kit to a tape track that will correspond with the other drums on that side of the kit (e.g., the right overhead should ultimately end up being panned to the same side as the floor tom).

Some engineers will start with the two mics about 16 inches over the cymbals, and point them straight down, looking directly at the center of the cymbals to achieve a more bell-like sound. Other engineers prefer to angle the mics toward the outer edges of the cymbals to get a brighter, wispier sound.

The closer the mics are to the cymbals in either case, the less chance you will have of experiencing phase problems. If you do experience phase problems, it's often fixable by simply flicking the phase switch on one of the mic inputs or the other. A good rule of thumb is to always make sure that the distance between the two mics is at least twice as far as the distance between each mic and the cymbal it's over.

To get a roomier or bigger drum sound, just raise the mics higher - try moving them six inches at a time. But remember, as you get farther from the cymbals, you increase your chances of phase problems.

If the drummer you are recording really whacks their cymbals, you may need to pad your mics or your mic inputs. If the mics you are using have a roll-off switch, then use it. Good chance you won't need the low end frequencies that the roll-off will eliminate. If your mics don't have a roll-off, you can use the high-pass filter on the mic inputs of the console to do the same job. If you don't have roll-off or high-pass capability, then roll-off 10 or 12 dbs @30 or 60Hz using your equalizer. Generally speaking, cymbal mic require very little, if any eq on the top end. If you feel that you cymbals are dull, and need to be brightened up a touch, try a smidge @ 8 or 10Khz. Be careful! A little bit can go a long way.

A little strip of strategically placed gaffers' tape can eliminate the nasty overring that some cymbals have. It can also mellow out an overly bright cymbal.

Experienced engineers concerned with saving tracks will often combine the stereo overheads/cymbals with tom-toms, using just two tracks panned far left and far right for all the toms and cymbals.

Remember that cymbals are loaded with transients, so keep your levels fairly low to avoid overloading your preamp, tape machine or the tape itself. -4 or -5 VU or +1 or +2 peak reading are typical levels.

Studio Buddy

How do I record a hi-hat?

Recommended mics: Sony ECM 50, Shure SM57, Sennheiser 421, Shure SM81

Many engineers find that they often don't need a separate mic on the hi-hat because they get enough of it leaking in to the other drum mics. If you do need to mic a hi-hat, generally, it's a good idea to place the mic about an inch away from the outer edge of the hi-hat. It's also a good practice to angle the mic away from the kit so that the hi-hat mic doesn't "hear" too much of the snare or other drums. Because you don't want any extraneous low-end rumble from the rest of the drum kit, it's recommended that you use the mic's roll-off switch if it has one, or use the console's high-pass filter or equalizer to eliminate low-end information from cluttering up your hi-hat signal. Remember that a hi-hat is full of transients, so keep your levels fairly low to avoid overloading your preamp, tape machine or the tape itself.

Studio Buddy

How do I record an acoustic guitar?}

Recommended mics: AKG 414, Neumann U87, Neumann KM 84, Neumann KM 184, AKG C3000, AKG C1000, Sony ECM 50

While the acoustic guitar remains one of the most simple instruments by design, it also remains one of the hardest to get a great sound on in the studio. It's really not brain surgery, but knowing some of the basic laws of physics doesn't hurt. Unfortunately, I skipped school that day and didn't learn my physics, so I had to learn how to get a great acoustic guitar sound one mistake at a time. After making those mistakes, I sat down and formulated these laws which are considered to be the Ten Commandments of recording the acoustic guitar (by me anyway).

For the sake of argument I'm going to assume that if you're reading this, you own a 4 track, or an 8 track recorder, a fairly small console, some basic outboard equipment, and you don't own any $2,000 microphones. If you own a 13 foot long console and a 48 track digital machine, you can skip this because you probably know what I'm about to tell you.

Rule #1 A condenser mic will almost always sound better than a dynamic mic for acoustic guitars. There are several condenser mics that are currently on the market in the $350 price range that sound great on acoustics.

Rule #2 New strings will always sound better for recording than old.

Rule #3 Skinny strings sound brighter than fat ones (can you believe I get paid to write crap like this?!)

Rule #4 The sound you get has a great deal to do with the dynamics of the player.

Rule #5 Get down on your knees and position your ear as if it were the microphone while somebody else is playing the guitar. Move your ear around to find "sweet spots". You'll learn more from that, than you will by reading this. Don't try it with an electric guitar!

Rule #6 If you have somebody that is assisting you on the session, have them move the mic around what you think will be the sweet spot while the player is practicing the part he or she is about to lay down. Have your assistant wear headphones so you can communicate with him or her while the moving of the mic is taking place.

Rule #7 A limiter/compressor will almost always help you get a better sound.

Rule #8 Don't believe everything you read. I only have seven commandments, not ten.

Let's get right to it. If the sound you want to get is a country/pop, strummed sound similar to the Eagles "Lyin' Eyes", here's the formula: Place the microphone about 6 to 8 inches from the guitar's sound hole, but angle the mic toward the area where the fretboard and the sound hole meet. If you point the mic directly into the sound hole, it will be very full - probably much too full, and very boomy. Use a compressor/limiter to knock down any peaks (3:1 ratio), and set the threshold a little lower to give it a slightly "squashed" or tighter sound. Set the threshold higher to just limit the peaks and give a more open sound. You may need to eq out some boominess. If so, try rolling off some bottom (100Hz), or cutting a couple of db at 300Hz. To add some "silk" on the top end, try something in the 8-10K range, but be careful, too much will add noise to the track. Positioning the mic so it angles toward the pick will give more attack - less sweetness.

For that John Mellencamp sound, try medium gauge strings, a little more compression, and try adding a little eq around the mids - lets say 700Hz to 1.2KHz. That will give you a sound that is a little more "woodsy" (a highly technical term).

Melissa Etheridge? Try this on for size. Use a guitar with a built-in pick up and a microphone to boot. You will undoubtedly get some phase anomalies, but that's part of the sound. Experiment with moving the mic closer and farther. That will affect the phase relationship of the two sound sources. Sooner or later, you'll hit on something that will put a smile on your face. You can pan the two signals left and right to get a broad stereo sound, but make sure that if you check the sound in mono, that the signal remains fairly well intact.

Gut string or classical guitar? Piece of cake. Once again, use a condenser mic, but place it about ten inches away from the guitar. As a matter of fact, try placing it about 3 to 4 inches up the neck, but aim it at the players picking fingers. This angle will reduce boominess by virtue of the mic's cardioid polar pattern producing a natural roll off when it's aimed off-axis, while simultaneously delivering the attack of the fingers. Try and say that three times in a row! The added distance will pick up some of the guitar body's resonance. A compressor/limiter is a must for this case because of unexpected peaks. A 4:1 ratio is a good place to start, but set the threshold fairly high so that the most of the guitar's natural dynamics are left intact.

When mixing acoustic guitars for rock or alternative tracks, you will usually have an electric guitar or two in the track as well. My personal preference is to pan the acoustic and electric across from each other. Send one full left, and the other full right. You'll quickly discover that the electric will overpower the acoustic and the most effective way to even them out is to compress the acoustic a little bit more than what you may have already done going to tape so you can bring the acoustic's level up high enough to compete with the electric.

Another simple but effective trick is to have the acoustic and electric guitars play parts that counter each other rhythmically (giving them each their own space), and have them each play in a different octave. That will give you a full sounding track that remains open and airy at the same time. You can also make an acoustic guitar sound bigger or more rock-like by panning the original to one side and a delayed signal (short delays are best) of the same guitar to the other side.That effect can be taken one step further by using the pitch change option on your delay to "de-tune" one of the guitars just a pinch (one cent is a good place to start). The delay will provide the brain with the psychoacoustic information it needs to perceive the guitar as bigger, while the pitch change will make it appear "fatter."

Studio Buddy

How do I record an electric guitar?

Recommended mics: Shure SM57, Sennheiser MD421, AKG C1000 and C3000, Neumann U87 or Tube 47.

The single most important factor in getting great electric guitar sounds is that the sound coming out of the amp should be great. That's determined by the guitar, the amp, and the person playing it. In the interest of brevity, let's assume that we have met those conditions and move forward.

As a general rule, it's a good idea to set up the SM57 right against the amp's grill cloth, pointing it directly into the speaker (sometimes at a slight angle from the outer rim of the speaker pointing toward the center). Next, place a condenser mic about two to three feet in front of the amp (at the same height as the amp) and point the mic at one of the speakers. If you have another condenser available, place it about five or six feet away, in front of the amp. You might want to raise the "far" mic to a height of approximately five or six feet off the ground.

By following this formula, you will have given yourself a choice of three different sounds - a close, ballsy sound, a mid-range room sound, and a more distant room sound. By setting all three mics up at the same time, putting them each in a different input, and assigning them all to the same track on tape, you've given yourself the option of having any one of those sounds immediately available.

Today's modern rock guitar sounds tend to be somewhat "dry" (less room ambiance and reverb), and most often use a close mic technique. There's really nothing to it. Simply use the close mic, run it through the compressor, set the compressor at a 3:1 ratio and adjust the threshold so that the compressor is usually working, but not squashing the signal too much. You will be able to make most of the tone adjustments you need at the amp or guitar, and chances are you won't need to tweak the console' s equalizer at all.

For a slightly more distant, but fuller sound, bring up the fader on the mid-distance mic. Slowly add that signal to the close sound described in the previous paragraph. You'll have the detail of the close mic, but with the fullness that comes with adding some "room" sound to it (just like sitting in the tenth row). This is a pretty standard approach that will give you a pretty standard rock guitar sound.

The far mic will give you a bigger, more heavy-metal type of sound with a boomier bottom end on it. The reason for that is low end sound waves take much more distance to fully develop than high end waves. Someone once told me that a low E note on a bass guitar takes thirty-three feet to fully develop. Whether or not that is true will only be known by people who have enough time on their hands to calculate such things.

The key to getting a great guitar sound is to constantly experiment and apply some basic physics. Try different mics, try moving them closer and farther, try different angles, try putting the amp in a corner, try putting the amp on a concrete floor, try it on a wood floor, try it on a floor with green shag carpeting, just try anything!

If you find that you want to eq an electric guitar, you will find that adding 100Hz will give you more bottom, rolling off 300 to 500Hz will eliminate some of the nasal quality, adding a touch of 700Hz will create a throaty or woodsy sound, adding a pinch of 1K or 1000Hz will give the guitar more edge, adding 3K or 3000Hz will give the guitar more bite, and adding 5, 8, or 10K will make it brighter.

Studio Buddy

How do I record a piano?

There are far too many variations on how one might go about recording a piano to cover them all, but here are some of the basics.

Rules of thumb:

1) Condenser mics are preferred by pro engineers

2) Pianos are usually miked and recorded in stereo, but don't necessarily have to be.

3) Much like recording drums, the sound is greatly affected by the touch of the person playing it, the type of music and register it's being played in, and the overall quality of the piano.

Assuming that you are recording a grand or baby grand piano, the most often used set-up is to record in stereo with two condenser mics of the same make and model. A favorite of pro engineers is the AKG 414 because of its robust bottom end, and bright, transparent top end. Neumann KM 84s and KM 184s seem fairly popular as well. Engineers recording classical music are prone to use identical, matched pairs, and often go for very expensive microphones like Scheopps.

For home or project studio recording where the engineer is looking to get great results for far less investment, I recommend either the AKG C1000 or C3000 series microphones. I like to think of them as the poor man's 414. Frankly, they sound pretty darn close.

The best starting point is to place one microphone over the bass strings at a height of 4 to 6 inches, and the other mic over the upper register strings at a matching height. It's important to keep the mics on the same plane to avoid phase cancellation problems. It's also important to keep the mics separated by a distance of a couple of feet to help in avoiding phase problems.

I would recommend using the pad switch on the mics to avoid overloading the console preamps. If the mics you are using don't have pad switches, then use the pad switch on the console. You will find that placing the mics closer to the strings gives you a sound with more attack or edge, while raising the mics up will give a richer, fuller sound with a little less attack.

If you are overdubbing the piano, it's usually preferable to keep the piano lid open to get a more natural, "airy" sound. However, when cutting the piano live with other instruments, it's often necessary to close the lid most of the way to avoid leakage. Most engineers will use gobos or moving blankets to cover the piano opening to further block out any unwanted leakage. Be careful as you are blanketing the piano to not let the blanket move the mic stands or boom arms.

If you are looking to get a sound with a great deal of attack, you may want to try placing both mics nearer the hammers in order to get the attack of the hammers hitting the strings.

Placing your head inside the piano while the pianist is playing the actual part you will be recording (with the lid open), and moving your head around until you hear the sweet spots is a great way to get a good sound, but don't try it if the part being played is loud and could damage your hearing.

It's always a good idea to see where the piano part is being played, and place your mics above those strings. You'll get a more direct sound and find that you'll need less eq both in the recording stage and the mix.

Engineers who are recording classical or new age piano parts will sometimes employ MS or XY miking. Those methods can be very effective, but can be pretty challenging for novice engineers, so I'm not going to cover those methods here.

Many engineers record their piano tracks flat (without eq), and add eq in the mix. If you do record with eq, I would suggest just adding a little bit of top end @ 8 or 10K, while rolling off some bottom on the high end mic. Adding a little upper mids around 4 or 5K to the low end mic might add a little definition and/or "growl" to the low end. Be careful not to add too much in the upper mids to the low end mic, as it may cause that mic to pick up the high end strings a little too much.

Some engineers will limit or compress the piano signal. This is typically done more so on rock and roll parts, and far less on classical parts. Unless you are specifically going for an obviously compressed sound, I would recommend being cautious. A little dab will do ya! Too much compression can be a dangerous thing on a piano, so my advice would be to add it in the mix rather than in the recording.

Just because a piano sounds great in stereo, it doesn't mean that it always sounds best in stereo when in the context of the entire track. Don't be afraid to try a mono piano panned to one side of your mix while balancing it out with a complimentary guitar part panned to the opposite side of the mix. There are times that a mono piano panned at ten or two o'clock can be very effective as well. When going for the middle of the mix, you may want to make sure that the piano part doesn't "step on" the lead vocal's register, making the vocal hard to hear or muddy.

Studio Buddy

How do I record a synthesizer?

Recording a synth is essentially a no-brainer. So much of the sound is "designed" by the player, that you don't really have to do much else to the signal other than avoiding peaks that will break up your mic pre or distort the tape (if you're using tape). It's definitely a good idea to have the player run down the entire part so you can see where the peaks are, and set your threshold and compression ratio accordingly.

If you're recording a bass synth part, you may want to compress it much more than you would a string part. I also recommend using a tube limiter if you have one to warm up the synth.

If you're recording a synth organ (B3, etc.), you may want to set the compressor's ratio at 5:1 but keep the threshold fairly high. By doing that you'll get some nice "natural" dynamics, but be able to slam down any excessive peaks that could cause distortion in any number of places in the recording chain.

Now for the fun stuff. Recording synth strings? Try running the signal out of the control room into the studio or another room, and into a pair of stereo speakers. Then mic the speakers at a distance as if you were miking a real string section. This will give your synthesized strings an airier sound. You can take it a step farther by slightly detuning one side of the signal that goes to the speakers to imitate the natural pitch variances that happen in real string sections. Experimentation is a wonderful thing.

A similar technique can be used to get a more authentic Hammond B3 sound. Place two speakers back to back (or even better, two guitar amps) in the studio. Send the stereo synth signal to the speakers/amps, and mic each side. This will give you the ability to add some "air" to the sound, and if you desire, you can overdrive the amps to add some distortion.

Studio Buddy

How do I record strings or a string section?

Recommended mics: Any condenser will do the trick. AKG 414's and Neumann U87's are often used, but you would be surprised at the sound that many of the less expensive condenser mics will produce.

The key to recording strings is to give them some air to let their full sound develop. When you hear an orchestra, you're not listening to individual players, you are hearing the sum of all the players.

I could write a book on recording large string sections or orchestras, but assuming that most of you reading this are overdubbing strings on pop records in small studios, I'll give you the pertinent information for that discipline.

Let's start with an imaginary section of six violins, three violas, and one cello. Let's also assume that you want to make this relatively small section sound larger, so you will be double tracking it. There for, I'm also going to assume that you will be recording each pass in mono, and then panning them far left and far right.

The set-up is fairly straight ahead. One row of three violins. A row of three violas behind the violins, and the cello in back to eliminate bleed from the violins and violas by virtue of the cello mic pointing into the cello and away from the other strings.

Place one mic (typically in cardioid pattern) at least a couple of feet over the violins with the mic stand placed behind the center violinist with the boom extending out over the player. It's probably a good idea to place the most proficient violinist in the center position as the mic will pick up a little more of that person in the blend. The higher the mic is, the more you will hear the outer players in the blend, and the more "air" you'll get. Be careful, as with most engineering, there is a point of diminishing returns.

Repeat the same procedure for the violas, and mic the cello with a single mic (I like the Electro Voice RE20) pointed at the area between the F hole and the lowest string. Try placing the mic about a foot away from the instrument. For an edgier, more grainy sound, make sure the mic is at the same level as where the bow makes contact with the strings. For a less strident sound, avoid the area where the bow hits the strings.

It's always a good idea to send the players a mono headphone mix containing bass, drums, and a main instrument such as piano, so they can get a good read on the tempo and pitch. Have the players remove one side of their headphones so they can hear their intonation relative to the other players.

Once you learn what a good blend sounds like, I would recommend bussing all three mics to one track, and getting your string blend on one track. Pan that track down the middle while overdubbing the second pass so the players can play to their last pass.

When you get what feels like a good overdub (second pass/double), pan the two tracks far left, and far right. Any major "clams" will readily show up. Slight imperfections are usually acceptable. They'll make your string section sound more like strings, and less like a synth.

I've been known to record a third pass of strings, and pan it down the middle. Be careful, too much of a good thing can be too much of a good thing. The smaller the section, the more desirable the third pass might become.

I've also been known to make the third pass a synthesized string part. Sometimes after the string players have gone home, you may hear an extra part and want to add it. Other times you'll need to drop an extra cello in the middle of the panorama to make a section more emotional, or to make it darker.

A little reverb is usually added to strings in the mix. The reverb you will need is often dictated by the tempo of the songs, and the way the entire production sounds. A good rule of thumb is that for string overdubs on a Pop record, a one to two second "plate" setting is a good starting point. For a quiet, sensitive, singer/songwriter, string quartet part, I might try a small "hall" setting as my starting point. I wouldn't recommend printing the reverb to tape. Save it for the mix.

I would guess that most engineers use a limiter to avoid any unwanted peaks when recording strings, but I wouldn't compress them. If you lose too much of the natural dynamics, you'll end up with a very expensive synthesized sound from human beings!

Studio Buddy

How do I record a cello?

I've tried many kinds of mics on cellos, and remarkably, most of them work fairly well. Most people prefer condensers, but I've had amazing results from Electro Voice RE20's and believe it or not, from the venerable Shure SM57. Almost any $200-$400 should do the trick nicely as well.

The most important thing to remember about a cello is that it takes a few feet of air for the sound to fully develop. If you're not in a situation that requires isolation, I would recommend placing the mic at least two feet away from the front of the instrument, and aiming it in the direction of where the bow meets the strings.

If I were recording a solo cellist, I might also try adding a "distant" mic that is six to ten feet away. When combining the two mics, be careful that you don't get phase cancellation.

A cellist with excellent feel and technique will require very little in the way of EQ, but may need some limiting to avoid unexpected peaks. Compression on a solo cellist would be considered a sin. Reverb should be kept to a minimum, and added in the mix. If you're recording a solo cellist directly to 2 track DAT, I would try adding a little room/hall reverb to the live mix. Go easy.

If you find that you do need eq, I would recommend a pinch of 10K for top end, and a pinch of 30 or 60Hz for fattening up the bottom.

Studio Buddy

How do I record a violin?

Recording a solo violin is very straight ahead. Rules of thumb: Use a condenser mic. Keep at least a couple of feet of distance between the mic and the violin to give it some air. The mic should be above the violin, and looking down at it. EQ should be minimal - if anything, just a little 10K on the top. Use a limiter to catch any unwanted peaks.

Studio Buddy

How do I record an upright bass?

Not that many engineers get the opportunity to record an upright bass. Too bad, it's challenging. The objective is to get the most you can from the bottom end, while still maintaining the detail that comes from the fretboard.

While there are many mics that would be suitable for an upright, I would lean toward a condenser with a large diaphragm. Short of that, don't put a gun in your mouth if all you can rustle up is a decent $300 condenser. Chances are that will work just fine, and nobody but a true jazz aficionado who also happens to be an accomplished recording engineer would know the difference.

The upright presents a strong case for giving distance between the mic and the instrument so the low end waves have room to develop. A Neumann U-87 is a popular choice because it has a pretty friendly bottom end on it combined with just a little bit of edge around 3K that does a nice job of accentuating the buzz and bend that come from the fretboard and fingers plucking strings.

Simply place the mic about two feet away from the front of the instrument aiming for the area around where the fingers pluck the strings. A tube limiter is a wonderful thing on an upright. If you have one, use it. If you don't, not to worry. Any reasonably good limiter will do the job. For a jazz combo sound, you'll want the bass to sound very natural, so go easy on the compression ratio (start at 3:1), and keep the threshold fairly loose.

If you find that the dynamics and the octave spread of a particular tune make for notes that aren't annunciated very well, or disappear altogether, then go for a little more compression. If you need to hear a little more "wood" from the bass, kick it up a couple notches at 700Hz, but be careful, this frequency can be lethal in large doses. 100Hz will fatten up the bottom, and 2.5Khz will give definition, but can also be dangerous in large doses.

Studio Buddy

How do I record a jazz guitar?

The key to recording Jazz anything is simplicity. Jazz guitar is no exception. Jazz tones tend to be mellow and warm. Players tend to use small amps, and they don't crank them up like a stack of Marshalls. Room sounds are usually achieved through bleed, not room mics.

The simplest way to record a Jazz guitar is a prescription that quite frankly works on just about any guitar - a Shure SM57 close to the amp's grill pointing just east or west of the speaker's voice coil will almost always give you what you need.

Other variations would of course include outrageously expensive classic tube mics and equally expensive mic preamps and equalizers, but hey, this is Studio Buddy, not Bank Buddy. We're assuming that you're recording in something less than a multi-million dollar studio.

Going back to relative humility, the Shure 57 is the default mic for almost any electric guitar application, and for good reason - it almost always sounds great. You simply can't go wrong with the 57 up close, through almost any limiter/compressor, and with minimal eq. Try it. You'll like it!

Another mic that I've always loved, but rarely see anymore is the Sony ECM 37P or it's successor, the Sony 377. I've found both of these mics to be exceptional for recording jazz guitar, and if you can find one today, I'd guess that you could buy one for a song (pun intended).

Some engineers will throw a "far" mic on a Jazz guitar to give it some room sound. All fine and good, but I've always found that due to the live tracking on Jazz dates, that I always got plenty of bleed from other mics in the room which gave me a very pleasing "room" sound.

Studio Buddy

How do I record jazz drums?

Recording Jazz drums is not that different from recording a normal drum kit except that the kits usually have less drums, the tuning is different and the drum heads are damped less.

I would suggest that before reading this, you might want to read how to record a snare drum, kick drum, tom-toms, etc., which are all addressed individually within Studio Buddy.

Jazz kits are usually smaller that regular drum kits, and engineers who record them often use fewer mics and go for a more open sound. They accomplish that goal by not damping the drum heads as much, or at all, and by placing the mics further away from the drum heads. While most drums are miked an inch or two above the heads, Jazz drums are typically recorded with the mics placed two to four inches over the heads allowing for more leakage from drum to drum, and more "air" in general.

Some engineers will use just a kick mic, a snare mic, and a stereo pair or stereo mic placed above the kit to pick up the toms and the cymbals. I've also seen some engineers forego typical overhead placements in favor of using mics placed between the toms and the cymbals in a figure 8 pattern. Bet you always wondered what that pattern was for!

Jazz recordings usually have more room sound than Pop records, but that is often a product of intentional leakage among all of the mics in the room. It's done to give a live/ensemble feel to the recording. The drum sound is often the lucky recipient of "good" leakage.

Because leakage and the "airy" sound are desirable, it's very unlikely that anyone recording or mixing Jazz drums would use a noise gate on any of the individual drums.

Studio Buddy

How do I record a trumpet?

Straight ahead, and I mean straight ahead! Pick just about any condenser mic. Pad it at the mic if the mic has a pad switch - if not, pad the mic pre at the console. If the trumpet part is an overdub with no potential leakage issues, then use the mic in omni for best frequency response, and most likely, some improvement in overall sound due to room reflections.

The mic should be placed directly in front of the trumpet's bell at a distance of ten to twenty-four inches. The reason I'm not suggesting closer than ten inches is that you increase the chance of distortion, and the sound coming off the horn's bell will be better with a little distance between it and the mic.

It's recommended that you use a limiter to catch any nasty peaks. Some engineers will throw heavy compression on a solo trumpet, then steep it in reverb with a long decay for an effect.

EQ will most likely be minimal. I do recommend using a high-pass filter if you have one. It's always a good idea to roll off bottom end if the instrument you are recording doesn't have any bottom end. Why muddy things up with air conditioning noise?! If you use a mic that's naturally bright, you probably won't need any additional eq. If you do need any eq, my guess would be just a smidge on the top end - +2@ 8 or 10K.

Studio Buddy

How do I record a saxophone?

A tenor sax can be a beautiful instrument to record. It's rich with all kinds of tones. The secret (like most horns) is to not get too close to the bell. Use the resonance of the whole horn to get a nice full sound. Condenser mics are favored, and most engineers use one mic placed about two feet away from the horn, and aimed at the keys. You'll get the best sound by making sure that the mic's pattern is looking over the top of the sax's bell while aimed at the keys. By doing that, you'll get the sound coming out of the bell combined with the resonance coming off the whole instrument.

As with most horns, judicious use of eq is key. The octave the part is being played in will dictate the overall tone, and where that part fits into the whole track will help you determine if the horn needs any eq. Be careful when eq'ing the sax. It's very easy to over-eq to the point where the sax will step on or mask the lead vocal if the two are happening at the same time. A cool mix trick is to add a couple of db's @1-3K to the vocal, while rolling off a couple at the same frequency on the sax. That will permit a happy coexistence between the two.

It's important to remember that eq'ing an instrument so that it sounds good when listened to all by its lonesome, may not be the best way to eq at all. EQ is more useful as a tool to make instruments stand out or hide behind other instruments in the track. Therefore, it's always a good idea to check your eq in context.

A saxophone is a great instrument to run through anything with tubes in it. A tube limiter is a wonderful thing on a sax. I prefer to go slightly on the heavy side when limiting or compressing a sax. There are a lot of dynamics in sax parts, and compressing the signal will help the sax cut through in the context of the entire mix. I would recommend cutting with some compression, then add more in the mix if necessary.

Tube mics sound great on a sax. So do U-87's and 414's, but my advice would be to lean toward mics that are warm and not too bright. As with other horns, if you're overdubbing the sax, with leakage not being a concern, put the mic in omni. It'll just sound better.

Studio Buddy

How do I record a flute?

The flute is a relatively easy instrument to record. I would suggest using a condenser mic. Unless you're recording a solo album of the first-chair flautist from the London Philharmonic, an inexpensive condenser mic will do just fine. I remember getting an exemplary sound once upon a time using a Sony ECM 50 lavalier mic (like newscasters wear).

The key to recording a flute is to capture the sound of the whole instrument. Easy enough - simply place the mic about twelve to eighteen inches above the flute, looking down. To get more of the lip and tongue (think Jethro Tull, Locomotive Breath) just angle the mic a little more toward the flautist's mouth. To get a less articulated sound, aim the mic away from the mouth, and more toward the keys. Beware of key noise.

If, on the rare occasion you are recording a flute live with a rhythm section, and need some isolation, just stick the mic a few inches from the flautist's mouth and go for it. There are times you'll need to sacrifice sonic perfection in favor of practicality. Remember, nobody ever had a hit record because their flute sounded "perfect."

Flutes have a relatively narrow frequency range, and I've typically found that they require little, if any eq. Rolling off the unused bottom end, using an equalizer or high-pass filter is a good way to eliminate unwanted low-end rumble that will ultimately muddy-up your mix.

Using a compressor/limiter is recommended by varying degrees. If the part being played leans toward staccato, use more. If it's a dreamy legato part, go easy. Judicious use of reverb is always a nice touch, but save it for the mix.

Studio Buddy

How do I record a clarinet?

I would suggest using a condenser mic. A nice warm sounding mic is preferred over an edgier mic. I've also had good results using a Sennheiser MD-441 dynamic mic. It's mellow tone is well-suited for clarinets.

As with the flute, and/or oboe, the key to recording a clarinet is to capture the sound of the entire instrument. Place the mic facing the clarinetist about twelve to eighteen inches from the instrument. Keep the mic at chin level, and tilt it down toward the player's belly button. This is a great technique if you're recording Britney Spears doing a clarinet solo! Beware of key noise.

Clarinets have a relatively narrow frequency range, and I've typically found that they require little, if any eq. Rolling off the unused bottom end, using an equalizer or high-pass filter is a good way to eliminate unwanted low-end rumble that will ultimately muddy-up your mix. I've known some engineers to recommend adding a pinch at 700Hz to make the "low end" of the clarinet more apparent. Sounds good to me!

As with other woodwinds, using a compressor/limiter is recommended by varying degrees. If the part being played leans toward staccato, use more. If it's a dreamy legato part, go easy. Judicious use of reverb is always a nice touch, but save it for the mix.

Studio Buddy

How do I record an oboe?

I would suggest using a condenser mic. A nice warm-sounding mic is preferred over an edgier mic. I've also had good results using a Sennheiser MD-441 dynamic mic. It's mellow tone is well-suited for an oboe.

As with the flute, and/or clarinet, the key to recording an oboe is to capture the sound of the entire instrument. Place the mic facing the oboist about twelve to eighteen inches from the instrument. Keep the mic at chin level, and tilt it down toward the player's belly button. Beware of key noise.

Oboes have a relatively narrow frequency range, and I've typically found that they require little, if any eq. Rolling off the unused bottom end, using an equalizer or high-pass filter is a good way to eliminate unwanted low-end rumble that will ultimately muddy-up your mix. I've known some engineers to recommend adding a pinch at 700Hz to make the "low end" of the oboe more apparent. Sounds good to me!

As with other woodwinds, using a compressor/limiter is recommended by varying degrees. If the part being played leans toward staccato, use more. If it's a dreamy legato part, go easy. Judicious use of reverb is always a nice touch, but save it for the mix.

Studio Buddy

How do I record a harp?

Ahhh... the harp - instrument of the angels. Go with your best condensers for this one - in stereo, of course. Two schools of thought: Some people like to mic the bottom end by placing a mic near or even IN one of the holes in the harp's body, while miking the top end by placing a mic near the higher octave strings. Other engineers like to treat the harp somewhat like a piano - one mic on the low strings, the other on the high strings. I've tried both methods, and have had good results from both.

In an overdub or solo situation, go for the outside mics (not the mic in the sound hole). A couple of AKG 451's or 460's will do nicely, as will Neumann KM84's or 184's. A stereo mic mount with the mics at a 45 degree angle should produce good results.

It's usually a good idea to use a stereo limiter while recording a harp just to catch the peaks, but I wouldn't recommend squashing the signal too much. Very little eq is typically used for a harp, but you may want to roll off some bottom end if it sounds boomy.

Studio Buddy

How do I record an organ?

The most commonly heard organ on records is the venerable Hammond B-3. The Leslie speaker cabinet, used with the B-3, has a rotating horn mounted near the top of the speaker enclosure, and a woofer at the bottom with a rotating baffle over the woofer to disperse the bass notes.

Most engineers mic the Leslie cabinet from both sides, top and bottom. Some of the most commonly used mics are Shure SM57's on the top, and condenser mics with a nice, rich bottom on the lower part of the cabinet - Neumann U87's are often used, but there are many great sounding inexpensive condensers on the market that will do very nicely. Dynamic mics that are well known for their bottom end response will also work well (e.g. Electro Voice RE20).

Place the 57's on opposite sides of the cabinet on the same plane as the rotating horns. A couple of inches from the sound vents on either side should do the trick.

Note: It's very easy to encounter phase problems when miking a Leslie. If the bottom end seems to disappear, try using just one mic on the woofer, and pan the signal down the middle of your mix, or just left or right of the middle. Pan the top mics in stereo. If the organ is playing subtle pads, you may want to pan the top mics to ten o'clock and two o'clock. If you want to feature the organ more in the mix, try going wide with the top mics - full left, and full right. If the organist is doing a solo, try panning the bottom to the left, and the top to the right.

This is another case where you'll want to use a limiter to catch the peaks, and believe me, a B-3 can have plenty. The organ is also an instrument where it is sometimes advisable, if not flat out desirable to squash the signal somewhat.

3Khz can be a good place to add a little EQ to make the organ stick out more in a mix without going for more level. Personally, I like to eq an organ while mixing. It tends to be an instrument that can clutter or mask things in the middle octaves, so it's a good idea to see where everything else fits, and eq the B-3 around it.

Studio Buddy

How do I record a lead vocal?

There are so many ways to record a lead vocal, that it would be impossible to cover them all in the space given here. But, here are some rules of thumb.

1) One of the main objectives is to make the singer feel comfortable so he or she will deliver their best performance. Make sure the studio and control room lighting, room temperature, and general ambience are what the vocalist feels most comfortable with. Unless the singer is the type of performer who likes a room full of people while he overdubs, ask everybody to clear the area. Send them to dinner. Send them home for the day.

2) Make sure the vocalist has exactly the kind of headphone mix they ask for. They need to hear what they need to hear, not what you think they should hear. Check the cue mix by listening to headphones yourself. Don't rely on the monitors to tell you what the singer is hearing. A little reverb in the cans is usually a good thing. Don't print the reverb to tape.

3) Most engineers use condenser mics to record vocals, but there will be rare cases when you'll find that a dynamic mic works better. I generally start with two or three mics that I think will sound good on a particular singer. Record all three to three separate tracks (simultaneously), then play them back and compare. The mic that sounds the best "raw" is the best one to work with. Once it has been chosen, then you can eq and limit accordingly.

4) The choice of microphone will often be affected by the octave the singer is working in because that will help determine the timbre and texture of the singer's voice.

5) Make sure to note the singer's exact position relative to the mic. They will undoubtedly take breaks or work on the same track another day. If you haven't marked their spot, and noted all console and outboard setting relative to the vocal, it will be very hard to match the exact sound.

6) When in doubt, pad the mic. If the mic doesn't have a pad, pad the input at the console.

7) If you're overdubbing the vocal (which is usually the case), it's a good idea to use the microphone in omni. The frequency response will be better, and the overall sound will generally be more "open" or transparent.

8) If you're recording a sensitive or dynamically quiet piece, make sure that you're not picking up air conditioning noise or other low-end rumbles like trains or jets passing overhead. The microphone's roll off switch is a handy tool for that. So is the console's high-pass filter.

9) A pop filter or windscreen is a beautiful thing. I recommend the nylon stretched over a hoop variety over the foam "condom" type. You can usually avoid pops without a

filter by angling the mic slightly across the singer's mouth rather than pointing the mic directly at the singer's mouth, but be careful not to point too far off axis. That will cause a degradation of frequency response unless you have the mic in the omni pattern.

10) For a loud, dynamic vocal, try placing the mic at least six to eight inches a way from the vocalist's mouth. For a more intimate, less dynamic vocal part, try getting the vocalist closer to the mic, but watch out for pops and lip smacks.

11) Because the human voice is one of the most dynamic "instruments," it's a good idea to use a limiter to catch the peaks. There will be times that you will want to compress the vocal by setting the threshold lower, and using a 5:1 ratio, rather than the normal starting place of 3:1.

12) You may want to try adding a little 8 or 10Khz while cutting the vocal. You may also want to roll off some bottom as previously mentioned. Generally, it's best to print the vocal with minimal eq, and save the rest of your eq'ing for the mix when you can judge how the vocal needs to be eq'ed relative to the other instruments in the track.

13) Performance means everything with vocals, so I recommend cutting the track top to bottom and not stopping the vocalist for punch-ins too often. Go for "vibe." Cut several takes on different tracks, then listen back, find the best one, and punch in the fixes on that track. Many engineers will make a composite vocal using the best sections from several tracks, then bouncing them to one composite track. That way, you will always have your original tracks intact until you've built a composite that you're happy with. Then you can erase the original tracks, and punch-in on the composite to clean up any remaining faux pas or bad notes.

14) Don't beat the track or the vocalist to death. Sometimes you'll hit the point of diminishing returns. When you get to that point, take a break. Go to dinner. Work on another song. Work on another instrument. When you revisit the track you were originally working on, the vocalist will be fresh and more productive.

Studio Buddy

How do I record background vocals?

There are typically two types of background vocals that you'll need to record. The first is the double or triple-tracked full ensemble. The second is the "Keith Richards" one or two-liner in the chorus variety.

Think Eagles or Crosby, Stills, Nash, and Young when you are doing ensemble background vocals. Those sweet, silky, multi-part harmonies that are usually spread wide across the stereo spectrum, and tucked nicely behind the lead vocal. To get that effect, simply place your best sounding condenser mic in omni in the middle of your singers standing in a circle. If you aren't using a mic that has an omni pattern, use a cardioid mic, and have the singers stand in a semi-circle around the front of the mic. The distance between the singers and the mic should be determined by the amount of "air" you want. Eighteen inches is probably a good starting point for group backgrounds. Season to taste!

Buss or assign the mic to two tracks (preferably adjacent). Record the first pass on one track, and the second pass on the second track.

Send the singers the track in their headphones, featuring the drums, main melody instrument, the lead vocal, and a good amount of the backgrounds they are singing. Most singers will want to work with one ear of their cans off, so they can hear their blend with the other singers in the room. Once the singers have moved around and found their ideal spots, it's an extremely important idea to mark their positions with tape on the floor. When they take a break, or go to the control room for a playback, you'll want them to return to exactly the same positions when they begin to sing again, or you'll lose their original blend and have to start from scratch. Conversely, there will be times when it's desirable to move the singers around to change the blend for the double track.

Set your autolocator to a point a bar or two ahead of where you want to punch in. Keep running the section by the singers (and recording each pass) until you get the performance you're looking for. Typically, you should be looking for things like pitch, blend, phrasing, attitude, texture, and dynamics. When you've got a take that meets with your approval, pan that track full left and go back to your locator start point. If the singers are all listening to the left side of their headsets (and they should be), then when the chorus passes the next time, they'll hear their live blend in the right ear (the one without the cans), and be able to make that match with the original track (in their left ear) that they're doubling. If the pitch, phrasing, etc. match, then move on to the next chorus and repeat the process. If you've got a hard disk system you can simply copy and paste in to subsequent choruses.

The end result will give you two, multi-layered tracks that can be panned full left and full right in the mix for a spectacular background vocal sound. If you're looking for more "silk," try a third track down the middle with a different blend of voices or notes.

Most decent condenser mics will give you enough top and bottom end that you won't need to eq much - a little roll off on the bottom to keep out the rumble, and maybe a little top end around 3Khz to brighten them up a tad should be all you'll need. Be careful you can always do more eq adjustments in the mix.

Use a limiter to catch peaks. A 3:1 ratio with a fairly high threshold is a good place to start. There will be times you'll want to squash the backgrounds a little more. A little reverb in the cans for the singers is a good thing. Don't print the reverb to tape.

For the Keith Richards type of background, place the singer closer to the mic, and set the limiter threshold a little lower. Typically, the part you'll be going for is a one or two-liner, and is not something that you'll want to double track and split to the sides. I often place this kind of background at eleven or one o'clock in the mix. I also find that I sometimes use less reverb on a part like this.

Studio Buddy

How do I record congas?

Congas are pretty straight ahead, and often sound great with nothing more than a couple of Shure SM57's on them in stereo. Place the mics about eight inches above the drums, and angle them from the center of the two drums looking out toward the sides. Make sure the mics "see" where the hands/fingers actually hit the drums.

Add a tad of 4-5Khz to brighten the top, and roll off a little low end on the lower conga if it's boomy. Put a limiter on each mic. Use a fast attack and release with a 4:1 ratio and a mid-level threshold, and you're good to go. Typically you'll want to record each drum on its own track and pan the tracks opposite each other. The width of your pans will be determined in the mix. For a tune where the congas should be featured, go wide. If they should be further back in the track, go narrower.

Studio Buddy

How do I record timbales?

Timbales, like congas, are pretty straight ahead and often sound great with nothing more than a couple of Shure SM57's on them in stereo. Some engineers use condenser mics. I don't think they sound any better, and you'll be risking the mics getting hit with the sticks. I've sent a couple of mics to their graves while recording timbales.

Place the mics about ten inches above the drums, and angle them from the center of the two drums looking out toward the sides.

Add a tad of 4-5Khz to brighten the top. Put a limiter on each mic. Use a fast attack and release with a 4:1 ratio and a mid-level threshold, and you're good to go. Typically you'll want to record each drum on its own track and pan the tracks opposite each other. The width of your pans will be determined in the mix. For a tune where the timbales should be featured, go wide. If they should be further back in track, go narrower.

Studio Buddy

How do I record a shaker?

Shakers are incredibly dynamic instruments. Pad the mic or the preamp, and keep the gain low. You'll be shocked to see how little shows up on a VU meter, while you're breaking up your mic pre or tape. When in VU, I typically record a shaker so the VU is barely moving. When the meter is switched to peak mode, it is probably nailing +3 or higher.

Just about any mic will do, but lean toward mics that have plenty of headroom. Use duller mics for bright shakers, and brighter mics for dull shakers. The last thing you want is a "spitty" sounding shaker! A Sennheiser MD 421 is a good all around shaker mike. A Shure SM 57 will also do just fine.

You'll almost always want to roll off the bottom end (a shaker doesn't have any), and you may find that you'll get a nicer sound by rolling off some lower mids around 400 to 700Hz. That will depend on the particular shaker.

One of the best shakers I've ever owned was made of two metal 35 mm film canisters with their butts taped together, and the insides filled with a teaspoon of rice in each. For a mellower sound, fill the canisters with pot seeds, but don't get arrested in the process of procuring them. Just clean the seeds from the pot, and discard the leftovers (right?!). And don't tell the cops where you got the idea!

An old dbx 160 limiter sounds superb on a shaker. Almost any other limiter is recommended as well. I tend to compress my shakers quite a bit.

Studio Buddy

How do I record a tambourine?

Tambourines are possibly the most dynamic instruments you'll ever record. Pad the mic or the preamp, and keep the gain low. You'll be shocked to see how little shows up on a VU meter, while you're breaking up your mic pre or tape. When in VU, I typically record a tambourine so the VU is barely moving. When the meter is switched to peak mode, it is probably nailing +3, +6 or higher.

Just about any mic will do, but lean toward mics that have plenty of headroom. Use duller mics for bright tambourines, and brighter mics for dull tambourines - if there is such a thing. If the tambourine sounds sibilant, you're breaking up something in the recording chain. I'd put my money on the pre-amp. A Sennheiser MD 421 is a good all around tambourine mike. A Shure SM57 is also a good bet. I prefer my tambourines to sound a little "gritty," so I typically go with a dynamic mic. For a more bell-like sound from the little mini-cymbals (Ôjingles' in tambourine parlance) contained within the tambourine, go with a condenser mic with plenty of headroom.

You'll almost always want to roll off the bottom end (a tambourine doesn't really have any), and you may find that you'll get a nicer sound by rolling off some lower mids around 400 to 700Hz, although you should be careful not totally eliminate the sound of the hand impacting with the instrument.

An old dbx 160 limiter sounds superb on a tambourine. Almost any other limiter is recommended as well. I tend to compress my tambourines quite a bit.

Studio Buddy

How do I record a horn section?

Recording a horn section is just plain fun. It's one of my favorite things to record.

I recommend not miking each horn individually. That's for wussies. I like to record a horn section much like I record background vocals - as a group that has to find its own natural blend.

Use the best condenser mic you have, and if you can, put it in omni. If your mic doesn't have an omni pattern don't fret, cardioid will work just fine. Pad the mic. If the mic doesn't have a pad, pad the mic pre.

Assuming you're working in omni, place the horn section around the mic in a circle or semi-circle. Run the track by the players and have them work on their blend by moving them closer or farther from the mic. Eighteen inches is a good starting point.

The players will probably want to have one ear of their headphones off to hear their blend relative to the other players. Send them a mix that has plenty of backbeat, and at least one main melody instrument for tuning purposes.

Once they're tuned up and are playing along with the track, adjust their balance again if need be by moving the players in and out. Once the perfect balance has been found, have the players or an assistant mark everybody's spot with some tape on the floor. That will give you the same blend over and over again if the players take a break or come in the control room for a playback.

Lay down the first section until you feel that you got the perfect performance. Pan that pass to one side of the mix, and have the players double track it. When you get a good take on the double track, move on to the next section and repeat the process. I've also been known to do a third pass of just baritone sax (if there is one), and pan it down the middle for extra "balls."

Definitely use a limiter to catch any nasty peaks. If you can get your hands on a tube limiter, all the better. Tubes sound great on horns. EQ on horns is a touchy subject. To play it safe, go with as little as possible. Maybe just a little 10Khz on the top end if you need to brighten them up. The more you record and mix horns, the better you'll understand the fine points of eq'ing them. Much of it has to do with the octave the part is in, the makeup of the section you're recording, and the dynamics of the part that's being played.

How do I record a French horn?

The French horn is very easy to record because it doesn't have a lot of high end or low end information, nor does it have a very wide dynamic range. Because the horn is played with the bell facing rearward, it's best miked from the rear with the capsule facing in to the horn's bell. In an overdub situation, it's often advantageous to place the player near a wall or a corner, and use a microphone in the omni pattern placed between the horn and the wall or corner. This method will give some natural ambience, and allows for the full rich sound of the horn to be captured, rather than just getting the direct sound that comes from the bell.

Just about any mic will do a pretty good job of recording a French horn. An Electro-voice RE20 is one that has been favored over the years, but dynamic mics as well as condensers will work well on the French horn. Compressor/limiters can be used to catch unwanted peaks, but in an orchestral situation, engineers seem to shy away from any compression.

Studio Buddy

Is it okay if the band records live and some instruments bleed into different mics?

Yes, a little bleed of instruments into various microphones is fine, but there's one condition. Every player has to get it right, or the whole band has to try the song again. If the musicianship is solid and everyone is focused, there's nothing that beats the feel and cohesion of a band playing live together. And if there's a little guitar in the drum mics, more often than not, that bleed adds to the overall sound. The vibe you've achieved is worth the little bit of control you've lost when you don't have complete separation from track to track.

The reason so many records and demos are made the other way - isolating the instruments and treating all but the drums as "scratch" tracks - is that sometimes the musicianship isn't ready for doing it all live. And that's okay. Often a recording project should proceed anyway: the songs are there, the studio is available, the singer's out of rehab...No point in waiting another year until everyone has their chops down just so you can record live.

Studio Buddy

How should I blend all the background vocals with the lead if I've sung everything myself?

If there's only one singer available for the lead and all the background vocals in a song, there are a few things you can do to make the tracks distinct in the mix.

EQ - Since the singer will have certain tonal qualities on every track, you can use eq to separate the lead and background vocals, so you don't have the exact same frequencies building up over several tracks. For instance, you could let the lead vocal retain the full sound you've captured on tape, with plenty of warmth, brightness and sheen. Then you could thin out the background vocals a little by cutting some low-end (say, -3 db at 200Hz). You don't need to take out so much that it sounds affected, just enough to give some distinction.

Reverb - Try not to use the exact same amount of reverb on all the vocals. Even if it's important that it sound like all the singing was done in the same environment, it helps to give each part of the arrangement its own space in the mix, especially if one singer is doing all the vocals. A common approach is to use a little reverb on the lead vocal (maintaining a healthy amount of presence) and more reverb on the BG vocals, setting them further back in the mix landscape.

Compression - this is another subtle way to add distinction between the vocals. Not only can you use different amounts of compression in the mix, you might even use different types of compressors, such as a tube comp/limiter on the lead vocal and a solid state device on the BG's.

Panning - this is a somewhat obvious way to make sure your vocals don't double and triple up on each other. If you've doubled your lead vocal, don't pan it exactly up the middle where your lead is - pan it around 11:00 or 1:00. BG's can go further out.

Mic Placement - you may prefer to record all vocals with the same mic placement, in order to get the best possible sound on tape and deal with this issue in the mix. Or you might get experimental, having the singer step back a couple feet for the BG vocals (or stand in the back of the room, or sing into a megaphone, etc.)

Studio Buddy

How many tracks should I use to record the lead vocal?

More often than not, the lead vocal is the track that contains the most emotional content of the song. With repeated attempts at recording the vocal, you run the risk of losing that emotion and "magic." So while it's ideal for the singer to nail the perfect take in one or two tries, a good engineer knows how to respond the other 90% of the time.

The answer is to compile the best elements of a few different takes into a single, composite performance where each line, each phrase and even each syllable is sung just the way you want. This process is called "comping." It's done on nearly every record you hear, even the ones you're convinced are single, complete takes.

Tip: if the singer is hesitant to record this way, claiming "artistic integrity," remind them that they're free to sing the song through from top to bottom, without interruption. Meanwhile, just switch tracks while you're winding back to the top after each take. (Make sure you're only sending the current take to the headphone mix - it can be very disconcerting for a singer to begin a song and hear two voices coming out of his mouth.)

In this digital age of virtually unlimited available tracks, it's tempting to record 5 or even 10 different takes before comping the vocal. But using that many can really overwhelm you and confuse the process. Try utilizing two or three tracks instead. Starting with your first take, tell the singer it's only a practice take for the purpose of further level adjustment (when in fact you've already adjusted everything and are ready to go.) This is useful for anxious singers, taking the "pressure" off them.

After two or three takes, stop if you have terrific performances overall. If not, go back to the track with the least inspired take and record over it. Hopefully, you have gained the singer's trust by now and don't need to inform them of these details. Continue with this process until you feel that, within those two or three tracks, you have the makings of a great performance.

When you're ready to start comping, draw lines on the lyric sheet so you can make little notes (check marks, yes, no, good, bad, maybe) on each line of each take. Involve the singer in this process only if they insist - the more they analyze their own performance, the less they're likely to respond with an inspired, heartfelt one. Once you have usable takes for each line, bounce the winners onto a fresh track (you can also bounce certain lines from "alternate" takes into one take that just needs a few fixes.)

Tip: after you have a comp'd vocal, get away from it for a while (dinner break, t.v. break, whatever). Then listen to it with fresh ears, and with the singer, to see if you still need to fix something.

What can I do about sibilance?

Sibilance typically refers to the hissing-type sound when words with an ÔS' hit the mic's diaphragm too hard. Other letters can have a similar effect, such as ÔF' and ÔT'. You can address this a few different ways. Singers who have excellent mic technique will carefully aim their mouth a little "off-axis" (not directly at the microphone) for the split-second when an especially sibilant word occurs. Pop filters can also be helpful here (see "When should I use a pop filter?") But even with professional singers, sibilance will get recorded now and then. This is where a de-esser is used. It's a signal processor that only looks for sibilance, and you should dial in the parameters (e.g. threshold and amount of attenuation) so that you eliminate the sibilance without dulling the sound of the vocal throughout the take. Sometimes you'll find a de-esser as part of a mic-preamp. Use this judiciously if you have to use it at all, because vocal performances vary from one take to the next, and it might be smarter and safer to control the sibilance in the mix.

Tip: Sometimes when you record double and triple vocals, even with harmonies, certain letters start to flam if they don't "land" at exactly the same moment. For example, if the lead singer and two background singers sing the line "take it to the limit," it can sound like "t-t-take it to the limit." One way to avoid this is to have one or both of the BG singers sing "ake" instead of "take."

It seems crazy, but on playback you won't notice. (Don't make this a habit on multiple vocals so much as a trick to use if you have a problem.)

Studio Buddy

What does it mean to have good "Mic Technique?"

Having good "Mic Technique" means two things: understanding that a microphone's diaphragm reacts with varying sensitivity to your vocal performance; and knowing how to adjust your body according to the dynamics of the delivery.

If you're going to deliver a quiet, intimate vocal from start to finish, you can afford to position your mouth just a few inches (or even less) from the microphone. If the vocal is to be sung full volume throughout the song, you may stand a couple feet away. Quite often, though, a song is dynamic enough to require different amounts of air to be pushed at different times. Singers with good mic technique will move their body closer to or further away from the mic as the song unfolds. Ideally, your mouth is as close to the mic as possible before overloading it with level (which will cause it to distort or, with super-sensitive mic's, to temporarily shut down - this will always be blamed on the engineer, even if it's the prima-donna singer's fault.)

Tip: don't be afraid to move your head back a couple inches for just a phrase or even a syllable - you can also aim your mouth slightly above or to the side of the diaphragm for particularly loud moments.

Studio Buddy

How do I fix pitch problems on a vocal?

Fixing pitch problems has never been easier with the invention of digital autotuners like the ones made by Antares, a device so popular that it is regularly being used not only for pitch correction but for its own noticeable effect, just like a flanger or phaser. (If you do have access to this device, be careful to use it judiciously. To many ears, the overuse of it - where you can actually hear it working - sounds cheesy and, well, trendy at best.)

If you don't have access to an autotuner (they're available as a computer plug-in or a standalone effects box) there are more traditional ways for an engineer to fix pitch problems. The first is to re-sing it! Seriously, one way to disguise a "pitchy" vocal is to use another take as a double. This works well if the lead vocal is consistently out of tune throughout the song. Another way is to use a little chorus effect, sometimes referred to as "smearing" or "harmonizing." Ideally you'd have a device that accepts a mono input and spits out a stereo output. By setting one side of the harmonizer's output a few cents under pitch, and the other side a few cents above pitch, you can then blend this sound in with your dry vocal and disguise occasional pitch problems.

Studio Buddy

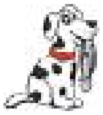
How can I get a unique vocal sound?

The opportunities to make a vocal sound unique are endless, bound only by your imagination. Sometimes the more obvious effects - 'telephone'-like filters, heavy-pumping compression, ethereal reverb - are exactly what works for the song. But you can also have the vocal sung into a megaphone, or come off tape into a guitar pedal, an amplifier, even the Leslie speaker that was built for Hammond organs (if you're lucky enough to own one.) A lot of digital effects boxes will simulate these sounds, but they don't always come out as good as the real thing. Wah-wah and distortion pedals are extremely useful in giving your vocal a different sound. And you can get great kinds of distortion by deliberately overloading a circuit. Try patching your vocal, from tape, into a mic-pre with its gain turned all the way up. Every model of mic-pre out there produces its own type of distortion when overloaded, so if you don't like the sound of one, try again with another. This trick also works with compressors - just turn the input all the way up. (Note: if you try this idea, start with the fader down on the channel where the signal is returning.)

And remember that too much effect can come off as gimmicky. Blending just a little bit into the main (dry) signal allows you to create a sound that's fresh without drawing attention to itself. (Of course, sometimes that's the point.)

Tip: Are you looking for a unique vocal reverb? Before you send the vocal to the reverb unit, patch it into a flanger first. If you dial in just the right amount, the listener may not even pick up on your little trick. But the overall vocal sound will be unique and more interesting.

Studio Buddy



How do I get rid of distortion?

Start by visualizing the problem. Somewhere in the chain there's a circuit that's being overloaded or has become incompatible. Eliminate each variable, one at a time, until the distortion goes away. Then you have isolated the problem. Some typical examples of distortion:

1. Electrical buzz - the guitar amp is plugged into the wall and the amp starts buzzing. Try plugging into a different wall outlet or into an AC strip first. Or plug into a ground lift first (every studio should keep a bunch of these lying around - they're cheap.) Try dimming the lights to see if the AC is just overloaded. If the buzz goes away, maybe work with candles until you can get someone to update your wiring. Or plug the lamp in somewhere else if that's the culprit. Also, you can insert an iso-transformer somewhere in your chain, which might do the trick.

2. Electrical hum - the effects processor starts to hum after warming up. You may have a bad tube or another worn part. If you can't replace the bad part and you must use that piece of gear, try eq'ing the hum out - it's usually around 60Hz (outside the U.S. it might be different due to various electrical standards.)

Determine which frequency contains the hum by boosting at 60Hz and sweeping around until the hum is at its loudest. Then cut as much of that frequency as possible without ruining the sound you're working on. (If the bass amp is humming, this trick won't work. Try recording the bass direct and re-amping the signal after you've fixed the amp.)

3. Microphone distortion - when a microphone's diaphragm receives too much signal, the sound seems to be "breaking up." You can move the mic back an inch or two, you can aim it slightly off-axis, you can play lighter (o.k., that's not really an option.) If it's an amp you're miking, you might be tempted to turn it down a little, but remember - sometimes an amp doesn't achieve the right tone until you turn it up a certain amount, so moving the mic further back may be the thing to do. If you're using a tube mic, instead of distorting, the mic might just Ôfold' instead - that is, the signal will suddenly disappear. In that case, turn your monitors down, power down the mic, wait a few minutes and turn it back on. That usually takes care of it (don't forget to move the mic away from the sound source a bit.)

4. Crackling - you probably have a bad cable or one that's only half-patched (which can also put your signal out of phase.) Try reinserting your cable or replacing it with a new one. Or patch the gear into a different channel of the console. If you have an analog console with channels that pull out, try reseating the one that is crackling.

5. Analog distortion - you're recording a vocal and every now and then you hear it distort (during the loud sections, oddly enough,). You have a few things in your vocal chain - an outboard mic-pre, a compressor, an equalizer, the line input on the console, and finally, the actual track you're recording on. The first places to check are your mic

and line inputs and the input to your tape machine. (You probably will see a red light indicating you've overloaded the circuit, but use your ears instead of your eyes.) Still, even effects processors can distort if they're being fed too much signal. Just isolate the distortion and turn the knob down a notch or two - making up for the gain loss somewhere else in your chain if you need to.

6. Digital distortion - there's a common myth that if you are using digital gear, you can't go into the red, ever, because red means automatic distortion. Not true. In fact, a little bid of red is a healthy thing when you're recording onto a digital multi track, DAT, or even CD-R. It's important to pack as much of your signal as close to 0 as you can, thereby utilizing as many of the available bits as you can. Occasional peaks that land in the red are fine. Remember to trust your ears. If you hear distortion, back the input down.

Studio Buddy

How do I use room mics? When should I use room mics?

Rooms mics are used to give a recording a more ambient and live sound. One of the best-known examples of an obvious use of room mics would be the Knack's "My Sharona."

Room mics can be used in stereo or mono, but one must always be aware that they do have an effect on the phase relationships of all the other instruments being recorded simultaneously in the same room.

Engineers often use stereo room mics placed about six feet high, and about ten feet from the drum kit to enhance the drum sound. Other engineers will use a single mic in omni placed near the floor, and pointed at the kick drum. There are endless combinations of placements, and ultimately, each has to be the decision of the engineer and producer. There is no pat prescription for placing room mics.

Home studio users may find that it's easier and safer to use room simulation programs on a reverb device to achieve the same effect. The programs have gotten so good over the years, that it's often just as satisfying and more expeditious to just use a box.

One last note: It's usually a good idea to not print room mics on the same tracks as your drums. Print them on separate tracks, and blend them with the direct tracks during the mix.

Studio Buddy

How can I make my guitar stop humming?

Oldest joke in the engineer's manual: How do I make my guitar stop humming? Teach it the lyric!

But seriously, guitar hum can most easily be cured by picking up a ground-lift plug at your local hardware store for less than a buck. You'd be surprised what ills that little device can cure. Simply plug the AC cable from your guitar amp into the ground lifter, and plug the ground lifter into the wall outlet.

Single coil guitar pickups, like the type found on Strats are much more prone to hums. Hum can usually be lessened by rotating the player and the guitar. There's usually a spot that will hum less - call it a null point.

Turning off fluorescent and neon lights in the vicinity will also help a great deal. Other things to check are improperly grounded guitar cords, patch cords, or outboard equipment.

Studio Buddy

How do I get rid of mouth pops?

Obviously, most engineers use pop filters to eliminate pops, but there are more than one kind of pop filter. Many stage mics already have them built in, hence the large ball-shaped screen over the mic's capsule. But most studio mics use a foam pop filter or windscreen that slides over the end of the mic. Those work well, but some people think they eliminate some of the microphone's ability to capture high end sounds. It's an arguable point.

A more popular type of windscreen that has emerged in the last decade is the nylon stocking variety. In its simplest, home-brewed version, it is just a piece of nylon stocking stretched over an "o" shaped piece of sturdy wire like a section of coat hanger. The filter is placed between the singer's mouth and the microphone to eliminate any blasts of breath that would cause a pop. There are several companies that now manufacture the nylon stocking type of pop filter.

Maybe the most effective way of all to eliminate pops is to just place the microphone so that the singer's mouth doesn't blow directly in to it. Placing the mic slightly off to one side, but angled at the singer's mouth will almost always cure a popping problem. Just make sure that you haven't placed the mic so far off-axis that you affect the sound of the mic by going outside the mic's pick-up pattern.

Studio Buddy

How do I get rid of lip noise?

The best prescription in the world for getting rid of mouth noise is to simply have the singer drink water, and lots of it. Another method that is sometimes used, but not often talked about, is to use one of the personal moisturizing products available in any drugstore in the feminine products area. That one's sure to get a few laughs around the studio.

Studio Buddy

What is the difference between a dynamic microphone and a condenser microphone?

In the simplest of terms, a dynamic microphone is basically an iron core surrounded by a coil of copper wire much like an electromagnet. When sound waves hit the core and move it, it causes the core to move within the coil which generates electrical impulses that become translated into sound when they go through a mic preamp.

A condenser mic (or electret condenser) is essentially two extremely thin, metal (typically gold) partical-coated mylar membranes which are separated by a very thin insulating layer of air. One side is positively charged, the other is negatively charged. When sound waves, or sound pressure hits the "diaphragm," it creates electrical impulses that become translated into sound when they go through a mic preamp.

Generally speaking, dynamic mics are less expensive, are less delicate, handle extreme sound pressure levels better than condenser mics, but don't sound as good as condenser mics. There are many situations in which a dynamic mic is the better choice though. Many engineers use them on drums of all types. They are very well-suited for applications where high sound pressure levels are anticipated.

Condenser mics are generally thought to be richer sounding, with more "detail." But while they may sound better, they are also more sensitive to high sound pressure levels, and somewhat prone to distortion if exposed to too much level. Condenser mics often have variable pattern switches on them, allowing engineers to choose a cardioid pattern, hyper-cardioid, figure eight, or omni.

Studio Buddy

What does cardioid mean?

Cardioid is a microphone pick-up pattern that for all practical purposes resembles a heart in its shape. Draw a heart, then round off the bottom point. Now imagine that the notch in the top of the heart is located at the mic's capsule. What that means is that the mic will pick up with the most level and best frequency response from the front. The sides of the pattern will give decreased level and frequency response, and the back (the notch) will give minimal pick-up and frequency response. This pattern is most useful when you have multiple instruments in a room, and you are trying to reduce the amount of "bleed" from one instrument to others.

Studio Buddy

What is a polar pattern?

A polar pattern is the pattern from which a microphone picks up the incoming sound. There are several types of polar patterns including, omni, cardioid, hyper-cardioid, and figure 8.

Omni is omni-directional, meaning the mic picks up sound equally well from all directions.

Cardioid is roughly heart-shaped, with the notch of the heart at the rear of the microphone. In other words, the mic picks up best from the front, with the frequency response tapering off at the sides, and practically no ability to pick up from the rear. This type of pattern is commonly used to eliminate bleed from other instruments. It is also the most frequently used type of polar pattern.

Hyper-cardioid is very much like cardioid, but with a tighter pattern, thus eliminating even more bleed from other instruments.

Figure 8 means that the mic picks up from two sides. A practical use would be to place a mic in figure 8 between a tom-tom and the cymbal that is above it. In the figure 8 pattern, you would pick up sound from the drum and the cymbal, but not from the tom-tom next to the one you want to pick up, nor from any other instruments off to the sides.

Studio Buddy

How does a limiter work?

A limiter is essentially a gain reduction device that senses the amount of level coming into the device, and keeps the level from exceeding a user determined limit. In even simpler terms, a limiter stops audio peaks that might distort something in the audio chain. A limiter is also the same thing as a compressor, except that in the compression mode, the limiter is used to "squash" the overall signal, not just catch and reduce peaks. The result will be that sounds that may have been difficult to hear are more apparent, as well as sounds that may be too loud will now be less offensive.

A basic limiter consists of input control, a threshold control, and a gain reduction control. The threshold is set by the engineer, and determines the point at which the limiter/compressor does its gain reduction. The gain reduction control is typically expressed in terms of a ratio. For example, a 3:1 setting would mean that for every 3 db of level over the threshold, the limiter will only output 1 db.

Studio Buddy

What is an equalization curve?

This is a really basic explanation, but one that worked for me when I was first learning the craft of audio engineering. An equalization curve, or eq curve is an imaginary horizontal line that shows how much or how little of certain frequencies are heard in a given audio signal.

Imagine a horizontal line that represents low or bass frequencies on the left, and high or treble frequencies on the right. If you started with the line being flat, then boosted the high end, you would see a bump on the right end of the line. If you boosted the middle frequencies, you'd see a bump in the middle.

When looking at a microphone's eq curve, you would typically see a range of frequencies from about 30Hz on the low end, up to 20Khz on the high end. The curve would start out below the "zero" line until it hits 60 to 100Hz, then be flat until the upper frequencies (around 10Khz), where you would see a slight bump above the line showing that the mic has a brightened top end to compensate for the natural roll off of the human ear. You would also see a rapid roll off after 10Khz, because most humans can't hear above that frequency, so it's essentially useless for a microphone to pick it up - unless of course, you're making records for dogs!

Studio Buddy

What is a noise gate?

A noise gate is a device that allows a signal (sound) to pass through as long as it's above a certain threshold. Any sound below that threshold is cut off completely. A typical use of a noise gate would be to "GATE" a snare drum to keep out extraneous noise from other drums in the drum kit. Another use might be to gate out a buzz that comes from an electric guitar. When no part is being played on the guitar, the signal path is gated closed. As soon as the player begins to play, the gate is opened, and the signal or sound passes through.

Studio Buddy

What is phase cancellation?

Phase cancellation occurs when one sound source hits two microphones at almost identical times. ALMOST is the key word here. A very slight difference in time delay can cause comb-filtering, which means that a normal signal that might look like a sine wave develops "teeth." The resultant wave-form looks somewhat like a comb that you would use on you hair (assuming you're not Telly Savalas).

The comb-filtering effect usually means that certain frequencies are more pronounced than others. If you were to get phase cancellation on a bass guitar, the bottom end frequencies would drop out and the bass would sound thin. The closer the mics are to the sound source, the less likely one would be to encounter phase cancellation. It's always a good idea to reverse the phase or move microphones a little bit if you think a particular instrument sounds "funny" or thin.

To get an idea of what something sounds like when it's out of phase, try reversing the leads on just one of your stereo monitors. You'll notice that the sound seems to come from "outside" the speakers, instead of between them. If you put your incorrectly wired speakers in mono, you'll notice that you'll get dramatic reduction in some very important frequencies. That's phase cancellation!

Studio Buddy

Why do my drums sound thin and boxy?

Most likely, you are causing phase cancellation by poor microphone placement. The quickest remedy is often just a minor adjustment in mic positions. The culprits are often the overhead mics. Try moving them closer to the cymbals. The next most likely place to look is the rack mount tom-tom mics. Try moving them closer to the drum heads, and make sure that the mics aren't too close to each other. Other things to avoid: place the kick mic in such a way that it doesn't aim through the kick and at the bottom of the snare drum. Also try angling your overhead mics out to the sides slightly. A good rule of thumb is to always make sure that your mics are twice as far apart from each other as they are from the source when miking in stereo or when using multiple mics such as with a drum kit.

Studio Buddy

What does a high-pass filter do?

A high-pass filter is simply an equalizer that "rolls off" all frequencies below a predetermined frequency, typically 60Hz or 100Hz. In even simpler terms, it's an equalizer that gets rid of unwanted bottom end and lets the highs "pass." A common use would be to eliminate air conditioning rumble coming in through a microphone.

Studio Buddy

What does a low-pass filter do?

A low-pass filter is simply an equalizer that rolls off all frequencies above a certain point - let's say, 10Khz. In other words, it only lets the "lows" pass. A common use would be to eliminate a high-frequency buzz coming out of an electric guitar.

Studio Buddy

What is a gobo?

A gobo is a device used to block sound from leaking from one instrument into another while recording. In other words, it's an acoustic isolation device. A gobo is typically made of plywood, 2'x4's, fiberglass batting, and covered in fabric.

Gobos come in all different sizes. Smaller ones may be used to block sound coming out of a guitar amp. Larger gobos may be used in orchestral recordings to isolate an entire section.

Studio Buddy

What is a click track?

A click track is a metronomically perfect beat or "click" that is sent to the drummer's headphones (and possibly the rest of the rhythm track players) when cutting a rhythm track. The purpose is to keep the tempo steady.

In "ancient" times, the click was generated by an old spring-loaded metronome, then miked, and sent in to the cue system. That method has been replaced by electronic click tracks that are commonly built into drum machines today.

The click track is often printed to track 23 or 24 on a multi-track so that it's always available for use. Sometimes, it's just used to establish the tempo, then dropped out of the drummer's cue mix. Other times, the full band will want to work with it.

Some drummers just can't use a click track -- it's too foreign. Most pro drummers have practiced playing with a click at home for many years. It's a developed talent.

Studio Buddy

What is a patchbay?

A patchbay is a centralized section in your studio that allows you to easily manage your signal flow. Most of the inputs and outputs of recording gear are located on the back of the device. Instead of running around behind the console and racks of gear every time you want to send a signal somewhere, you can do it all at the patchbay.

(Before buying one, you need to decide what type of connector you want to use. The two most popular are 1/4" and TT. The main consideration is rack space - TT inputs take up much less space.)

A proper patchbay is set up so that all the "normal" operations of signal flow work without anything plugged into the front of the bay. That is, buss 1 of the console goes to track 1 of the recorder, the stereo mix is sent automatically to your DAT machine, etc.

Setting up a patchbay is a matter of running input and output cables from all your gear to the back of the patchbay. In a set of two rows, it's usually wired so that outputs appear on the top and inputs on the bottom.

Say you want to send buss 1 to both track 1 and 6. Insert a patchcord, coming out of the top row "buss 1" output to the track 6 input, all at the front of the patchbay. (In a "half-normal" configuration, making this patch does not prevent the signal from also going to track 1.) If you want to insert a stereo compressor between the output of the console and the input of the DAT machine/CD burner, this can also be done at the patchbay.

You can have certain cables wired in a parallel configuration (if, for instance, you want the output of the DAT player to be patchable to two cassette decks,) or an open configuration (you wouldn't want the inputs and outputs of your CD burner connected to each other.)

Tip: Say you've just recorded a vocal, using buss 5, onto track 5 but the singer hates it, and insists that the vocal be erased and wants to lay down some whistling instead - immediately. You want to preserve track 5 for a future vocal (for consistency), and you're ready to use the same signal chain for the whistling. If you simply insert a patchcord from buss 5 to track 6, the whistling might also appear on track 5. To avoid this and still wipe track 5, you'll need a "deadpatch." This means one end of a cable goes to the input of track 5, and the other end is left hanging, going nowhere. You'll have recorded the whistling on 6 and utter silence on 5 in one pass.

Studio Buddy

What is a Ômult'?

A "mult" refers to the routing of one signal to two or more places.

There are several instances where a mult is useful. For example, you may want to have a delay effect on the lead vocal, but only for the last line of each verse. While you could easily use an aux send from the console to get the vocal to the effects box, that would limit your ability to control exactly when the vocal hits the delay.

By running a mult - holes in the patchbay set up to multiply access to your signal - you can patch the vocal to a separate channel, which is then unmuted only at the right spots.

Another instance is if you have a snare drum that needs a lot of work in the mix. Maybe you want to eq the dry signal right off tape. And you want to overly compress the snare and blend it in to the overall drum sound. And maybe you also want to gate the snare because there's too much kick drum on that track. Patching into a mult simply allows you more ways to manipulate your signal.

Studio Buddy

Should I print my effects to tape?

You should almost never print effects to tape unless you are really short of inputs or tracks. Why commit to a sound that you may want to change later? The context around that particular sound may change, and you'll be stuck with something that no longer works. An obvious exception would be when using a device like the built-in reverb on your guitar amp that might be hard to duplicate with an effects box.

This doesn't mean that you shouldn't print eq and/or compression to tape. I don't consider them an effect in most cases. If you are just starting out, you may want to reserve most of your compression and eq decisions for your mix. As you become more experienced, you'll find that you'll feel more confident about what you commit to tape.

Studio Buddy

How should I start my mix?

Start by plugging in the basic gear you'll most likely need, e.g. primary reverb, delay, and a stereo compressor (if you have one) between your console and your 2-track recorder (DAT, CD, etc.) Even if you start with the compressor in bypass, it should be plugged in. As you build your mix, you should occasionally check your levels to the recorder (see "setting 2-track levels")

When you're ready to push up the faders (i.e. ready to listen to the tracks), you have a couple of options:

1) if you have a full arrangement (e.g. vocal, guitar, piano,bass & drums), imagine you're building a house. Start with a solid foundation - first the drums, then bass, then the rest. But remember that the vocal is the most important instrument in a song - as you build your mix, push up the vocal every now and then to make sure the track will ultimately support the vocal and not overwhelm it.

2) if you have a relatively sparse arrangement (e.g. acoustic guitar, vocal and violin), start with either the vocal first or the most important supporting instrument.

Studio Buddy

How should I approach the mix process?

If you think of songwriting and recording as an artistic process, it's not hard to draw an analogy between mixing a song and painting a picture. Let's say you're looking at a painting of some French farmland. Your eyes move from the green rolling hills, to the creek winding past the red farmhouse, to the farmer greeting the cow as the sun shines above.

Now think of your mix as a "musical landscape." Every element is distinct but integrated, flowing into each other. A great mix allows the ears to wander from one sound to the next but always focusing back on the main elements - the melody, the hook, etc. In your painting, the frequencies are your colors. Allow your instruments to stand out, with some colors brighter than others.

Studio Buddy

Why should I use a stereo compressor on my entire mix?

The more tracks you have recorded for your song, the more information there is competing for space on just two tracks when you mix down. Compressing your mix makes the individual tracks take up less (dynamic) space without losing their sonic impact.

As you begin your mix, start conservatively with your compressor settings: the attack and release should be relatively fast, set the ratio at 2:1 or 3:1, and set the threshold so that only the peaks are getting over the top and being "squeezed."

You've probably heard the phrase, "Punchy Mix." Stereo compression is a key element in making your mix more "punchy."

Studio Buddy

Should I use eq on my stereo buss while mixing?

EQ'ing the stereo buss - the two wires that funnel all the various sounds from your mixing console to your 2-track recorder - is generally discouraged. Since this eq affects the entire signal, from your bass guitar to your lead vocal and everything else in the mix, you're likely to affect some sounds you didn't intend to. For example, if you decide that the mix is lacking bottom end, try adding a little here and there (on the bass, the kick, toms or whatever needs it). If you add bottom end to the whole mix, you're also adding to the vocal, the cymbals, percussion, etc. It's best to leave such overall eq'ing to the mastering engineer, who doesn't have the luxury of addressing individual tracks, but has trained his ears specifically for the art of overall eq.

Studio Buddy

Where should I pan my tracks in the mix?

You can pan your tracks anywhere you want that sounds good to you, but here are some tips on how it's usually done:

Lead Vocal - almost always panned right up the middle, aka Center or 12:00 (panning positions are often referred to as positions on a clock.)

Bass guitar / Kick drum - these also tend to be panned at 12:00. Typically responsible for most of the low frequencies in the mix, bass & kick get spread evenly in both channels, thus making them appear to be in the center.

Snare & rest of drum kit - pan these according to how the drum kit was set up: snare up the middle or maybe 1:30, toms @ 2:00, 10:00 & 8:00, hi-hat @ 3:00, ride @ 9:00, overhead/cymbal tracks @ 5:00 & 7:00, stereo room tracks all the way right and left - known as "hard right" and "hard left", mono room tracks @ 12:00. (This drum panning is known as "audience perspective." You could also reverse it to have "drummer's perspective.")

Guitars - if you have one main guitar that carries the song, try panning it around 10:00 or 2:00; if there's a second guitar part, use the other position of these two. If you've "doubled" your main guitar, pan the two tracks hard left & right. This should sound like one huge guitar. These suggestions work for both electric & acoustic guitars.

Keyboards (piano, organ, synth) - if your keys play a fundamental role in the song, pan them either hard left & right or maybe 3:00/9:00. If they play more of a supportive role in an ensemble or otherwise dense arrangement, you can tuck them in the middle. For instance, the piano could be panned at 8:00/1:00 and the organ at 11:00/4:00, which allows you to hear the spread of the keys without these instruments stepping on each other.

Background Vocals - if you have one harmony vocal, or a double of the lead vocal, pan it just off center (11:00 or 1:00) so the two vocals blend well but remain distinct. If it's "oohs" & "aahs" you've recorded, try around 10:00 & 2:00.

Miscellaneous "Ear Candy" - some sounds, like synth pads, percussion, or even subliminal voices, can be panned just about anywhere. For incidental sounds that may only last a short while but need to be noticed, (e.g. a cool drum loop) try panning them hard left or hard right so they can stand out without overwhelming the mix.

Just as you add the elements of a mix like building the foundation of a house, panning the instruments is like painting a landscape.You want to spread things out so that all the elements can be "seen," or in this case heard, but you also want to maintain a sense of balance so that the mix doesn't become left or right heavy.

If you have a midrange mono guitar part and a mid range mono keyboard part, you may want to try panning them opposite each other to give width and balance. The same can be true for almost any combination of instruments.

Studio Buddy

How do I balance the kick drum and the bass guitar?

Another classic battle in the mix process is the kick drum vs. bass guitar. That's because they share similar frequencies. While in some scenarios (e.g. a folk song) the bass plays a far more important role than the kick, more often they are expected to carry an equal load.

Low-end Tip: try assigning different low-end frequencies to each instrument. For example, you could boost 60Hz (with a somewhat narrow bandwidth) on the kick drum, while boosting 100Hz (also narrow bandwidth) on the bass. Or vice versa. This allows them to fill up the bottom end without competing with each other.

High-end Tip: Usually, though not always, you want to see the kick and bass as well as feel them. That is, you want to hear the actual notes of the bass and the pattern the kick drum plays. Finding the right upper-midrange frequencies to feature is the answer. On the kick, try adding around 2K for presence and 10K to put a "point" on it. For the bass, it might be 5K or 8K that allows the instrument to speak in your mix; you can also look at 800Hz for an aggressive bass sound.

Bottom line: Be careful to let these two instruments occupy their own space. Adding the same frequencies to both will just muddy up your mix.

Studio Buddy

Cutting (Rolling-off) vs. Adding Equalization

The natural tendency when reaching for an eq knob is to add certain frequencies to a signal. For example, you're listening to the mix, and you realize you can't hear enough shimmer on the cymbals, so you decide to add 4 db of 12K to your overhead tracks. But another effective way to make each instrument distinct in the mix is to remove unnecessary frequencies. This creates more space and allows each signal to breathe more.

Tip: As you consider each track of music, think about which frequencies it might be pushing that you really don't need to hear. Take the bottom end: if you want your kick drum and bass guitar to sound as big as an A-bomb, see if other instruments are competing at the same frequencies. Maybe you don't need 50-100hz on the electric guitar, or the lead vocal. If you're worried about making those instruments too thin-sounding, add a little 200hz. High-pass and low-pass filters are extremely useful for this purpose. With percussion like shaker and tambourine, you can filter out a lot of extraneous low-end eq to clear-out room for other instruments in the mix. Remember: all these instruments on your multitrack are competing for valuable space on TWO tracks in mixdown. Cutting certain eq is a challenging, but rewarding, way to make room for everyone.

Studio Buddy

How do I get rid of a boxey sounding kick drum?

While different songs call for different kick drum sounds, most of the time the engineer looks to feature the drum's most appealing characteristics: a booming low-end and a sharp attack. The kick's inherent ugliness, where it sounds "boxey," resides around 500hz. Sometimes it's closer to 400hz. Whichever, it's always a good idea to address this while recording, but you can also deal with it when you mix.

Hopefully you have an equalizer that allows you to specify the bandwidth, because in this case a really narrow bandwidth (aka a "high Q") will allow you to zero in on the most offensive frequency without affecting the rest. Start at 500hz, and take it all out. That's right, if the maximum attenuation is 12 db, then cut 12 db. Suddenly, the kick sounds a lot rounder. After you've decided which frequency to focus on - look at the range of 400-550hz - dial some of it back in if the sound you've achieved is unnatural. This is a cool trick because few instruments have a certain frequency that's so inherently ugly you can simply get rid of it all.

Studio Buddy

How do I know which frequencies to cut?

As you become a more experienced engineer, you'll learn which frequencies to look at depending on which instrument you're dealing with. Until then, you can quickly learn what to cut with a simple trick called the "boost and twist."

Say you're dealing with a guitar (though this works for any instrument) Grab the db knob of that channel's equalizer and boost the signal as much as you can. Then twist the frequency knob around the dial until you hear the most horrifying, unappealing sound you can find. Stop. That's the frequency you want to cut. Then turn your db knob the other way until it sounds right to you.

Tip: If you're trying this while a musician is helping you get sounds by playing, and they have headphones on, you'll learn to do this test as fast as possible to avoid freaking them out or pissing them off.

Studio Buddy

How much compression should I use on individual instruments?

After the microphone and mic-pre, compression is the engineer's single most valuable tool in terms of signal processing. It's also the most difficult to understand. Essentially, compression can be used to control the level (volume) of your signal and also to manipulate the signal so it sounds effected, vibey, etc.

Use a little amount to limit the peak levels of the instrument. This can be employed to fix inconsistencies in a musician's performance (e.g. a bass player who doesn't hit all the notes evenly) or to keep an appropriately dynamic performance from "jumping" out of the mix too often. In these cases, set your threshold so only the peak (loudest) sections get processed, and set your ratio conservatively (2:1 or 3:1, for example.)

If you want to use the compression as an effect as opposed to an invisible tool, be more liberal with your parameter settings. Open up your threshold and crank up the ratio until you hear the compression really working - often called a "pumping" effect. This is great on vocals, bass guitar, or anything else when you want to set your mix apart from everyone else's.

Studio Buddy

Should I use delay instead of reverb?

The most common choice for novice engineers who want to make an instrument sound "special" or more like a record is reverb. And while reverb is an extremely useful effect for giving mixes their own unique ambiance, there's another way to achieve that sound without eating up as much space in your mix.

Next time you want to liven up an instrument without sounding gimmicky or obvious, send its signal to a delay unit. A delay usually consists of some, but not all, of the parameters you'd find in a reverb processor. You can use a short delay to add thickness (try 4-8 milliseconds with 10-20% feedback on a rhythm guitar); you can go a little longer for a "slap" effect often found on lead vocals (try 30, 60, 90 or 120 ms to employ an Elvis or John Lennon vocal effect.) Or you can use a really long delay (e.g. 800 ms w/ 40% feedback) for a more obvious effect, like having certain words of the vocal repeat and trail off in the distance.

Tip: For your lead vocal, try using a delay and a reverb. The delay is only used to thicken the vocal - dial up anywhere from 30-90 ms, with little or no feedback, and leave the fader just below where you can detect the sound of the delay. Then utilize the reverb (short hall, small plate, etc.) to create the more noticeable ambience for your vocal.

Studio Buddy

How much reverb should I use on my lead vocal?

Most of the time people think of reverb in terms of extremes: either they like a lot of reverb or none at all. But there's a middle ground that's very useful when you want a natural sounding vocal that's neither too wet nor too dry. Like when you want to process it so it sounds unprocessed.

Tip: Listen to your vocal (with reverb) in solo and dial-in a cool, vibey reverb that has a relatively short decay and 0-2 reflections (feedback). In solo, the reverb should be plenty audible. Then take those faders out of solo and while listening to the whole mix, adjust the amount of vocal reverb to the point just below where you can detect it. By setting it to where you can't hear it but it's definitely there, you're using the reverb more as glue between the singer and the band than as an obvious effect. This is great for when you want the whole band to sound like they're in the same room without settling for a totally dry, unexciting ambience.

Also see: Reverb vs. Delay

Studio Buddy

How do I give my vocal more presence without making it louder in the mix?

Use EQ - add a few db of upper-midrange, anywhere from 2K to 8K (5K is a popular sweet spot for adding presence.) This can get the vocal noticed without pushing up the fader. This is called "apparent loudness."

Use Compression - take a "mult" (using the patch bay to make a "Y" cord) of your dry vocal, (off tape and before any eq, compression, reverb you're already using on the vocal) and patch it into a compressor - preferably one that allows you to hear a little "pumping" as opposed to being relatively transparent. (The classic compressor /limiter for this effect is the UREI 1176.) Set your knobs liberally, so you're getting 8-20 db of compression. It's not supposed to sound that good all by itself.

Then bring that signal up on a fader and, panned right up the middle (or wherever your main vocal fader is panned) blend in the highly-compressed vocal into the main vocal. You'll likely pull the fader down on the main vocal until you find the right balance. In the end you should have a vocal that isn't overdriving your mix buss and pinning your meters, but jumps out of the mix with more presence.

Studio Buddy

Why does the lead vocal always seem too loud in my mixes?

One of the great battles in the mix process is between the lead vocal and the snare drum. Engineers know the challenge of getting a terrific snare sound on tape. And since we usually start a mix with the drums, it's a common mistake to give the snare undue priority. In the frequency spectrum, the snare and the vocal occupy similar space. So by the time you add the vocal to the mix, you might struggle to have it heard because you're subconsciously trying to protect the snare's prominence in the mix.

Tip: There's a saying that "there's no such thing as too much lead vocal," and it's mostly true. But you still want to create space around the vocal so it doesn't have to fight to be heard. After the vocal is introduced to the mix, bring the snare fader down to where it's out of the way but still propelling the rhythm with the kick drum. Remember, most people don't turn on the radio or buy a CD to hear a killer snare drum. It's the vocal that sells the song -- umm... well, that may not be true for disco or dance music, but that's another story!

Studio Buddy

How do I make my recordings sound unique?

The answer is to break the rules every now and then. Of course, before you can break the rules, you need to learn the rules. By picking up the tricks of the trade, you can get to the point where your recordings sound as big and compelling as the platinum records on the radio. But those engineers didn't get there by simply copying everyone else.They pushed the envelope and came up with a new approach, often by breaking the rules.

One of the most exciting things about the digital revolution is the availability of affordable software that gives you great sounding tracks with minimal effort. But with the proliferation of programs like Sonic Foundry's ACID, keep in mind that thousands of people are using the same sounds as you. In order to make your songs stand out, occasionally do the opposite of what your supposed to do. Some examples:

1. The next time you need to dial up a killer vocal reverb, don't twist the knob until your processor reads "Killer Vocal Reverb." Look further - why not use the preset called "Electric Bass" or "String Quartet"? The trick is to know when such a move enhances the track, as opposed to weakening it or just sounding gimmicky.

2. Usually the least appealing frequency in a kick drum sound is 500Hz or so. Most of the time you'll want to get rid of it completely. But maybe you've got a song (or a section in a song) that needs some new colors. Why not boost the very frequency that's so unappealing? And boost it a lot. Your kick drum may not sound like any other in memory, but maybe that's the point.

3. It's common to hear a tambourine enter a song on the downbeat of a chorus. It provides a nice lift. But why not let the tambourine start a few bars before the chorus? It's unexpected, and it can heighten the excitement of the groove going on before the chorus hits. (And you still get a lift in the chorus.)

This approach can work for you if you use it selectively. When every element of a recording sounds wrong (the vocal is buried, there's no bottom end, the hi-hat is louder than everything else combined - you get the picture) the sound comes off as amateurish. But if most of the elements are about right and one thing is off (like a well-balanced mix that has a trumpet blasting out of the right speaker for eight bars), you've got something that's memorable and unique.

A final thought: e.e. cummings didn't begin his career by writing without punctuation. He started by learning the rules of great poetry. By the time he ditched the periods and commas, he'd mastered all the rest - imagery, character development, strength in an idea, etc. How does this apply to making records? Just that it's easier to rise above the others once you've learned the tricks of the trade.

When should I Ôpad' a signal?

To 'pad' a signal means to attenuate, or lower the volume of, a signal. The first place you might need to do this is if a signal you are recording is too loud for a microphone's diaphragm, causing distortion. Some mics have pads built into them, usually in either 10, 20, and/or 30 db increments.

Another area in your chain where padding is necessary might be between the mic and the mic-pre (some classic mic-pre's run extremely 'hot' and do not have a built-in pad. For this scenario you can plug the mic cable into a "line attenuator" to reduce the level of the signal hitting the mic-pre.

And the third opportunity to pad your signal is at the mic-pre itself. Maybe you want to record something extremely loud and aggressive, but you don't need the kind of distortion you're getting due to the mic-pre being overloaded. Pushing the pad switch there can also give you proper control of your signal.

Studio Buddy

How do I get the feel of an instrument to sit right in the track?

Sometimes one musician on a session can't find the pocket but manages to play consistent time - they're just a little behind or a little ahead of the groove. (Oddly enough, this is a common result of a poor headphone mix. If a bass player has too much bass in their cans compared to the drums, they might play behind the beat; too little bass and they're on top of the beat.) Not to worry. This can be "fixed in the mix" with the use of really short delays.

The easiest scenario is when one player is slightly on top or ahead of the beat. To fix it, run a delay unit between the multitrack and the console. (For self-contained digital workstations, you'll probably have a delay that's accessible on each channel.) We're only talking about a few milliseconds here, just enough to lay the errant track into the groove.

It gets a little trickier if the performance is behind the beat. The traditional fix in pro studios, with expensive analog multitracks, was to put the 2nd output in 'sync' mode (which plays back 50-60 milliseconds before the 'repro' head) and patch the offending track into a delay before coming back to the console.

These days you can fix it much more easily. If you're working with an MDM (modular digital multitrack recorder), you can simply delay all the tracks except the one that's behind the beat. Of course, the ideal solution is to have the musician play it again.

Studio Buddy

Is it easier to mix if each part of the drum kit has been recorded onto a separate track?

Whether you like to use many mics and several tracks for the drums or go for a more minimalist approach, there are pros and cons either way when it's time to mix the song. Splitting up the drum kit into individual instruments (e.g. separate tracks for kick, snare, hi-hat, toms, cymbals, and near & far room mics) certainly gives you more control during mixdown. If you need more kick drum, it's easy to raise the level without affecting the snare's volume. Besides the obvious downside of possibly running out of tracks too soon, there's also the issue of phase cancellation. The more mics you have aimed at a drum kit, the more likely they will be somewhat out of phase with each other. This can lead to a smaller sound overall as certain frequencies get cancelled out (see "phase" question.) It's a common myth that more mics/tracks equals bigger-sounding drums.

There are a couple good reasons to record drums with only a few mics. First, it saves tracks you might need later. Second, it forces you to commit to getting a good sound on tape (as opposed to spreading a sound over 10-12 tracks, hoping that some blend of which will give you a decent snare sound.) Third, it's a good safety mechanism against overstating the drums' presence in the mix. There's a tendency among novice engineers to overindulge themselves while balancing many tracks of drums. The resulting mix can focus too much on the drums, when the song might require a better balance between the instruments. With only 4-5 tracks dedicated to drums (e.g. kick, snare, overheads, and a mono room mic), you might find yourself moving on to the other instruments sooner and avoid subconscious drum worship.

The downside, of course, is less control. (You want more toms, you get more cymbals.) The minimalist approach is recommended once you've gained some confidence as an engineer. Knowing how to anticipate the drums' role in the song also helps. Will they simply be providing the rhythm and momentum, or will they also be relied upon to bring excitement and dynamics to the song?

Studio Buddy

Do I need to use both the direct signal and the amp sound when I mix the bass into the track?

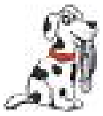
If you were able to record the bass guitar using both a direct box and a mic in front of a bass amp, you'll have much more flexibility in dialing in the perfect bass sound in mixdown. It's common to rely on the direct signal for the attack and presence of the bass, while using the amp to provide the bottom end and roundness (the technical term is "whoomph.")

While countless records have been made where only the direct signal was recorded, there's something to be said for the sound you get when you turn up the amp until the speaker cabinet really moves some air.

A lot of times an engineer is limited not by the number of tracks or mics available, but by the size of the studio. If you're tracking a band that has guitar, bass and drums, but want the option of fixing or re-doing some guitar or bass, you'll need to keep the amps from bleeding into the drum mics. And if the studio layout doesn't provide for decent separation (lack of iso booths, closets, etc.) you may be forced to turn off the amp and just go direct with the bass.

Tip: If you find that the bass DI (direct box) alone just isn't cutting it in the mix, but you like the original performance, you can always send the signal, off tape, back into the bass amp and record the amplified sound as an overdub. They even make "reamp" boxes for this purpose, making this old trick less "tricky" in terms of matching your impedance properly. (And if you're out of tracks, just run the amp as you mix and bring it up on a fader - but remember to turn your amp off periodically so it doesn't overheat and turn it back on when you're ready to print the mix.)

Studio Buddy



How do I make the guitars sound really big in the mix?

There's a misconception that, in order to have an electric guitar sound really big on a record, you need to mix the guitar much louder than the other instruments. The truth is it's all about how well you record it to begin with. If you've done that right (turning up the amp until you're "moving some air", aiming the mic or mics so you get the full range of frequencies, recording without compression, etc.) you'll be in great shape for the mix. Remember, an electric guitar makes its impact not just from your tone or how fast you play, but also in the way it relates to the groove provided by the drums and bass.

And then there's the persistent dilemma: "Do I double this guitar part or not?" There's something undeniably huge about doubling a rhythm guitar and panning the two tracks hard left and right. But before you commit to that, think about what other parts will be added in the arrangement. If there's a solo, what would the doubled guitar play there? (Put the solo on a separate track?) Is the rhythm guitar the featured instrument, or will there be several others competing for space in the stereo spread?

If you do decide to double the guitar, think about altering the sound slightly on the double track. This can give you a little more thickness. You can change guitars and keep the amp the same, or vice versa. Maybe you just change pickups on the second track. Whichever, make sure your performance is really tight, matching the first track's phrasing note for note. Otherwise you'll have a cluttered mix that would be better off with only one track of guitar.

Studio Buddy

Why can't I get my effect to sound natural throughout the song?

Sometimes an engineer gets an idea in his/her head to put a cool-sounding effect on one of the tracks... say, an 800 ms delay on the lead vocal. Then they spend three hours dialing it in and trying to balance it into the mix so it sounds natural, only to be frustrated because the delay doesn't flow well throughout the song, just on the third line of every pre-chorus.

This is a reminder to be choosey in your use of effects. If the delay doesn't work except twice in the song, just use it twice. If you dialed in a massive, thundering reverb for the drum kit, but it sounds awkward when the drummer is playing cross-stick, then don't forget to pull the reverb down in those sections.

When making records, some people prefer to take a "naturalist" approach - as in, "I don't want to do anything that I can't recreate live." And if that's your philosophy, so be it. But don't feel confined to any rules about what you can or can't do in the studio. Let's say you've got an acoustic guitar that starts the song, and it sounded so good you decided to double it on another track. But when you get to the mix stage, you realize that the two guitars only sound good in the intro and the breakdown. Once the band kicks in, that second acoustic just clutters up the track. Just lower the fader or mute the track all together for those parts of the song. Don't worry about whether the listener might be 'confused' by the disappearance of the second acoustic. If you can balance the tracks to where it feels natural, that's all that matters.

Studio Buddy

How do I make my mixes sound unique (pt. 2?)

One way to avoid mixes that sound like every other project studio out there is to build your own echo chamber. This may seem like a ridiculous idea, given the home recording revolution that has put every imaginable effect and device into affordable little boxes. But to further abuse an already overused clichŽ, why not "think outside the box?"

Remember, the same recording gear you can afford is what every other project studio engineer has also just purchased. And most people never venture past the basic presets that come with this gear. So if you have the means, try something adventurous like building an echo chamber in your studio.

You don't a huge space, just a cool-sounding ambience that will help to set your mixes apart from everyone else's. Here's what you do. Run a patch from an aux send or buss on the console to a separate amplifier (any cheap stereo amp will do.) Then run a speaker off the amp to your designated chamber - it can be a bathroom, hallway, garage, attic, you name it. Then set up a cheap mic in that space. You'll want to put some distance between it and the speaker, but how much and where the mic is aimed is up to you. To do this properly you'll need an assistant. Have them move the mic around while you're in the control room listening for a sweet spot. Remember: the amount of reflections you get is determined not only by where you place the mic (straight at the speaker or up in a corner facing the opposite wall) but how much signal you send to the amp. The louder it is, the more times the signal will bounce around your newly created echo chamber.

Simply return the mic's signal back to the console using an open fader or an "echo return," and combine the signal with the original to hear the reverb. Obviously, the more of the chamber you mix in, the more reverb you'll hear. You can also use stereo speakers and two mics in stereo in your "chamber," and return the signals to two open faders and pan them opposite each other to get a stereo chamber.

Studio Buddy

How do I get my mix to sound good on everybody's stereos?

The answer is simple: try to check your mix in as many different listening environments as possible before putting it to bed. Not only is it smart to have a couple different sets of monitors at the console, but you can do more. Make a cassette or CD and check it on a boombox. Listen in your car. If you have a stereo in your living room, check it there. (One pro studio even created a one-watt radio station that allowed you to sit in the parking lot and tune in your mix on the car radio! The clients loved it.)

With console mix automation more affordable these days, schedule time to get away from the mix for a day or two. Take the mix to your friend's house. Each of these playbacks is likely to reveal a weakness in your mix. A perfect sounding bottom-end might suddenly overwhelm the mix on your friend's stereo.

This can be a baffling process. (How can my mix sound awesome in the first four places and still sound wrong this time?) But, of course, every set of speakers has its own frequency response. Mixing is a matter of little tweaks and balances until it's just right. It's not easy. If it were, everyone would do it.

Studio Buddy

How do I make sounds appear to come from "beyond" the speakers?

There are a couple of scenarios in a mix where you might want to make it sound like the instrument is coming from "outside" or "beyond" the speakers. Maybe you have an extremely dense arrangement and need to create space for everything. Or maybe you just want a unique effect. Regardless, you can utilize the concept of Ôphase relationship' to achieve this.

Most of the time an engineer aims the microphones so that all the signals are in phase. For instance, if you have two microphones being used to record a kick drum, the mics are placed so that the peaks and valleys of the drum's waveforms arrive at each microphone at the same time. This allows the final kick drum sound to be as full and complete as possible.

Tip: In mixing, you might want to have some fun with phase. Say you have a big band playing a dozen instruments, and the musician on B-3 organ is just playing subtle pads and swells. A supplemental role. If you have recorded the signal in stereo, you can put it out of phase and make it sound like the organ is sitting to the left and to the right of your two speakers.

First, "mult" (combine) the two stereo tracks of B-3 with an aux send or buss. Then return that new mono signal to a fader. In simpler terms; Send equal amounts of signal from each of the B-3 tracks to an aux send. That will create a mono "mix" of the stereo B-3 signal. Patch out of that mono "return" to an open fader. Pan the original tracks hard left and right, and pan the new mono signal up the middle, at 12:00. At this point, when you push up the new fader, the organ should just sound louder. But when you pop the phase button on that fader - reversing its phase relationship with the original stereo B-3, you'll hear an odd thing happen. It will sound as though the B-3 was "pushed" to the sides, even wider than the actual speakers. As you play with the fader level of the mult (the mono signal), you can decide how much of that signal sounds good to you, and how wide you want the B-3 to "appear."

Studio Buddy

Should I compare my mix to other recordings or just focus on my own sound?

Checking other recordings can be an extremely useful tool in the mix process. Here are a few reasons why:

1. Your mix will (hopefully) not exist in a vacuum when it's done. It will be heard before and after other songs by other artists, recorded and mixed in other studios. Certainly you want your work to stand up to records being purchased and played on the radio.

2. Even if certain factors - limited studio gear, your developing talent as an engineer, etc. - make it unrealistic that your mix will be as huge and impressive as the latest chart-topper, comparing your current mix with current hits can be both a reality check and inspiring. Your once thunderous drum kit suddenly seems small and tinny after a/b'ing with a favorite CD. Do you let this defeat your spirit? Or do you listen again, critically, until you can learn what the differences are between the mixes? Don't worry that getting into this habit will make you a mere imitator. Comparing mixes is just a fraction of what you do at the console.

3. Don't just compare to hit records. Use your own past work as a reality check. The deeper you go into the 'trees' of a mix, the harder it is to see the 'forest.' Every few hours, play an old mix of yours that you know is great. Even if you listen to only a minute or so, this practice will often reveal a flaw in your mix much faster than thinking about your mix really hard. Some engineers will place six of their favorite CD's - some current hits, some of their own work - by the CD player and have them playing continuously while they mix. Every time the tape is winding back to the top, they switch to the CD playback for 30 seconds, then go back to the mix. (Not as useful a habit when you're working on hard-disk.)

4. If you find yourself working in an unfamiliar studio, bringing along some of your past mixes and a few favorite CD's is an absolute must. How else can you learn the characteristics of the control room and make sure your mix leaves the studio as you intend it?

Studio Buddy

Why should I print my mix as 'hot' (loud) as possible?

There's a common myth that if you are using digital gear, you can't go into the red, ever, because red means automatic distortion. Not true. In fact, a little bid of red is a healthy thing when you're recording onto a digital multitrack, DAT, or even CD-R. It's important to pack as much of your signal as close to 0 as you can, thereby utilizing as many of the available bits as you can. The more bits you use, the more resolution you have, which means - in laymen's terms - Bigger! Fatter! Stronger!

Occasional peaks that land in the red are fine. Unless you've deliberately squashed your mix with a stereo compressor on the way to your DAT or CD, you'll have some dynamics in your levels. If you have a "soft-limit" feature like the one on Apogee A/D converters, you can set your peaks at 0 db. Otherwise, go ahead and let a couple peaks go over 0 when you print the mix. (Some CD manufacturing plants will complain about too many "overs," but a few here and there - along with your written approval - won't be a problem.)

If you get nervous about going into the red, remember to trust your ears. If you hear distortion, back the input down.

Studio Buddy

What's the best way to fade out a song?

Getting a good fade is all about feel, and the only way to learn that is to practice. Each song will dictate how short or long a fade should be, although there is one concept that is common in most fades: during the first half of the fade the volume decreases gradually, and in the second half it decreases more rapidly.

If you want the fade to be obvious, start it at the beginning of a section and give it a quick drop of -1 or -2 db. If you want to be more subtle - the fade has begun before the listener has even noticed - start it a couple bars into the section. Say you have a double chorus at the end of your song, and you want to be subtle. The first lines of the chorus are "Love you till the end of time/Baby, baby say you'll be mine." Let one full chorus go by. Then at the double chorus, begin the fade, slowly, after the word "Baby."

Then there's what is commonly referred to as "the Motown fade," which is really quick; the whole fade from top to bottom only lasts 4-8 bars. And if you are fading on a vocal hook (as opposed to random vocal riffing) try to have the fade end between vocal lines. This is a pretty minor point, but it can subconsciously make it easier for the listener to remember the hook.

If you're struggling to get the fade right but can't, it might be that the music is no longer interesting by that point. Maybe the band is vamping on a chord progression and there's some random soloing. If you're not happy with how the song is ending, don't be afraid to go back and re-record a part or two to keep the song compelling until the very end.

Studio Buddy

Why should I check my mixes in Ômono'?

It's true that we've been living in a stereo world for many years now. Television is broadcast in stereo. FM radio is stereo. And AM radio mostly broadcasts talk shows or older music recorded in mono to begin with. So why does 'mono' matter? Because there are still some cases where your mix might get played back in mono or something similar to mono. For instance, there are several wireless speakers on the market that are switchable from either left channel/right channel or mono. There are alarm clocks that'll play a CD but only have one speaker. And more often than not, PA systems are set up for mono.

So what might you discover when you listen to your mix in mono? Some stereo effects panned hard left and right can practically disappear in mono playback. This might not be a concern if you're only dealing with reverb returns. But if you've recorded a piano in stereo and panned the tracks hard L/R, the balance in relation to other instruments may seem totally out of whack in mono.

There are ways to combat this. You might buss (send) both channels to a third channel, panned up the middle, and blend it in a little. Or you can move your hard panning in toward the center a bit, maybe around 7/5 o'clock or 8/4 o'clock.

Bottom line: This issue is really minor compared to, say, 20 years ago. But it's worth spending 15 minutes per mix checking the balances in mono, just to be sure that your work will retain its impact no matter where it is heard.

Studio Buddy

Should I use headphones when I mix?

It's not a bad idea to occasionally check your mix on headphones. For the same reason you'd want the luxury of playing back your mix on a variety of stereos before putting it to bed: a great mix should sound "right" in the studio, on a boombox, in a car, and even on headphones.

But this is a tricky area, because the way sound impacts the ear from such close range is entirely different than any other means of reproduction. The size of the listener's ear, for example, is now a factor. Or where the headphones sit on their head. Many of the most popular phones in the world - the ones you get for free when you buy a Walkman - are inherently tinny and cold sounding, with hardly any bass frequency response.

So be careful not to make critical mix decisions based solely on how it sounds in the headphones. And make sure you understand the characteristics of the phones you're using. Probably the best advantage to checking the mix with the "cans" is to get a fresh perspective. A new angle almost always exposes some flaw or at least suggests a question you hadn't yet considered.

Studio Buddy

What can I do about Ôear fatigue'?

The first step is acknowledging that ear fatigue is a common problem in the studio, primarily during overdubs and mixing. The best way to combat this is to work at low volume. While turning the monitors up may inspire you to create a great mix, you can work much longer with a clear head by keeping the levels down. (Naturally, you'll need to crank it up every now and then to make sure the mix is exciting.)

Tip: If the guitar player is in the control room and needs to hear it really loud - maybe even to the point of creating feedback through the control room's monitors - wear earplugs and let the musicians' ears bleed if they want.

One of the keys to a successful mix is to take frequent breaks. Maybe it's a five-minute break every hour or a 20-minute break every few hours. It's easy to get so deep in the process of tweaking levels and frequencies that you start to lose perspective. Eventually, you don't realize that you can't hear anymore. If the client is there with you and paying you by the hour, don't be shy about stepping away from the console several times.

Also, be careful about packing too many hours into a mix session. It's tempting when you're on a roll (or if you've blocked out a studio for 24 hours) to keep going for 12, 16 or even 20 hours of mixing. But there are two reasons to cut your session short before then. One, your ears naturally shut down after too much exposure, and finding the right frequency to tweak can get elusive. Two, there's a tremendous benefit to sleeping on it. You can learn the same things in fifteen minutes the next day as in 3 hours of late night hair-pulling.

Studio Buddy

Can I go back and forth between analog and digital signals?

Technically, of course, you can go back and forth between analog and digital signals as often as you like. But, ideally, switching from one domain to the other should be done as little as possible. That's because every time you convert a signal to/from analog and digital, there's a certain amount of degradation (even with high-end, meticulously crafted converters.)

Let's say you have a digital mixer and a couple of MDMs (modular digital multitrack recorder). You know you want to utilize the vintage compressor and the tube equalizer in your rack for a lead vocal you're about to record. If you were recording with an analog console & multitrack, you'd have more options for when to employ the compressor and eq. You could use them going to tape or coming off tape. But since you're working with digital gear, put the compressor and eq in the chain before the console and multitrack. Record the sound precisely as you want it. Otherwise, you'd have to add another level of D/A and A/D conversion (between the MDMs and the console) to make use of that vintage gear.

A good rule of thumb is, once a signal has entered the digital domain, it should stay there until someone unwraps your CD and hits 'play'. But for every rule there are exceptions. You may want to insert an analog stereo compressor between the digital console's output and the DAT or CD burner. (If so, it's advisable to invest in a high-end, dual D/A A/D converter.)

Some engineers like to bounce (copy) the final, stereo mix from DAT to an analog 2-track recorder (e.g. 1/4" tape at 15 ips) for added warmth, punchier bottom-end, or natural tape compression. Others feel it's not worth the extra signal degradation. Who is right? You are.

Studio Buddy

How can I make sure everything is perfect?

As any pro engineer would tell you, "Don't even try to make everything perfect." Instead, why not learn to be an imperfectionist. Learn to leave some mistakes in. Plenty of great records were made live to 2-track, with no chance to fix things. Plenty of hit records have mistakes in them. If it worked for the Kingsmen on "Louie Louie," it can work for you. Life is never perfect, why should the songs that make up life's soundtrack be perfect? It's not that you shouldn't make the extra effort to be sure your recording and mix is everything you want it to be. But if you obsess about perfection, you run a couple of risks.

In the recording stage, you might erase a take by the band that had so many great qualities - passion, groove, energy, magic - just because of one tiny flaw in the bridge. Or maybe it's not such a small issue. What do you do if the tempo picks up slightly at the chorus? Do you stop recording mid-take in the name of consistency? Do you record over that take upon noticing this "flaw" on playback? Maybe you ask yourself a couple questions first, like "Did the whole band pick up the tempo or just the drummer?" and "Does the slight tempo change feel natural and exciting or unrehearsed and amateurish?"

The second risk in being a perfectionist appears in the mix process. If you're thinking about things too hard, for too long, you can take an originally lively performance and tweak it into a sterile, boring snoozefest. Since so much of mixing is thinking about details and subtle increments, it's easy to lose sight of the big picture. Is it still a compelling song? Does it make me sit up in my chair? Part of developing as an engineer is learning when to leave it alone.

Studio Buddy

What is Ômastering'?

The mastering process allows you to perform final adjustments after you have mixed your multitrack recordings down to two stereo tracks (we'll leave quad and 5.1 surround-sound scenarios for another day.) Some adjustments are made to improve a particular song's sonic quality. Others are made within the context of an album - ensuring that many songs strung together have a similar sonic "consistency." Typical areas of concern for a mastering engineer are: equalization (eq), compression, levels (volume) relative from one song to the next, and spacing between songs.

Equalization: Sometimes you'll want to adjust the eq or compression on a mix after you've done the final mix. Or you may have ten songs mixed by three different engineers in five different studios. Each song's eq may seem perfect by itself, but if you sequence them together, suddenly one song sounds too bright (or too dull...). Adjusting the eq can even everything out. Tip #1: remember that any eq changes to your stereo mix affect the whole mix - if you want to cut 3 db at 80Hz because your mix sounds muddy, remember to check how that affects all the instruments (e.g. the vocal), not just the bass guitar and kick drum. Tip #2: if you're unsure about an eq decision during mixdown, know that it's easier to cut lower frequencies in mastering than to boost them, and easier to boost higher frequencies than to cut them.

Compression: In mastering, this is used not just to control a mix or to add character, but also to "print" or send as much level to the master as possible without clipping the signal. This can almost feel like a competition for who has the loudest cd ("my record sounded great until I listened on my CD carousel and Green Day was 5 db louder!"). But mastering engineers must balance level with sonic integrity.

Levels: Ideally, a listener can play your record and not have to get up to adjust the volume. This is addressed in mastering, after the record has been sequenced. Only then can you really know how levels relate to each other as one song ends and the next begins.

Spacing: there are different philosophies as to how one should approach the spaces put in between songs on a record. Some feel the downbeat of one song should fall at the start of a new bar, in the tempo of the previous song (to continue the flow.) Others think you should avoid this like the plague, because it diminishes the impact. In the end, do whatever feels right. There is no standard. Crossfade your songs if you like, or place six seconds between them. (2-4 seconds is common in most popular, non-classical records, but it's up to you.)

Final tip: you may be inclined to master the same recordings that you mixed, whether it be for financial reasons, creative reasons, or merely because you can. But I strongly recommend that you get someone else to master your project. The objectivity and fresh ears they bring to the table invariably result in a stronger, more cohesive album.

What is essential gear for a home studio?

Nowadays, quality gear often seen in professional studios is affordable enough to buy for your home. Going on the assumption that you already own a guitar or keyboard with which to write, these are the basic necessities of a decent home studio.

Let's start at the beginning: The microphone. It's no accident that the Shure SM57 is the most popular and enduring of them all. It's thoroughly reliable and relatively transparent. Use it on vocals, drums, guitars, anything. For a basic dynamic mic, that's the one. If you want to invest more, get a large-diaphragm condenser. It will add warmth, presence and even excitement to your sound; AKG and Neumann, among others, are known for their ability to fashion their legendary technology into affordable all-purpose mics.

The next step in the chain is the microphone preamplifier, or mic-pre. All consoles have these already built-in, so why should you buy a separate one? Since "outboard" mic-pre's have only one dedicated purpose, they typically perform their job much better than the one in your console. Try to get one that has two channels, so you can use it for drum overheads, as well as guitars, keyboards, vocals, etc. If you're recording to a digital medium, make sure to get a mic-pre with tubes inside to help add warmth and counter any inherent "coldness."

This brings me to the recorder, most likely either a computer disk or a tape-based modular digital multitrack (MDM.) I highly recommend recording your signal as 'hot' (loud) as possible, because you won't get the best (digital) resolution unless you utilize all the bits available on each track. Higher resolution means fatter, bigger sounding results.

The way to control levels to your recorder is with a compressor. This allows you to pack the most signal onto tape/disk without overdriving the audio chain and pinning the meters. Put it in your chain after the mic-pre and twist each knob, training your ears as to what happens when you raise the threshold, adjust the attack, etc. Too much compression will remove the dynamics of the signal, so keep tweaking until you like what you hear.You'll want it to be a stereo compressor so you can use it again when you mix. Put it on your stereo buss, e.g. between the output of the console and the DAT player. If you've been mixing your songs to a cassette deck, I suggest upgrading to DAT or CD-R. Not only do you get bigger dynamic range and truer frequency response, but subsequent dubs (copies) will be cleaner this way.

But let's backtrack and look at your options for consoles & recorders. The most affordable way to go is still the 4- or 8-track machines which record onto analog cassette. The next step up would be a digital recorder that houses a mixer, 64 or more virtual tracks of recording, and well-designed effects such as reverb, delay, chorus, etc. This means you don't have to drop more dollars on effects processors.

If you're in a position to spend a few thousand on your console/recorder setup, you first have to choose between tape or disk. If interfacing with computers is perfectly natural to you, then go with a sophisticated system like ProTools, Digital Performer, Cakewalk, etc. But if you're not completely computer literate, you should consider tape-based recording instead. I like the combination of analog/digital consoles with MDM's. If you step up to this level of gear, you'll be putting yourself on the same playing field as professional engineers who record many of the albums you buy.

Other essentials include near-field monitors and an amp (either a dedicated power amp or regular stereo receiver.) You don't have to go crazy buying fancy cables for everything, but I do suggest investing $100 or so in 2 instrument cables and 2 mic cables to put in your signal chain from microphone/instrument to tape/disk. Also, make sure you have a decent tuner.

I deliberately left out certain areas of recording, such as equalization, drum machines and stomp boxes, because you can get a good recording of your songs without them. More important than any of the gear I've mentioned is your ability to develop and trust your ears. You can do this by regularly comparing your recordings with CD's you like. After you've mixed the song, ask yourself: Can I hear the melody? Can I make out the words? Is the hiss louder than the music? With a little investment in time and money, your songs can keep the impact they have when you play them live.

Studio Buddy

How much should I spend on an all-purpose mic?

While it's possible to get decent sounds out of cheap microphones (such as the Shure SM-57, which costs about $100) this is one item that's truly worthy of a little investment. The microphone, in fact, is the single most important link in your chain after the source of the music (voice, guitar, kazoo, whatever). You shouldn't have to spend over $1000 to get a brilliant mic. For that price, the classic AKG 414 is only rivaled by Neumann's TLM103 in my book. If that's a little too steep, try AKG's C3000B (about $350) or one of Rode's condenser mics like the NTV ($800) or NT2 ($400.) A large-diaphragm condenser makes for a good all-purpose mic because its low self-noise will keep quiet sounds quiet and its internal design allows for higher signal processing levels. You could get a tube-design mic (the good ones are usually quite expensive) but they're more sensitive to loud signals - don't put one right up against the beater of a kick drum - and therefore less "all-purpose."

Studio Buddy

Should I record to hard-disk or tape?

Whether you're building-up your studio from scratch or just thinking about upgrading your recording gear, it's essential to choose the type of multitrack recorder that fits your needs and your abilities. They fall into two basic categories:

Hard Disk: The trend towards hard-disk, whether it be compact, all-in-one digital workstations, computer-based setups like Pro Tools, or standalone hard disk multitracks like the TASCAM MX-2424, reflects the popularity of features from random-access to an integrated environment (with tracks, effects and automation all in one place.) Recording to hard disk, however, requires a certain familiarity with computers. It's not hard to lose an overdub, a mix, or a whole song if you're not careful about saving and backing up your data.

Tape-based: If that scenario sounds a bit daunting, then a tape-based system may be the way to go. From inexpensive 4-track cassette recorders to 32-track MDM setups, you'll find that it's much more difficult to accidentally erase an entire song. The downside to this option is that you need a little more patience as you move to different points in the song (or from one song to the next.) Mix automation is still an option, but it requires additional gear, such as using a digital console with your MDMs.

Analog, tape-based recorders are famous for their warm sound. Digital, hard-disk systems are loved for their non-linear editing capabilities. Digital tape systems are inexpensive, easy to set -up, and easy to use. It's really a matter of your specific needs and personal preference.

Studio Buddy

Do I need two sets of monitors in my studio?

Professional studios typically feature both near-field monitors and much larger "Main" monitors, mounted high in the wall or on stands. Home/Project studios usually cannot afford such a luxury, due to both cost and space issues. But is it such a luxury to have a second set of monitors? The answer is Ôno' and Ôyes'.

No - Pro engineers spend 80-90% of their time using the near-fields. The two purposes of the "mains" are to check the bottom end of your mix (it's difficult to tell just how much air you're pushing with near-fields,) and to impress the client (artists and A&R-types want to get excited by the final mix and like to "turn it up 'til their ears bleed.") You can definitely get by without super-sized monitors if they're not feasible for your studio. But...

Yes - You should absolutely have an alternate set of monitors to complement your near-fields. A perfect mix will sound good in all different types of speakers out in the real world, and the engineer should have a clue how the mix "translates" in different settings. You should play a dub (copy) in the car as part of the mix process, but having two places to check in the studio is useful and efficient. Your alternate monitors should have different characteristics than your near-fields, whether they're wall-mounted loudspeakers with 26" woofers or a cheap boombox.

Studio Buddy

Why do I need a collaborator or producer?

The best reason for having a collaborator in the studio is perspective. If you choose to do everything yourself, from writing and performing to producing, recording and mixing, it's difficult to know what's not working and what might be improved upon. No matter what area you choose to collaborate on, your work will almost always benefit from having someone there whose opinion you trust.

A typical example is vocal performance. A producer can recognize if you need to try it again and guide you through phrasing, pitch, emotion, etc. A good producer will also know to stop you when you've nailed it.

There are so many scenarios where it's nice to have a partner in the studio - arriving at the ideal guitar sound, finding the right tempo, choosing the sweetest harmonies. Your band might be convinced which three songs out of ten should be recorded for a demo. If you ask someone you trust down to rehearsal, their objective ear can bring a fresh perspective on which songs will work best.

Of course, the best reason to have a collaborator is that at the end of the project, if you're not happy with the outcome, you have someone to blame!

Studio Buddy

How do I jump from Ôdemo' recordings to Ômaster quality' recordings?

There are several issues involved in determining whether a recording is considered "master quality" (also known as "broadcast quality") or "demo quality." There's a persistent myth that you need 24 or 48 tracks in order to create a M.Q. recording, but it really has nothing to do with the number of tracks. It's more about how well you use the equipment you've got. A pro engineer will make a better recording with a 4-track Portastudio than an inexperienced engineer will make with the equipment found in a $2,000,000, world-class studio.

The first factor is that it be clean, which generally means two things - no substantial hiss and no unintended distortion. Hiss is a function of analog recording. You combat this by setting your record levels as hot as possible without causing distortion, but it's an inherent problem for users of low-priced, analog cassette multitracks. While they're terrific for demos, such machines are a major uphill battle if your goal is master quality. Distortion can be a wonderful effect when used on purpose - it helps to create a sense of edge or aggression. But unintended distortion - as a result of too much level hitting a microphone, or setting your record levels too hot - is a fast indication of demo status (meaning that film/TV supervisors and record labels would need to re-record it before using it in a scene or releasing it to the public.)

The second factor is instrumentation. Say, for example, you've written a song that calls for a horn chart in the bridge. If you have the money, you can hire a horn section to play the parts. If not, you can use a synthesizer to demonstrate what the part should sound like. Of course, with modern technology, there's a third option. Certain samplers will give you such a realistic sound for horns, strings, etc. that most people can't tell the difference. They can get pricey, but it's still cheaper than hiring the string quartet every time you've got new material. This is one of the major reasons why composers are now able to create master quality recordings at home.

A related area is the drum machine. If you intended for a real drummer to play on the song you've written, but circumstance requires that you use a machine, then make sure you know how to program it to sound as human as possible. Recent programs like Acid have made this easier than ever, but it still requires you to invest your time and money. Naturally, this point doesn't apply to all genres of music. In much of Electronica, for example, the drums are meant to sound cold and machine-like. But if that's not your aim, make sure your drums sound warm, fluid and human. (Tip: don't use the Ôquantize' feature, it'll just add to the stiffness.) Until you've mastered the art of drum programming, you'll be hard pressed to create master quality recordings with a drum machine.

The final issue that comes into play when evaluating the sound quality is the listener's gut feeling. Can they picture the music as a record in a store, or matched to pictures in a film or TV show? Or do they picture the artist in their bedroom, still learning their gear. While you can't control their gut feeling, you can do these two concrete things: practice aiming mics and setting levels to find the fine line between hiss and distortion;

and make sure you've invested in recording gear that gives you realistic instrumentation.

Studio Buddy

What does "buss" mean?

"Buss" can be used as a noun or a verb. Here's a very grassroots explanation of each use:

When I first started out as an engineer, we didn't use the word "buss," we used the word "route." For instance, "Would you like me to route the signal from input #4 to tape machine track #9?"

"Buss" used as a noun might be a reference to the stereo "buss." That simply means the stereo output of the console. It can also refer to the output of signal from an input (a.k.a. channel) to a track or other channel. As in, "select buss 8 on channel 1 to go to track 16."

But, I've always found that if you use words like "buss" with great authority, your clients will be confused enough to think you know what the hell you're doing!

Studio Buddy

What is a safety?}

A safety is a copy of the original recording. It can be a copy of the multi-track or the 2-track mix. Any pro engineer will tell you that you must always have a safety of both. Tapes get lost or damaged, and hard drives crash. Take the time to cover your butt. The day will most certainly come when you'll be glad you took the time to do the prudent and professional thing.

Studio Buddy

Should I do the work on my main hard disk or should I buy a separate one for the songs?

The conventional wisdom is that you need a separate hard disk, but that's sort of old thinking now that hard disks deliver data so quickly. Plus it would be a waste of all that capacity modern disks have - nobody needs 40GB of space for programs.

[Jargon alert:]

However, it is probably a good idea to partition the drive into two: a smaller partition for the system and programs, and the bulk for audio files. This is just so you can optimize the audio partition without bothering the other partition.

[And jargon explained:]

When you partition a drive, it appears onscreen and behaves like two drives (assuming you've created two partitions) even though it's just one. This is done with the drive formatting software; formatting and creating partitions erases the drive's contents, so you have to save everything elsewhere if you're not partitioning a brand new drive.

Optimizing: especially when you're punching in a lot, the computer writes data all over the hard drive. If it runs out of continuous space in one sector, it drops a "pointer" and continues writing the data somewhere else. When this happens a lot the drive becomes fragmented and its performance suffers.

Optimizing utility software rewrites all the data on the drive so it's in continuous files. You always want to have everything backed up before optimizing any drive, because it involves rewriting the data. While good optimizing programs check to make sure they're not causing any errors, this is an intrinsically risky business.

If you're using any copy-protected software, be sure to check whether optimizing will cause you to lose installs. Modern copy-protection schemes don't, but some earlier floppy disc-based ones do. If so, be sure to de-install these programs before optimizing.

Studio Buddy

Which is the best back-up medium for my songs?

"Best" depends on four factors: cost, capacity, speed, and reliability. Pick any two or three (although the first three have been improving at a screamingly fast pace over the past few years). Also, the cost can be figured in price per MB of storage or the price of the drive; manufacturers have a tendency to tout the former to justify the latter.

Here are some of the options. And please remember that technology changes so quickly today, that what I tell you today could, or should I say, will be out of date very soon.

Tape (various formats, including data DAT) Ð These drives tend to be relatively expensive but have a low cost per MB. The media often have a high capacity (like up to 30MB per cartridge/tape/whatever) and aren't expensive, but sometimes they're proprietary. Tape drives aren't very fast, but you can leave them overnight; due to their high capacity, they generally don't need to be attended.

But the drives have a historical tendency to wear out, and there's no guarantee that a replacement drive's heads will line up perfectly with your old tapes. And that assumes your model or even format is still made. Finding out that your past few years' work is gone can really ruin a perfectly good day.

The other thing about tape formats is that they don't just appear on your computer like a plug-and-play drive - you must have their software installed. And it's usually necessary to choose what you want to restore from some kind of list rather than in the standard way. Tape back-up formats tend to be best when you only need to get at their contents when something else breaks, rather than for convenient storage.

With those drawbacks duly noted, tape drives are used all over the professional world because of their advantages.

CD-R (and DVD formats) Ð How much storage capacity do you actually need? At about 7.5MB per track minute at 24-bits/44.1 or 48Khz, a 3-1/2 minute 24-track song with all guns blazing from beginning to end is going to take about 650MB. And in the real world, tracks aren't full from beginning to end, making the fit even more comfortable.

Coincidentally, that's just under the capacity of a 35¢ CD-R. CD recorders can write Cd's reliably at eight or more times real time, the media are pretty durable, and you'll be able to read CD-ROMs anywhere for years to come. You can also create audio CD masters with these drives.

This is the clearly the most sensible back-up and archival format for most musicians who aren't dealing with really high volume.

Removable cartridge hard drive (Omega Jazz, Caseload Orb) Ð These drives are great for temporary storage and transportation, but not reliable enough for long-term archival.

Regular hard drives (really!) Ð As of this writing, fixed hard drives are about the same cost per MB as tape. Why not just use regular hard drives in hot-swap bays? The idea isn't total lunacy, and it's certainly the most convenient.

A couple more points. First, there are obvious differences between back-up, archival, and working storage (i.e. the hard drives you're working on); each can require a different solution. It's always a good idea to maintain current back-ups of all three, because things do happen.

And second, back-up software such as Dante Retrospect requires you to have the software installed in the computer to see what it's written. That's fine as long as you're using a computer that still supports it, but who knows what machines it will run on ten years from now. So long-term storage really should be done using the Windows or Mac file system. That's why CD-R is such a good back-up medium, despite its size limitation.

Studio Buddy

Are expensive MIDI interfaces really much better than cheap ones or do they do practically the s...

They do the same job, but they generally have more MIDI ins and outs and usually have some sync features built in.

If you only have one MIDI instrument, a 1x1 interface is fine. But then how do you hook up a second MIDI instrument?

It could receive MIDI by connecting to the first instrument's MIDI Thru (sic), but it has to receive on different MIDI channels than the first one (if you want it to respond independently). And where do you plug its MIDI out? You can build a MIDI A/B switch fairly easily, and there are MIDI mergers on the market that combine the two datastreams. Add a third and fourth instrument and the hassles increase exponentially.

Most multiport interfaces feature MIDI merging of all their inputs, along with routing, data filtering, and other processing. Many can network multiple interfaces of the same type, which is very useful for large MIDI rigs. These interfaces are available in different sizes; as with most things, it's a good idea to leave room for expansion when you buy them.

It used to be that the Mark of the Unicorn MIDI Timepiece spec was the de facto industry standard; all manufacturer's multiport devices were capable of emulating it. However, software developers have figured out how to use their interfaces to improve MIDI timing (by time-stamping the data). Each developer has his own way of doing this with his own interfaces only, so things may be a little proprietary for a while.

Studio Buddy

How many tracks are available at the same time on a DAW?

A lot of tracks. Tracks are no longer the precious resource they used to be. Some software may limit the number to 24 or so, but 32+ track sessions have become normal.

The exact number of available voices - which is what most people mean when they say Ôtracks' - is a function of the computer's disk subsystem and to a lesser degree horsepower and memory. Edit density also has a lot to do with the number of voices you can squeeze out of a computer; the computer will run out of steam a lot earlier if you've edited every drum hit than if you're just playing back linear tracks.

Note that many DAWs will allow you to record many more tracks than they have available voices to play back simultaneously. On most systems, tracks can be recorded or loaded and armed for playback without taking up voices. These "virtual tracks" are useful for alternate takes, for example.

The other side of the coin is that many DAWs are billed as providing some absurd number of tracks - "up to 256" or "unlimited." That means very little in real life.

As of this writing, a lot of questions come up about hard drive subsystems. Which hard drive protocol (SCSI and its flavors, IDE, EIDE, Ultra ATA/100, etc.) is necessary? How many RPM should the drive have? Do I need to install a [insert a few letters] high-performance card if I want to get xx number of tracks?

The answer to these questions can be reduced to two simple answers: first, ask the manufacturer of the audio hardware you're using what configurations they recommend; and second, try the stock hard drive that comes with your computer and see how well it works. You can always add on later.

Studio Buddy

Are third party plug-ins available and reliable for DAW systems?

Oh yes. It's a cottage industry that's really grown over the past few years, and there are some amazing ones around. Today's digital audio sequencers all come with a huge collection of plug-ins; one of these programs and an off-the-shelf computer with decent audio I/O hardware is all you need to put together a very respectable production rig.

Having said that, the quality of plug-ins is variable. In general, most of the stock plug-ins that come with digital audio sequencers are average - with some notable exceptions in every bunch.

But the initial rush to slap the name and graphics of a high-end processor on any old bunch of DSP code has died down considerably. Plug-in companies that have been around for a while are all producing high quality processors that you can trust; if they weren't, they wouldn't still be around.

To produce something that sounds like a record, though, you'll want to use as high quality an outboard recording chain as possible: mic, mic preamp, and probably compressor. Many plug-in compressors sound very good these days, but it's better to use an analog compressor in front of your DAW's A/D (analog-to-digital) converters (so it can't overload them).

Finally, the inherent danger with plug-ins is that with so many sitting in your computer, the tendency is to feel that they must all be put to use. That's probably not a good idea!

Studio Buddy

How do I transfer a loop from a loop CD into the tracks of the song I am working on?

It's really simple. You need to use a program that extracts the audio from the CD and writes it to your hard drive in a format your digital audio sequencer understands - and they all read the common formats (.wav, Sound Designer II, AIFF). Then you just import the audio into your session and stick it in a track.

The process of extracting the audio is called ripping, and the chance that you don't already have a program that can do it on your hard drive are very small. It's built into QuickTime, for instance, and QuickTime is installed on every Mac and can be downloaded for Windows. Or you can just search the internet for shareware. It's everywhere.

Studio Buddy

Can effects and eq parameters be automated on a DAW?

In most software, yes, although some programs make it easier and better than others as of this writing. Digidesign Pro Tools and MOTU Digital Performer both do this very well - you enable the parameters you want to automate, press Play, and move knobs and faders. Each parameter is then editable.

The digital audio sequencing programs that don't have sophisticated automated mixing of all parameters (including eq and effects) are hustling to implement it. Automated mixing opens up a lot of creative possibilities. Most of Digital Performer's effects include in-tempo parameters, for instance.

Studio Buddy

How do I get rid of clicks when I record digitally?

Whenever you transfer (i.e. record) audio digitally between devices, both of them must be clocked from the same source. This is because no two crystals on the planet, which all digital devices use as timing clocks, oscillate at exactly the same frequency. When they're not clocked together you get clicks.

The simplest set-up is to make the sending device the master and have the receiving device slave to the clock embedded in the digital datastream. Better yet in most cases, clock both devices from an external word clock generator.

Why? Because analog-to-digital and v.v. converters are subject to jitter, which is minuscule variances in the timing clock. External word clock-generating devices tend to have more stable clocks. But this isn't always the case, so you have to experiment with your rig to find out which clock source makes it sound the best.

Note that only converters are affected by jitter - digital transfers between machines either succeed or they fail (in which case you'll be hearing the clicks that prompted you to read this discussion). So you may hear it while monitoring a digital transfer between two machines, but it won't be present on a CD you burn from the actual audio data.

If your digital sync set-up seems correct and you're still hearing clicks, check that both devices are set to the same sample rate; some devices get confused when what they're seeing isn't coming in at the sample rate they're expecting. Another common source of problems is when you want to monitor a DAT or CD recorder connected to the digital 2-track return of a digital mixer - some boards aren't smart enough to know they must change their clock source to the incoming digital datastream when you switch over to the 2-track monitor.

Studio Buddy

There is a noticeable delay when I record digitally between striking the note and hearing it fro...

This is called latency, and it comes in two main varieties.

The first is the latency in and out of the digital converters themselves, which is on the order of 3 milliseconds at today's 44.1khz and 48khz sampling rates (half that at 96khz). That's 3/1000 of a second, about the amount of time it takes sound to travel three feet.

Most mortals aren't bothered by this small amount of latency in and of itself, although some experienced session drummers reportedly don't like to track through digital consoles because of it. The point at which it can become a problem for many people is when the delayed signal they're hearing in their headphones combines with the live sound to cause a comb filtering effect (certain frequencies are canceled out, resulting in a phasey/flanging sound).

There's not much to do about this problem other than to take one ear off the headphones and/or lower the musician's level in his or her headphone mix. Or track through an analog board, which won't have this problem. DAWs that supply their own processing power on PCI cards and just use the computer as a sophisticated front end (e.g. Digidesign Pro Tools, Ensoniq Paris) are subject to this kind of latency but not the second kind.

What you're probably complaining about, though, is the second variety of latency. This kind affect host-based DAWs, which are DAWs that use the computer's processor(s) for horsepower. Here the latency is caused mainly by the computer's operating system not being designed for real-time applications, and it's in the 7 Ð 30 millisecond range. Seven milliseconds is fine for most musicians under most circumstances, and 30 is pretty hard to deal with.

Running live input through DSP plug-ins on the DAW will also result in latency; there's nothing wrong with using aux send-type plug-ins (reverbs for example), but you might not want to track through a plug-in compressor or reverb if this is a problem. And you certainly don't want to use these plug-ins on one side of a stereo pair only.

There are a few things that can help with host latency. Again, one is to track through an analog board. Another is to lower the audio interface's RAM buffer settings as much as possible; adjusting these settings puts performance and latency on opposite ends of a scale. Use too low a setting and the audio will stutter; at too high a setting the latency becomes unbearable.

Finally, the ASIO 2 spec allows for direct loopback monitoring through the digital converters (with levels and pan remembered). This is a similar concept to monitoring off the record head on a tape recorder. ASIO 2 support is being added to programs as of this writing.

I am thinking of buying a DAW (Digital Audio Workstation) - should I go the Macintosh or Windows...

Macintosh.

Just kidding. While many of us who work on Macs all the time are very attached to them, the truth is that there's lots of great music and audio software for both platforms. Both are used day in and day out by audio and music professionals around the world.

A couple of major digital audio sequencers (which are programs that record digital audio and MIDI side by side - the type of program musicians use the most) are only available for one platform, but most are available for both. MOTU's Digital Performer is Mac only, and Cakewalk Audio is PC only; the others - Steinberg Cubase Audio and Emagic Logic Audio come in Windows and Mac versions. There's no functional difference between the two versions (unless one revision happens to be slightly ahead of the other at any given time).

PCs are definitely less expensive at the entry level, but above that the playing field levels. Choose the software and audio hardware you want to run, then choose the computer platform. If you've already picked one platform over the other, then you may not want to switch just to make use of the other platform. It may be a better idea to buy a program like Virtual PC.

Studio Buddy

How often should I clean my gear?}

More often than you want to. It's easy to get lazy about things like dust and dirt when you're pouring your heart out into a song. But recording gear isn't like other investments. A car, for example, will still drive no matter how much dirt it collects. But even a little bit of accumulated dust can have a subtle effect on your sound.

For the tape heads, all you need are some cotton swabs and denatured alcohol. While it might seem like overkill, the smart move is to give the heads a once over before recording each day. (Digital multitracks that use tape, sometimes require a special cleaning procedure that you may want to have done professionally.) If your heads have even a little bit of debris on them, it can have a significant effect on your frequency response.

Keep some canned, compressed air around (found in photo and electronic stores) to periodically spray on your console and outboard gear. And depending on how much action your patchbay sees, you might want to burnish each hole once a year with a burnishing tool to keep the contacts from becoming intermittent because of oxidization.

Studio Buddy

How do I learn to become a producer?}

There is simply no better way to learn how to produce records than to look over the shoulders of top-notch producers. Unfortunately, most people don't have access to sessions where one would find the "masters" of the craft at work. Short of that, the best thing you can do is study their records. Listen carefully with headphones on. Make notes. Ask yourself questions like, "Why is the bass line playing half-time over the drum track?" "Why is there a second guitar part on the chorus only?" "Why is there a pre-chorus?"

To become a credible producer you must be extremely well-versed in several disciplines; engineering, songwriting, arranging, and psychology. There are a lot of people who call themselves "producers," but haven't mastered all the aforementioned disciplines. Unfortunately, they often get paid to do a half-assed job, or worse yet, destroy a project.

Short of learning from the "masters," the best thing you can do is become obsessive about listening to records, and analyze them until they can't be analyzed any more. Read every book you can get your hands on about the generic subject of production, take classes when you can find them, read books about each of the disciplines mentioned before, and finally, use guinea pigs. Just be sure to let your "victims" know that you're still learning, and don't try to position yourself as a "pro" until you've logged thousands of hours in the control room with several different types of acts.

Studio Buddy