Why Do My Recordings Sound Like Ass ?

Question:
What is the single biggest thing you can do to improve your recordings?

Answer:
Fix the weakest link.

Follow-up question:
Okay, wise-ass, what's the weakest link?

Answer:
Read on.

First, a bit of theory to set the tone:

"All you need is ears."

So said George Martin, legendary producer of the Beatles, among others. Regardless of whether you regard the man as the final authority on all things audio, his resume is worthy of respect, and the simplicity and contrarianism of this statement makes it worth a few moments of thought.

If you have more or less functional hearing, then you have everything you need to make the same evaluations that million-dollar producers do (in fact many of them have less functional hearing than you do, probably).

Your objective is simple: to make recordings that sound good. And regardless of the complexities along the road, you, as the creative mind behind the recordings, are the final arbiter of what sounds good. So all you have to do is fix it so that it sounds good to you.

There is this notion of "golden ears," of people with a super-magical ability to hear the difference between good and bad sound. The idea is that this this supernatural hearing is what makes their recordings so good. That is nonsense. If their hearing were so much better, then none of us would be able to detect how much better their recordings were. They make "golden recordings" that are still "golden" even to those of us with regular ears. If you cannot distinguish between good-sounding recordings and bad ones, then yes, you should give up, but that's not the case, because otherwise you wouldn't be reading this thread. You'd be perfectly happy with bad recordings.

The fact that you can tell the difference between good-sounding recordings and bad-sounding ones means that you have the necessary physiological attributes to get from A to B. Skills, experience, and learned techniques will speed up the process, but the slow slog through blind trial-and-error can still get you there if you keep your eyes on the prize of getting the sound from the speakers to match the sound in your mind's eye (or mind's ear, so to speak).

In other words, if it doesn't sound good, you have to fix it until it does. This is sometimes easier said than done, but it is always doable, as long as you are willing to turn down the faders, take ten deep breaths, and repeat out loud: "all you need is ears."

Following the above, and this is going to disappoint a lot of people, I'm afraid, we are going to start with the very un-glamorous back end of the recording chain.

Before you can do anything in the way of making polished recordings, you have to be able to trust your ears.

This cannot be over-stated. You must be able to trust what you hear, and only then can you start to make good decisions. This is partly a philosophical, state-of-mind thing, but it is also partly a practical matter. You need to be able to trust that what you hear in the control room (or in the spare bedroom you use for recording) is what is actually on the tape or the hard disk. And that means that you need to have at least a certain bare minimum of room acoustics and monitoring quality.

If there is one area in your studio to splurge on, it is monitors (aka speakers). I'm going to do a detailed buying guide later, but for now it is enough to say that the studio monitors are the the MOST important component. I would rather make a record in mono on a four-track recorder with a single decent monitor in a good room than try to make a record on a Neve console with a Bose surround-sound setup in a typical living room. And I'm not even kidding.

Passable monitors don't have to be all that expensive, and they don't have to be glorious-sounding speakers, they just have to be accurate. Let's talk for a moment on why home stereos often make bad monitors, even expensive or impressive-sounding home stereos:

The purpose of a studio reference monitor is to accurately render the playback material. The purpose of a good home stereo is to sound good. These goals are often at odds with one another, and a simple frequency chart does not answer the question.

A common trick among hifi speakers is a ported design that delivers what I call ONB, short for "one note bass." The speaker designer creates an enclosure designed to deliver a dramatic "thump" right around the frequency cutoff of the speaker. This gives an extended sense of low-end, and it gives a dramatic, focused, powerful-sounding bass that can be very enjoyable to listen to, but it is the kiss of death for reference monitoring. Every bass note is rendered like a kick drum, and the recordist cannot get an accurate sense of the level or tonality of the low-end. If you play back something mixed on a ONB system on a different stereo, the bass is all over the place, reappearing and disappearing, with no apparent consistency or logic to the level. This is especially acute when you play a record mixed on one ONB system back on a different ONB system. Notes and tones that were higher or lower than the cutoff of the other system either vanish or seem grossly out-of-proportion.

Another serious consideration is handing of the crossover frequency. On any enclosure with more than one driver (e.g. a tweeter and woofer), there is a particular frequency at which the two speakers "cross over," i.e. where one cuts off and the other picks up. The inherent distortion around this frequency range is arguably the most sensitive and delicate area of speaker design. Hifi speakers are very often designed to simply downplay the crossover frequency, or to smooth over it with deliberate distortions, and often manage to sound just fine for everyday listening. But glossing over what's really going on there is not good for reference monitoring. The fact that this often occurs in the most sensitive range of human hearing does not help matters.

Other common issues with home hifi systems are compromises made to expand the "sweet spot" by, for instance, broadening the overall dispersion of higher frequencies at the expense of creating localized distortions in certain directions, a general disregard for phase-dependent distortions that occur as a result of simultaneously producing multiple frequencies from a single driver, nonlinear response at different volume levels, as well as the more obvious and intuitive kinds of "hype" and "sizzle" that are built in to make speakers sound dramatic on the sales floor.

The important thing to understand is that none of the above necessarily produces a "bad sounding" speaker, and that the above kinds of distortions are common even among expensive, brand-name home theater systems. It's not that they sound cheap or muffled or tinny, it's just that they're not reliable enough to serve as reference-caliber studio monitors. In other words, the fact that everyone raves about how great your stereo sounds might actually be a clue that it is *not* a good monitor system.

In fact, high-end reference monitors often sound a little boring compared to razzle-dazzle hifi systems. What sets them apart is the forensic accuracy with which they reproduce sound at all playback levels, across all frequencies, and without compressing the dynamic range to "hype" the sound. On the contrary, the most important characteristic is not soaring highs and massive lows, but a broad, detailed, clinical midrange.

The two most common speakers used in the history of studio recording are certainly Yamaha NS10s and little single-driver Auratones. Neither one was especially good at lows or highs, and neither was a particularly expensive speaker in its day (both are now out of production and now command ridiculous prices on eBay). What they were good at was consistent, reproducible midrange and accurate dynamics.

Whether or not to use a subwoofer with monitors is a topic for another thread, but it's worth touching on here.

The main thing to be aware of is that reference-caliber subwoofer systems tend to be expensive and tend to require some significant setup, unlike a home-theater or trunk-mounted thump box. The second thing to be aware of is that subwoofers and very low frequencies in general are not always necessary or desirable for good recordings.

The old RIAA AES mechanical rule for vinyl was to cut at 47Hz and 12k, and some great recordings were made this way. Human perception at extreme highs and lows is not all that accurate or sensitive, and a little goes a long way. If you have accurate monitoring down to say 50 cycles or so, and you simply shelve off everything below that, then you are making recordings that will probably hold up very well in real-world playback on a broad range of systems. The real-world listeners who have the equipment and acoustics to accurately reproduce content below that, and who have the sensitivity to notice it and care are very few and far between.

If you do monitor with subs, make sure the record still sounds good without them.

The second part of trusting you hearing is having decent room acoustics in the listening room where you make decisions. This is the most commonly-overlooked aspect of home studios, and it affects everything, so it is worth putting a little effort into. You *CAN* treat a bedroom studio pretty easily and inexpensively, and the difference is anything but subtle.

There is a sticky at the top of this forum where I and others have said quite a bit already, so refer to that for details. (Hint: do NOT stick any acoustical foam or egg crate on the walls until you understand what you're doing).

The next most important thing, after trusting what you hear, is to trust your recording chain. This means mic>cable>preamp>converter>recording software (REAPER, presumably).

Notice that I said "trust" is the most important thing. That is, it is more important to trust it than to have it be a great one. If this seems counter-intuitive, it is. More time and money is wasted by home recordists second-guessing their gear and wondering whether the preamp or whatever is good enough than anything else. If these people simply trusted that what they had could work, and focused confidently on technique, they would achieve more in an hour towards improving their recordings than by spending months reading reviews and forums and how-to books.

So if you have any doubts about the ability of your gear to capture good recordings, try this test (suggested by the brilliant Ethan Winer in this month's Tape Op):

Take a great-sounding CD and record it through your soundcard. Play back the recording. If it still sounds great, then you know that your soundcard is capable of rendering great-sounding recordings. No more blaming the interface.*

Next take the same CD and play it back through your monitors, recording the playback with your favorite mic (this is actually how the earliest records were duplicated). Still sound good? No more blaming the mic, cable, or preamp. If it doesn't sound good, then go back to the above post and make sure that your monitors and room acoustics are up to snuff. Even the lowly SM57 should reproduce a pretty accurate picture of whatever you point it at.

If you cannot get a good capture with what you have, then it's time to try and wring out the signal chain for the weakest link. But since I suspect that most home recording rigs will more or less pass this test, I'm going to set that part aside for later.

*Please note that none of this is to say that preamps or converters or mics don't matter. Better tools make things easier. But merely adequate tools can still build a great project. The pyramids of Egypt, the Taj Mahal, Buckingham Palace, and John Hammond's brilliant recordings of the Benny Goodman Orchestra were all created without tools that modern craftsman take for granted.

The idea here is not to say that you never need to buy anything other than an Mbox and an SM57, on the contrary, upgrading the studio becomes a lifelong process for most of us.

The idea is rule out fruitless anxieties about the gear, and to focus on listening and good techniques, which are the most important things in any studio, at any budget. If you are not confident in the ability of the gear to render acceptable recording quality, then that doubt will hamstring everything you do, and will cloud your judgment every step of the way.

How to get golden ears in one easy step (seriously)

Level-match playback anytime you are making any kind of comparative decision. The world of making good audio decisions will become an open book. This is going to be a long post, but it's important. Bear with me.

"Level-matching" does NOT mean making it so that everything hits the peak meters at the same level. Digital metering has massacred the easiest and most basic element of audio engineering, and if you're using digital systems, you have to learn to ignore your meters, to a great degree (even as it is has now become critical to watch them to avoid overs).

Here's the thing-- louder sounds better. Always. Human hearing is extremely nonlinear, due to a thing called the "fletcher-munson effect." In short, the louder a sound is, the more sensitive we are to highs and lows. And as we all know from the "jazz" curve on stereo EQs, exaggerated highs and lows means a bigger, more dramatic, more detailed sound.

Speaker salesmen and advertising execs have known this trick for decades-- if you play back the exact same sound a couple dB louder, the audience will hear it as a more "hifi" version and will remember it better. This is why TV commercials are compressed to hell and so much louder than the programs. This is why record execs insist on compressed-to-hell masters that have no dynamics (this "loudness race" is actually self-defeating, but topic for another thread).

What this means for you, the recordist, is that it is essentially impossible to make critical A/B judgments unless you are hearing the material at the same apparent AVERAGE PLAYBACK VOLUME. It is very important to understand that AVERAGE PLAYBACK VOLUME is NOT the same as the peak level on your digital meters, and it absolutely does not mean just leaving the master volume knob set to one setting.

Forgive me for getting a little bit technical here, but this is really, really, important.

In digital recording, the golden rule is never to go over 0dBFS for even a nanosecond, because that produces digital clipping, which sounds nasty. Modern 24-bit digital recording delivers very clean, very linear sound at all reasonable recording levels* right up to the point where it overloads and then it sounds awful. So the critical metering point for digital recording is the instantaneous "peak" level. But these instantaneous "peaks" have almost nothing to do with how "loud" a thing sounds in terms of its average volume.

The old analog consoles did not use the "peak" level meters that we use in digital, and they did not work the same way. Analog recordings had to thread the needle between hiss on the low end, and a more gradual, more forgiving kind of saturation/distortion on the high end (which is actually very similar to how we hear). Peaks and short "overs" were not a big deal, and it was important to record strong signal to avoid dropping below the hissy noise floor. In fact, recording "hot" to tape could be used to achieve a very smooth, musical compression.

For these reasons, analog equipment tended to have adjustable "VU" meters that tracked an "average" signal level instead of instantaneous peaks. They were intended to track the average sound level as it would be perceived by human hearing. They could be calibrated to the actual signal voltage so that you could configure a system that was designed to have a certain amount of "headroom" above 0dB on the VU meter, based on the type of material and your own aesthetic preferences when it came to hiss vs "soft clipping."

In REAPER's meters, the solid, slower-moving "RMS" bar is similar to the old analog VU meters, but the critical, fast-moving "peak" indicator is something altogether different. If you record, for instance, a distorted Les Paul on track 1 so that it peaks at -6dB, and a clean Strat on track 2 so that it also peaks at -6dB, and you leave both faders at 0, then the spiky, dynamic Strat is going to play back sounding a lot quieter than the fatter, flatter Les Paul.

The clean strat has big, spiky instantaneous peaks that might be 20dB higher than the average sustained volume of the notes and chords, while the full, saturated Les Paul might only swing 6dB between the peak and average level. If these two instruments were playing onstage, the guitarists would adjust their amplifiers so that the average steady-state volume was about the same-- the clean Strat would sound punchier and also decay faster, the dirty Les Paul would sound fuller and have more sustain, but both would sound about the same AVERAGE VOLUME.

Not so when we set them both according to PEAK level. Now, we have to turn down the Strat to accommodate the big swings on the instantaneous peaks, while we can crank the fat Les Paul right up to the verge of constant clipping. This does not reflect the natural balance of sound that we would want in a real soundstage, it is artificially altered to fit the limits of digital recording.

*Note that, contrary to a lot of official instruction manuals, it is not always good practice to record digital right up to 0dBFS. Without getting too far off-topic, the reality is that the analog front-end is susceptible to saturation and distortion at high signal levels even if the digital recording medium can record clean signal right up to full scale. The practice of recording super-hot is one of the things that gives digital a reputation for sounding "harsh" and "brittle." Start a new thread if you want more info.

I broke this off because this is where it gets important.

Continuing the above example, if you compare a half-finished home recording to a commercial CD that has been professionally mixed and mastered, the the commercial CD is likely to be a lot more compressed, and is therefore going to play back at a much higher volume than your record in progress, unless you turn down the CD or turn up your recording.

It is not a fair comparison to listen to two sources unless they are at the same average level. See if this sounds familiar:

Joe Blow records some stuff. Doesn't sound as good as his favorite records, sounds a little dull. He adds some highs. Sounds better, but a little thin. Adds some lows, sounds a little better, but a little hollow. Adds some mids, sounds a little better, but still sounds kind of harsh. He adds some reverb, sounds a little better, but now he notices it's clipping. So he turns down the levels.

Now it sounds a little dull, so he adds some more highs. Better, but a little thin, so he adds some lows...

Repeat until 2am, go to bed, and wake up to find that the "improved" recording sounds like a vortex of shit.

Now replace every instance of "better" above with "louder" and see if you get the idea ;)

So now that we understand that it's important to compare sounds at consistent playback levels, and that simply adding more effects without adjusting playback for the additional signal level can be deceiving, the obvious question is: how loud to monitor?

For people of a technical bent, the first answer is 83dB SPL (but hold your guns). SPL means "sound pressure level," meaning the actual air pressure of the moving sound waves. There is no way to measure it in within reaper or any other software, you can only measure it in open air, after the sound has left the speakers.

83dB SPL is right about where human hearing is most linear. It is about as loud as city traffic, or a noisy restaurant. Alarm clocks are supposed to ring at 83dB. THX movie mixes are supposed to be calibrated with an average speech level of 83dB SPL, somewhat louder than typical conversation in a quiet room. 83dB sounds "loud," but not painful. OSHA requires no more than 8 hours continuous exposure to 83dB for workplace hearing safety, so it's right on the cusp of where you could spend a full work day without hearing damage. The legendary Bob Katz recommends that mastering engineers master music recordings at an average level of 83dB (actually, he recommends mastering at comfortable levels with a system calibrated to have a certain amount of fixed headroom above 83dB playback, but that's getting ahead of ourselves).

As it happens, 83dB is not only where hearing is most linear, it is also right about the average level where average listeners tend to set the playback volume when listening to music on a capable system. Just before "too loud." (what a coincidence!)

So, 83dB seems like an obvious level for monitoring, but not so fast, partner!

Remember what we said above, that louder always sounds better. We can make this rule work for us as well. As it happens, almost anything that sounds good quiet will sound even better loud, but the reverse is emphatically not true. Cranking up the playback speakers (or just adding more gain with piled-on effects) makes shitty mixes sound great. By the same token, turning something down makes it sound worse.

This effect is especially brutal on live recordings of metal and hard rock bands. When you're standing in the crowd, and hearing a roaring 110dB that shakes your bones and pierces your ears, the effect is massive. But when you record that sound and play it back at workplace-background levels, the huge guitar sounds like nasal fizz, the furious double-kick turns to mushy paper, the churning bass becomes clackety mud, and the screaming singer sounds wimpy and shrill. These kinds of acts require a lot of tricks and psycho-acoustical funny business to achieve the right effect of power and loudness WITHOUT the actual power and loudness (more later).

But the same principles apply to anything. If you want your recording to sound right to every listener, then you cannot rely on high-quality 83dB playback every time. Your records are (hopefully) going to be heard in noisy cars and bars, on crappy speakers at 50dB in shopping malls, and so on. Unless you want them to sound wimpy and limp, it is really important to make sure that they sound good even in worst-case scenarios, because that is often where they will be heard.

So there is a really good case to be made for monitoring at very quiet levels as much as possible. In fact, I think it is safe to say that a majority of commercial mix engineers do a majority of their work at conversation-level or below, occasionally turning up the volume to check the lows and the balances at higher playback volume.

Monitoring at quiet levels has another practical advantage. Even before we hit the levels of hearing damage, our ears get desensitized by loud sound. Listening to 83dB for extended periods is like being in bright sunlight-- it's hard to see when you walk indoors. Keeping the lights dim allows you to occasionally focus spotlights where you need to check detail without dulling your overall vision. So it is with sound.

If you can create recordings that sound good at very quiet playback levels on decent nearfield monitors, they are almost guaranteed to sound better or at least as good in any other circumstances, including headphones and louder systems. But of course, it's always easy to double-check by putting on some headphones or cranking the volume for a few seconds.

There are a lot of schools of thought, but if you haven't already done so, I would encourage you to try recording and mixing at very quiet levels, and see if you don't start making better decisions, and generally better recordings.

Having said all of the above, I will now contradict a good deal of it in a short follow-up. If you get in the practice of level-matching AB comparisons, and of monitoring at infuriatingly quiet volume levels, you will rapidly start to develop an ear for fletcher-munson effects, and taking these measures will become less necessary.

This is where the "golden ears" business starts to kick in. You ears are the same, your hearing is the same, but your perception becomes better-attuned to the effects. This happens fast, like learning to detect an out-of-tune instrument, but it requires a certain amount of careful, educated, practiced listening.

Recording, like any process that is both technical and creative, is a state-of-mind thing. Any single aspect of the process has the capability of being either a launching pad or a stumbling block to better records. Experience brings a sense of proportion and circumspect "big picture" awareness that is hard to get from reading web forums and eq recipes.

It is important to work fast. **Finished is always better than perfect.** Always. In more ways than one.

For one thing, you will change your mind about things as the recording develops. There are a thousand steps along the way, and if you get too stuck on one, you lose your inspiration and sense of proportion, you'll get frustrated and your ears will start to burn out, and you will start to hate the song and the sound. Recording it will start to feel like a chore and a burden and that state of mind will show in the finished product, if it ever gets to that state. More likely, the project will become a half-forgotten waste of hard disk space that never gets completed.

The best way to work fast is to take as much time as you need to *get ready* for recording, before you actually start the creative process.

This is actually a big problem with new clients in professional studios-- they show up late, with worn-out strings and drum heads, out-of-tune instruments in need of a setup, they're hungover (or already intoxicated), they only got four hours sleep and haven't rehearsed or even finished writing the material, and so on. This is frustrating but manageable for the engineer to deal it with, it simply means that the client is paying for a lot of wasted hours to restring their guitars and so on. The engineer can take care of the setup for the first day or two and then get on with the business of recording.

In a self-produced home studio setting, this approach is fatal. If you're trying to write the song, learn the part, demo plugins, set up your instruments, figure out your arrangements, and mix each part as you go, you will spend two years just tracking the first measure.*

So the next couple of posts are going to deal with methods and techniques designed to get you moving fast and making constant progress, and also with figuring out when you've stalled out. The whole idea is to keep the actual recording process a primarily creative and inspiration-driven one, and to separate, as much as possible, the technical aspects that a dedicated engineer would normally perform.

*Please note that are certain kinds of loop-based and sequenced/automated electronic music where sound design and stuff normally thought of as "production" is an integral part of the compositional/performance process. The same principles of efficiency apply to any kind of production, but they may apply a little differently if your core creative endeavor is built around selecting, mixing, and processing existing sounds, as distinguished from music that is created and performed from whole cloth on more conventional instruments.

The best way to make sure that you are always making forward progress while recording is to set specific and achievable goals for each session. In other words, if you have three hours to record tomorrow, decide in advance what the "deliverable" will be, as though you were answering to a boss.

For example, you're going to get the main rhythm guitar track for this song recorded all the way through in three hours, come hell or high water, even if it's only half as good as you hoped. This means no shopping for plugins, no second-guessing whether you need different pickups, no deciding that the bridge needs to be re-written, no surfing the web for guitar recording tips, no testing to see how it sounds with a new bassline, no trying out alternate tunings, etc.

If you need time to do any of the above before you can be sure you're ready to cut the rhythm guitar, well, then, THAT is your project for tomorrow. Instead of trying to record the guitar part, you've got three hours to decide on the best bridge arrangement, or to try out different plugins, or to test alternate tunings, or to research and test different setup recipes, or audition plugins, or whatever.

The whole point is that no matter how many things need to be done or tested or thought through or tried out, come the end of tomorrow's session, you will have absolutely and decisively crossed one or more of those steps off your list.

No sane person would ever deliberately decide that "I'm going to spend the next three months second-guessing the amp tone and the particular voicing of the palm-muted riffs on the second turnaround," but this is exactly the danger if you don't decide in advance how much time you're going to spend on these things. Boredom, ear-burnout, and self-doubt are your enemies.

In a commercial studio, you'd have the reassuring hand of an experienced engineer and/or producer to tell you when it sounds great, or when it's time to stop and re-examine that 7sus4 chord and so on. You don't have that. So you have to trust your prior decisions, and just as important, you have to trust your future decisions and your overall talent.

It's one thing to say "we'll fix it in the mix." That's bad. But it's another to say, "I know that this is a good song, and that I can play it, and that I've been happy with this sound before, and I know that everything is going to sound bigger and better and more polished and professional once I've laid down all the tracks and have processed and mixed the whole thing."

It's very easy to get trapped in self-doubting tunnel-vision. It's important to get it done right, but it's also important to get it done. You may not achieve every goal you set for yourself in the time alloted, but at least you'll reach a point where the clock runs out and you can set yourself a better goal for next time, armed with specific knowledge of what you need to work on.

Setting specific goals in advance hedges against dangers on both sides of this see-saw. You have the opportunity to set aside enough time to do it right, while simultaneously preventing yourself from getting lost in an open-ended vortex of trying to reinvent the wheel.

I'm going to step back for a minute here and make some general points about preparation and organization.

It is really important to have an organized studio. Set aside a day for this, and it will save you weeks in the coming year, not to mention immeasurable inspiration-killing frustration. You need to make it easy for yourself to be creative, and hard for yourself to get distracted.

Organized is a different thing from appearing tidy. Scoop up all your cables and tuners and notes and headphones and stuff them in a drawer and the room will appear tidy. And you will spend an hour of your next session untangling everything and finding what you need. Hide all your patch cables and tie them up in bundles behind the desk and things will appear tidy, and it will take you an hour to get behind there and patch in a "B" set of speakers or a new midi controller.

Organized means that the stuff that you need is easy to identify, easy to reach, and easy to do what you need to do with it. A well-organized studio might actually appear pretty messy, and if that's a problem with a significant other or some such, then you might need more than a day to figure out the right compromises. A studio is a workspace, like a garage or a woodworking shop.

There are three categories of stuff in your studio:

1. Stuff you need to access regularly, and that needs to be right at hand.
2. Stuff you only need to access rarely (a few times a year), that can be stored away.
3. Trash.

Notice that there is no category for stuff that might useful someday, or that you plan to work on when you have spare time. If it were useful, you'd have used it. If you had spare time, you'd already have worked on it. Here's a hint-- old magazines are trash. The useful wisdom in them is either already on the internet, or has been or will be published in book form for that day 3 years from now when you need to search for it. And when that day comes, the chances of your actually finding the article you needed in three years' worth of old magazines is nil. There is no Google for old magazines.

Bad cables are trash. If you're going to fix them, put them in a brown paper bag and do it this week. If the week goes by and you haven't fixed them, throw them away. Cables that crackle when touched, or that hum, or hiss, or that have to be plugged in at a certain angle to work have no place in a recording studio. Same with broken instruments, broken headphones, obsolete electronics, old speakers and computers, and so on.

If you have trash that has value, put it in all in a box, and write a date on it by which time you will sell it. If that date goes by, and you have not sold it, take the box of stuff down to the Salvation Army or Goodwill and make someone's day. But make the decision that you are running a studio, not a junk shop. Which is more important, to eliminate the distractions and time-wasters that get in the way of your music, or to squeeze the few extra bucks from your old soundcard?

I know this thread might seem like it's getting away from "why your recordings sound like ass," but the little stuff matters. A lot. Organization makes for better recordings than preamps do. Seriously.

Go to the hardware store and buy the following (it's all cheap):

- Sturdy hooks that you can hang cables and headphones from. Pegboard, in-wall, over-door, whatever. Dedicated hooks for guitar cables, mic cables, patch cables, and computer cables.

- Rolls of colored electrical tape. From now on, every single cable in your studio will have one or more colored stripes on each connector. So when you see the mic over the snare has a red stripe and a white stripe, and you go look behind the desk or the soundcard, you will see a white stripe and a red stripe and you will know instantly where the other end of the cable is plugged in. Headphones should be similarly marked (assuming that you ever have more than one set of headphones in use at a time).

-Velcro cable ties. Every cable will also have a velcro cable tie affixed to it, so that you can easily coil up slack.

- Extra batteries. Every studio should buy batteries in 10- or 20- packs. You should never have to stop a session to look for batteries, or for a lack of batteries.

- No-residue painter's tape. This is very low-stick masking tape that you will use to label all kinds of stuff. Stick in on the console or your preamps and mark gain settings for different mics and instruments, stick on guitars and keyboards to mark the knob settings, stick it on drums to mark the mic locations, stick it on the floor to mark where the singer should stand in relation to the mic, whatever. Peel it off when you're done and no sticky residue.

- One or two universal wall-wart power adapters (the kind with multiple tips and switchable output voltage). A broken wall-wart is a bad reason to hold up inspiration, and having a spare handy makes troubleshooting a lot easier. Keep in mind that a replacement wall-wart has to have the same polarity, approximately the same output voltage, and AT LEAST the same current rating (either Amps A or milliamps mA) as the original. So splurge for the 1A/1,000mA one if they have it. If you're not sure what the above means, find out before experimenting.

Next, go to the guitar depot and buy the following:

- 5-10 sets of guitar strings of every gauge and type you are likely to record. This means 5 sets of acoustic strings, 5 sets of electric strings, and each type in both light and medium-gauge, assuming that you might be recording guitars set up for different string gauges (this includes friends or bandmates who may come over with guitars that haven't been re-strung for months. Make them pay for the strings, but have them. Charge them double or more what you paid, really). These strings are meant as backup insurance for the times when there is a string emergency, not necessarily to replace your existing string-replacement routine. So they can be the cheap discount ones. They only need to last through one session, and are there for the occasions when a guitar needs to be recorded that has dead strings. Watch for sales and stock up.

- 2 extra sets of bass strings, same idea.

- A ton of guitar picks, of every different shape, size, material, and texture. Go nuts. Don't skip the big felt picks for bass (although you can skip the expensive metal picks if you want-- they suck). You are going to put these all in a big bowl for all to enjoy, like peanuts or candy. Or better yet, in lots of little bowls, all over the studio. Changing picks is the cheapest, easiest, fastest, and most expressive way to alter the tone of a guitar, and it absolutely makes a difference. Just as important, holding up a session to look for a pick is the stupidest thing that has ever happened in a recording studio. Don't let it happen in yours. Make your studio a bountiful garden of guitar picks.

Drum heads are a bit trickier, especially if you ever record more than one set of drums. You might have to save up, but get at least one set of extra top heads for your best drums, starting with your most versatile snare. The whole idea is not to hold up a session over something that is a normal wear-and-tear part. The long-term goal should be to buy replacement heads not when the drum needs them, but when you've just replaced them from your existing stock of extras. Sad to say, it's also not a bad idea to keep your eyes peeled for deals on spare cymbals, especially if you have old ones or thin ones or if you record metal bands. (Again, this is stuff that you should make people pay for if they break, but it's better to have spares on hand than to stop a session).

If you commonly record stuff like banjo or mandolin, then splurge for an extra set of strings for these. If you record woodwinds on a semi-regular basis, then reeds are an obvious addition. Classical string instruments are trickier, but if you commonly record fiddle, then pick up some rosin and a cheap bow, just to keep the sessions moving.

One of the most important things any studio should have is an ingenious device known as **a pad of paper**.

You may already own one and not even know it. This should have a dedicated, permanent spot in easy reach of the mixing desk (please have extra pens to go with it). Your hip pocket is a great place. Its purpose is to record "to do" and "to buy" items as soon as you think of them. Even better if you can have separate ones for each. Its value will become immediately apparent.

The "to do" list is the place to write down things like "find best upright piano preset," or "create new template for recording DI-miked hybrid bass," or "find better way to edit drum loops," or "re-write bridge for song X" or whatever you think of that needs to be done while you are focused on the deliverable goal that we talked about above.

This pad should be different from the one that you use to write lyrics or recording notes, assuming you use one. The idea here is to have a dedicated place to write down the stuff that could otherwise become a distraction while recording, as well as a place where you can capture recording-related ideas as they come up, and set them aside for future consideration in the sober light of considered reflection.

It should also be a place to write down stuff you wish you had, or wish you knew more about, so that you can shop and research in a systemic way. If you find yourself fumbling around with the mixer and the soundcard trying to get enough headphone outs or trying to rig up an A/B monitor comparison, then write it down. You might be able to rig up a simple setup on a Saturday afternoon, or you might decide it's worth getting a cheap headphone amp or monitor matrix (Behringer probably has one of each for $30).

If you can't find the right drum sample or string patch, don't stop recording to look for a patch now, instead, get the tracks laid down with what you have and make a note to look for better samples tomorrow. Tomorrow, you might have a totally fresh perspective and realize that it's not the samples that were the problem, but the arrangement. Or it might turn out that after a good night's sleep and with fresh ears, it sounds just fine. Or maybe you do need to find better sounds. In any case, it will be a lot easier to keep the processes seperate, and to focus on the issue at hand. Your **pad of paper** makes everything possible.

Anything that distracts your time or attention should be written down. Don't try to solve it right now, instead set it down as a problem to look into in the future.

You need storage and furnishings for your studio. It should be stable and quiet. Things should neither be falling over nor rattling. This does not have to be expensive. Places like Ikea and office-supply stores sell sturdy computer desks that are just as good as dedicated-purpose "studio" desks.

You should play various loud bass tones and suss out your studio for rattles before you start recording. Do this periodically, since things loosen over time. Duct tape, wood glue, silicone caulk, and rags such as old T-shirts are useful for impromptu rattle-fixing.

I think the best studio desks in the long haul are probably just plain, sturdy tables. A big, open, versatile space tends to age better than a preciously-designed contraption with fixed racks and speaker stands and shelves and so on. It's easy to put those things either on top of or underneath a plain table, but it's hard to rearrange stuff that's permanently built in.

Avoid cheap chairs with lots of wheels and adjustments, they are apt to rattle and squeak. Plain wooden or even folding chairs are preferable. Herman Miller Aeron chairs are excellent studio chairs, kind of a de-facto standard, but they're expensive, and complicated knockoffs are sometimes worse than simple, silent hard chairs. Musicians often benefit from a simple bar-height stool without arms, for a half-sitting, half-standing position.

If you are on a tight budget and need racks, they are ridiculously easy to make. Just build a wooden box with sides 19" apart, and screw your gear into the sides. Road worthy? Probably not. But infinitely better than just having the stuff sitting in a pile that will inevitably get knocked over. You can even cut the front at an angle pretty easily if you are marginally competent. A quick sanding and coat of hardware-store varnish and it looks like actual furniture. Best part is you can build them to fit your spaces and put them wherever you want.

Keep your eyes peeled in discount stores for plastic toolboxes and drawer systems. The cheap soft-molded plastic stuff is a great place to store mics, cables, adapters, headphones, tuners, meters, CDs, and all that other stuff. Soft-molded plastic bins might be sticky and crooked to open, but they tend to rattle and resonate less than metal or wooden stuff, unless you are buying fairly expensive.

Unless you are going to forbid drinks in the studio, you should make space for them in places where people are likely to be. The floor is a bad place, but is vastly better than on top of keyboards, mixing consoles, or rack gear. I like little cocktail tables with felt floor sliders on the bottom. They are inexpensive and movable and having a few of them makes it easy to be a fascist about saying that drinks are not allowed on any other surface, ever.

Boom-type and/or gooseneck-type mic stands are a studio necessity, and are sadly expensive, for the stable ones. If you must use the cheap $30 tripod base, then understand that you are putting the life of your mic on the line every time you set it up. Budget accordingly. Do not put an expensive vintage mic on a cheap, flimsy stand. They all get knocked over, most sooner than later. The best deals are probably the heavy metal circular bases that are commonly used in schools and institutions. Plan on either putting them on a scrap of rug or on little sticky felt furniture sliders or something to deal with uneven floors, and to provide a modicum of decoupling.

Please own enough guitar stands to accommodate every guitar that will be in use in your studio. Guitars left leaning against anything other than a guitar stand invariably get knocked over, which screws up the tuning and endangers the instrument.

Bear with me, there is juicier stuff coming.

The key to organization is a place for everything and everything in its place. The PLACE FOR EVERYTHING bit is the most important.

In a well-organized tool shop, you'll likely see a pegboard with hooks and marker outlines of every tool. They'll have outlines of each hammer, drill, pliers, and so on. Hex drivers will be kept in a specific drawer, screwdriver bits are kept in a little canvas zipper-bag, nails and screws are organized by size in rookie kits or drawer boxes, and so on. Everyone knows where to find anything.

Your Mom's kitchen is probably similar. Plates in one cabinet, spices in another, pots and pans in another, tableware in this drawer, cooking spoons and spatulas in another, sharp knives in this place, canned goods in that, and so on.

The point with both of these is that it is obvious when a thing is in the wrong place. A wineglass does not go in the spice cabinet. Plates do not go in the knife drawer. Drill bits do not get hung in the hammer outline of the pegboard.

Your studio should be the same way. When you set out to organize it, and you don't know where to put a thing, stop. Your task is to decide where this thing goes, where it will always go, and where everything like it goes. "Everything goes in a drawer" is not an acceptable answer. You might have to buy or select a thing to put it in. But it is important to make a decision.

Knowing where to find a thing and knowing where to put it are the exact same question. If you don't know the answer to either one, then you have to get organized. Every adapter in your studio should be in the same place. Every wall-wart should be in the same place. Every battery should be in the same place. All kinds of tape should be in the same place. Spare drum keys should be in a specific place, as should guitar strings. All software should be stored in the same place, along with the passwords and serial numbers. Cables should be coiled and hung on hooks, according to type and length, so that you always know where to put it when you're done, and so that you always know where to get it when you need it. If I come to your studio and gift you a new piece of gear or ask to borrow a piece of gear, you should know exactly where it goes or comes from, without having to think about it, and before you decide whether to accept.

If you have a thing and really can't decide where it goes, put it in a box and mark a date on it one year from today. Put it aside. If a year goes by and you haven't opened the box, deal with it as trash, above.

The point is to keep the stuff you need ready and accessible. and this means getting rid of the stuff that's all tangled up with it. Your time in the studio should be spent on making music recordings, not on sorting through junk piles or looking for a working cable.

Okay, I apologize again for all the stuff on organization, but if I didn't get the boring bits out of the way first, then I'd never get to them once we start talking about sound. So now that we have space to work and to focus and think about the sound, and a setup that allows us to hear a good, accurate representation of what's going on with the sound, let's start to talk about sound.

There is a lot to say, and a lot to think about, and there's a big two-steps-forward-one-step-back element to all this, because everything affects everything. Principles of mixing apply to tracking, and principles of tracking apply to mastering, and principles of mastering apply to getting good sounds in the room to begin with, and principles of sound in the room apply to everything. So no matter where we start, there's a lot that comes before it, and a lot that comes after it.

That said, the most basic and critical element is critical listening and judgment. And one of the hardest notions for beginners to disabuse themselves of the value of recording "recipes" or presets. So that's the first thing I'm going to spend time on. And without a clear place to begin, I'm just going to start with my favorite instrument: electric bass.

Let's say, to keep things simple, that we're recording a DI bass track (i.e. a bass just plugged right into the soundcard or preamp, no mic). And let's say that the bass player is playing a bass with a maple neck and jazz-type pickups. And let's say she's using a pick, and that she does a pretty good job of controlling dynamics. Got all that? good.

So we fire up the recording rig and she starts playing. From here on, because this is a DI track, it doesn't actually matter whether we're talking about stuff we do during mixing or tracking, because we're going to pretend that none of this affects her headphone mix or how she plays (which is a whole nother can of worms). We have also, by virtue of recording DI, eliminated anything relating to mics and rooms and phase and any of that. There are also no chords to deal with, and presumably no intonation or tuning problems. We are also pretending that we have perfectly neutral "gain staging" and that it therefore doesn't matter whether we make these changes before or after tracking. Please note that these are actually HUGE assumptions that we will see later are NOT "safe bets" at all (even with sampled bass), but we have to start somewhere.

So she's kicking out her funky bassline and everything is groovy and we start to listen carefully, not just to the groove, but to the forensics of the sound. We're going to pretend for the sake of sanity that the player and the instrument are both good and there are no serious problems of fret buzz or strings clacking or serious flaws in the tone, and that the player is hitting about the right balance of warmth, string, and growl for the material (I just glossed over about a year of prep time on that one, but all in good time).

So we've got the sound under a microscope, soloed, and here are the little sonic microbes crawling around, the molecular structure of her bass sound:

-We have the initial, mostly atonal attack of the plucked string, which could sound like a lot of things, but since we stipulated a jazz-type bass with a maple neck and a pick, it's probably going to sound a little clicky, with a slight "rasp" or chunk, and have a little subsonic bump, like *un petit* kick drum. If we're really industrious, maybe we want to sweep an EQ around, and see if we can identify some particular frequencies where these things happen. Not making any changes, just "parking" eq nodes at the spots where these aspects of the sound seem to be exaggerated. Like maybe the click is up around 6~8k, maybe the raspy chunk hits a broad range somewhere around 700~1500Hz, maybe the subsonic thump seems most pronounced when we bump the eq at 40Hz. Maybe it's completely different. Truthfully, how she holds the pick and how close to the bridge she picks and what kind of pick she's using and a hundred other things will change all this. But that's okay, for now we're just listening, taking mental notes.

- Immediately following the attack, we have the steady-state "note." On a good maple-neck jazz bass, this is likely to to be a fairly deep and transparent sound, with a smidgen of low-end growl, a little "scooped" in the lower mids, and some good upper-midrange clarity, with a little bit of stringiness that we can use to add some bite and punch, or that we could downplay to mellow out the sound and push it back into the mix a little. Again, if we want to, we can sweep the parametric eq around and see where these elements are most pronounced. Not changing anything yet, just listening and thinking.

- Next we have the decay, where the sound starts to taper off. The best studio bass players are masters of this oft-overlooked corner of the musical world. A bass line played with every note ringing out until the next note gives a vastly different vibe and feel to the whole mix than a bassline where each note has a definite end point. Neither is necessarily better or worse, but how long the bass notes hold and how they taper off has a big effect on the way the drums and the beat breathes and pulses, and and it can "lock in" the whole band to make it sound like a unit, or it can create separation and clarity. This is not necessarily your call to make as the engineer, but being aware of how it affects the mix will help you to make better decisions. It might not hurt to give a careful listen to how the bass decays. Does the "growl" hold on longer than the note? Do the notes end with a little finger squeak or death rattle? Is the "note" the last part to die? These "last gasp" elements are all going to amplified if we end up compressing the signal, as the louder parts get pushed down and the quieter parts get pumped up ("IF we end up compressing ELECTRIC BASS?"-- that's a good one).

-Last but DEFINITELY not least is the "silence" between notes. This is the point at which the discernible sound of the bass falls below the noise floor. Because we are recording direct, we can pretend that there are no resonances to worry about, and we can stipulate that this should be dead silent. No hiss, no hum, no rumble, no radio signal, just pure audio black space. If it's not, we're going to have some serious problems. But that's a topic for another day.

So far, we've just been listening, not making any actual *judgments* about the sound, nor alterations. In fact, we already stipulated that the sound is pretty good. Let's take a look at how some of our observations above might relate to judgments and alterations that we could make to improve the sound of the bass, or the way it fits into the mix.

Starting from the beginning, let's take another gander at that pick attack. Let's say for the sake of argument that we have a fairly clean, snappy, telecaster playing on the guitar track. If we put this bass track beside it, then the pick clicking could start to be a problem. For one thing, it's competing with the clean guitar attacks, and potentially confusing the waters up there in the highs. If the two instruments are not plucked in absolute lock-step, then the bass clacking around is apt to screw up the syncopation and feel of the guitar part. And for a whole lot of good reasons, it is likely that a good bass player is NOT picking on exactly the same nanosecond as the guitar player, because the bass takes more time to develop, and because the has an important role to play in relation to the dynamic decay of the drums.

So maybe we want to back off that initial pick attack a little bit. Compression or fast limiting might help, but maybe we start to lose some definition that way. Maybe we're better off trying to nail it with eq. That lets us keep some of the slower, midrange chunky rasp that actually overlaps nicely with the guitar. As it turns out, turning down the highs a little might also solve some problems in the "steady-state" portion, where the stringyness might be similarly fighting the guitar.

On the other hand, let's say that the guitar is not a clean, snappy tele, but a roaring overdriven SG. Now we have a whole nother set of considerations. Here, that little ghostly "chunk" might be completely blown away by the guitar, and those clicky, stringy highs might be just the ticket to cut through that wall of power and give some bite and clarity to a bass sound that could otherwise get drowned into wub-wub.

Simply cranking up the highs on the bass might not be the best solution though, since these are fairly small elements of the sound, and are apt to turn brittle and fizzy if over-played. Compression or other dynamics control might offer some help, but here we start to run the risk of mucking up the whole sound of the bass just to try and get the string sound to cut through. This might be a good time to get creative, and try a little sansamp or guitar distortion to get that saturated harmonic bite. Or maybe it's time to plug into the crunchy API or tube preamp or whatever. But that might also change our nice, transparent low end in ways that we don't like (or maybe we do). Maybe we could split or clone the track with a high-pass filter, and just raunch up the highs a little to give the right "cut" to the sound.

Before we go much further, let's double back for a second. Notice that the whole post above is about dealing with one little aspect of the sound. And recall that where this element falls in the frequency spectrum and what proportion of the overall sound it comprises is entirely dependent upon factors such as: how the player holds the pick (or certainly whether she even uses a pick), how close to the bridge she picks the strings, the type of wood on the fretboard, and a ton of other stuff.

If the same player were playing a P-bass with the same technique, then the whole sound would be completely different. The chunk and growl would be much increased, and the clickly, stringy highs would be almost non-existant. Turning up the highs that help the Jazz bass cut through the SG might merely turn up hiss and fizz on a P-bass with a rosewood fingerboard. If she were fingerpicking or playing slap-style, the whole world would be different.

Now think for a moment about presets and "recipes." Even if they come from a world-class producer/engineer recording your absolute favorite bass player, what are the chances that every variable is going to line up exactly the same so that YOUR bass player, playing HER bass, with HER technique, in YOUR mix, of YOUR band, with all of the specific instruments and sounds, so that the settings and practices that worked best for one recording are going to be ideal for another? Is "rock bass" really a useful preset?

And just in case you think I've "gamed the system" by starting with the hardest part, think again. Life is about to get worse for bass presets. Read on...

I'm skipping right over the "thump" part of the bass attack, but that does not at all mean that you shouldn't think about how it might muddy up the all-important kick drum beat, or how it affects the sense of weight and definition of the bass guitar part, or how it interacts with the guitar and other instruments in terms of body and rythmic feel, or what kinds of effect it might have on your overall headroom in the track. I'm skipping over it because we have a lot of ground to cover, and there's always going to be stuff to double back to. And electric bass is just one example, and a DI recording of it is about the simplest thing we're likely to deal with in a project.

On to the "steady-state note" portion of the sound.

So maybe we made a few tweaks above to get the high-end definition right. The sound is still the good bass sound we had at the beginning, but we've done a little work to get the highs to sit better with our other instruments. So far so good. (please note that starting from the highs is not necessarily the recommended methodology for bass, it's just where I started posting)

So now we're listening to the bass, soloed (or not, whatever), and we start to focus again on our "steady state" sound-- the "average" sustained note portion off the sound. And it sounds good, but something doesn't quite "feel" right. The bassline sounds good, but just seems a little uneven, maybe a little jumpy. The "body" seems to waver in strength. We throw up the other faders, and sure enough, there it is, the plague of the recording world: **the disappearing/reappearing bass line.**

The bass just doesn't seem to articulate every note consistently. What should be a solid foundation of low-end tonality instead seems a little like a spongy, uneven house of sand. It's not precisely a "sound quality" problem-- the tone is there, the meter seems to show pretty consistent bouncing around the average, the picking is well-articulated and good, so what is it?

Well, because this is my example, I actually know the secret in this case, but I'm not going to tell you just yet. I'm not going to tell you, because there are a whole lot of things that can cause this symptom, and the cause is actually not all that important, or even that helpful when it comes to the practical reality of fixing the problem. The fact is that for a whole bunch of psycho-acoustical reasons and realities of the nature of the instrument, bass is prone to this syndrome. Bass notes are far further apart in wavelength than the notes of higher instruments, and broadband aspects of the "tone" of the instrument that would encompass a whole octave or more of high-frequency notes can disproportionately affect perception of individual notes, or ranges of notes, or certain harmonic relationships of notes, when it comes to bass instruments.

So let's take a closer listen to this bassline. Let's say that the bass player is bouncing around a range of about an octave or so, and the lower notes seem good, but the higher ones just seem to lose their tonality. You can still hear the string attack just fine, but the body drops out. And it's not that the foundation moves up in range, it just kind of lacks balls. So you try a compressor, and that helps a little, but the compression is getting pretty heavy and affecting the sound of the instrument. So you try sweeping some eq boost around where you think the problem might be. As it turns out, right about 100Hz works pretty good. But interestingly, a few ticks higher actually makes the problem worse.

So you settle on 100Hz, feed the boosted signal into some light compression, and now you're getting close to where you want to be. Cool, but what happened? Why did that work? Is it because 100Hz is a magic frequency for restoring consistent body to bass? NOT AT ALL.

For the secret, read on...

In this particular case, here are two things that I know and that you don't, that are the keys to understanding why 100Hz was the magic frequency. Before you read the explanation below, think about the following two facts and see if you can guess why a boost at 100Hz fixed the problem, but a boost at 110Hz made it worse:

-The song is a I-IV-V progression in D

-This particular bass guitar tends to sound notes on the "D" string quieter than notes on other strings (this is not *at all* uncommon, even on good basses)

(If you don't know how a bass guitar is strung or what a I-IV-V progression is, then don't hurt yourself, just skip ahead).

edit:
I realized after working it out that this was kind of a confusing example/trick question, so skip ahead before you dig out the slide rule.

Here's the key (literally and figuratively):

In the I-IV-V progression in D, the three most important notes are D,G,A.

On the bass guitar, the first position has prominent G and A notes on the D string. The frequency of the low G note on a bass (E string, 3rd fret) is around 48Hz. The frequency of the Low A note on a bass (E string, 5th fret, or open A) is 55Hz. So the frequencies of the first octave of these two notes (D string, 5th and 7th frets) are 96Hz and 110Hz, respectively. Those are the notes that are not sounding loud enough. If we boost at one frequency or the other, we not only boost that note, but the first harmonic of the lower-octave note of the same name, making the problem worse for the one we're not boosting. Boosting right in the middle of the two (technically, I guess a little higher, like 103Hz) gives a boost to G#/Ab (a note not played in D), and a little overlap boost to both notes, evening out the sound.

edit:
Reading this, I realize I made a little oversight that might confuse astute readers. Technically, I guess we might have trouble if the player also used the open D, especially if she alternated between the open D and closed D on the A string (time to dig out the multiband compressor).

So anyway, if the above puzzle gives you a headache, that should actually just hammer home the point that trying to think through this stuff is actually a lot harder than just listening. Moreover, there's no way to expect yourself to keep track of things like this and mentally cross-reference them.

All you need is ears. **If you can hear the problem, you can hear the fix.** The theory is not only unnecessary, it's not really even that helpful. I have never, ever, thought through an eq problem that way, and I doubt anyone else has either (the example was something that I figured out after the fact). And even if I did have a flash of insight and figured out what the cause was, I'd count myself clever and then STILL suss it out by ear.

But the real point of the above exercise was to illustrate the problem with presets. Whether you understand all the ins and outs of the breakdown or not, the real point is that the above fix *would not have worked on a bass that didn't depress the D string, nor for any song that was not in the same key.* Theory-minded bass players will recognize instantly that a boost of the second octave G# would be a serious problem for songs in the key of E, especially if the D string were NOT quieter than the others.

You can't just dial in a good bass sound and then use that for everything and expect to get the same effect. I can't go so far as to say that presets and recipes are *useless*, but I think there is more danger for the novice in over-reliance on them than there is in simply not using them at all. In some respects, the less you need them, the more useful they can be. The great danger is in trusting presets more than your ears, and sadly, I think that is often the norm among beginning home recordists these days.

So, having partially dissected a very simple DI recording, let's talk about microphones next.

There is no best microphone. There is no best mic in any given price range. There are some bad mics, but for the most, there are a lot of different mics. And frequency response is not a very important part of what makes a mic a good one or a bad one (at least, not within the realm of reasonable choices for studio recording). If frequency response were the ultimate measure, you could just use an eq to make an SM57 sound like a C12 and save yourself $15,000 or so.

And before we go any further, let's just clarify that there are times when an SM57 is actually preferable to a C12. In other words, there is no best mic. Any more than there is a "best ingredient." Spanish saffron is not necessarily much better than Nestle chocolate chips if you're making cookies. White truffles are great for veal, but not so much for lemonade. Whether you're better off using caviar or strawberry syrup might depend on whether you're serving toast points or ice cream (I always go with strawberry syrup, myself).

So it is with mics. And well-equipped professional studios that have access to all kinds of mics in all kinds of price ranges use a lot of different ones for different applications. Ask a dozen rock stars which mic they recorded their last vocal track with and you might get a dozen answers, and that's not because they don't know about or have access to the other mics.

It is a pretty safe bet that any well-known mic that costs over, say, $500 will be a pretty good mic, otherwise nobody would be paying for them. But there are also good mics that are inexpensive, and a more expensive mic does not automatically make it a better one for any given application. In fact the humble SM57 is probably the most widely-used microphone in the world, in professional applications.

Even if you're rich, a home studio is unlikely to have the same range of mics available as a professional recording studio, anymore than a rich person's kitchen is going to be as well-stocked as a professional chef's commercial kitchen. But that does not mean that homemade food is necessarily worse than professionally-made food.

A professional chef has to be able to make dozens, maybe hundreds of different dishes on demand. Maybe thousands, when you count all the sides and sauces and garnishes. And she has to cook for hundreds of people every night, and every single meal that leaves the kitchen has to be top-quality, and there have to be choices to satisfy thousands of different palettes. A home cook just has to make dinner for themselves and their family or guests, and they only have to make one meal, and they only have to please themselves.

Similarly, a commercial recording studio might be cranking out a new album every week, made by an engineer who has never heard the band before, who might not even like the band. The band might have instrumentation or sonics that are completely different from anything the engineer has worked on in the last year. The band might be incompetent and bad-sounding. But the studio is still accountable for turning out top-quality product, quickly, day after day, making every band that walks in the door sound like rock stars. This is a categorically different task from recording your own material that you love and have worked on and can spend time on without a meter running.

So put out of your head any notion of trying to compete with commercial studios in terms of GEAR, and put into your head the notion that you can still compete in terms of SOUND (albeit in a more limited range). If your Aunt Minnie can make a great pot roast at home, you can make great recordings at home. All you need is ears.

So anyway, what makes a good microphone? Read on...

There are a lot of different, interacting factors that go into the "sound" of a microphone. Perhaps more to the point, it is more common for the "sound" of a mic to change with the particulars of its application than not. In other words, how you use and where you place a mic is just as big a component of the "sound" as the mic itself.

In no particular order, some things that make one mic sound different than another in a given application are:

- Directional response-- an SM57 has a very tight cardioid pattern that is excellent at recording the stuff you point it at and rejecting everything else. This gives it a very close, focused, tight sound that happens to complement other features of the mic. It also makes it very difficult to use for vocal recordings, because every movement of the singer's head alters the sound. It furthermore lends the mic a potentially unnatural "closed-in" or "recorded" sound, which could be good or bad. A U87, on the other hand, has a very broad, big, forgiving pickup pattern, which is reflected in the sound. The U87 gives full-bodied, open, natural-sounding recordings of pretty much whatever is within its intuitive pickup radius. This makes it a very easy-to use mic for vocal recordings, but also a potentially problematic one to use for, say, close-miking a drum kit. It also makes the mic susceptible to the sound of the room. Which could be a problem in subpar recording environments. The U87 will give a full, lush, natural recording of a boxy, cheap-sounding bedroom studio if that's where you put it. Could be good or bad.

-Proximity effect. All directional mics change in dynamic and frequency response as you move closer to or further from the source. Speaking broadly, the closer to the source you get, the more the low end fills out and builds up. This can work for you or against you, and different mics can have different kinds and degrees of proximity effect. A mic with a big proximity effect can give a singer with a weak voice a big, movie-announcer, "voice of God" sound, but it could make a rich, gravelly baritone sound like the mic is in his stomach. It could make an airy alto diva sound like a throaty roadhouse karaoke girl. It can give power and throaty "chest" to screaming rock vocals, but it can also exaggerate pitchiness or vague tonality in untrained singers. With instruments, the same kinds of problems and benefits can pose similar conundrums. Moving the mic further away or closer to the source changes the proximity effect, but it also changes other aspects of the sound in ways that are inter-connected with the mic's polarity and sensitivity. Any of which may be good or bad.

- Sensitivity and dynamics response. This is intrinsically related to both of the above effects. The afore-mentioned U87 is a wonderfully sensitive mic, that picks up and highlights shimmering harmonics and "air" that can sound realer than real. They can also turn into gritty, brittle hash in the extreme highs when recorded through cheap preamps or processed with digital eq. The afore-mentioned SM57 is, on the other hand, a rugged, working-class mic, designed for military applications to deliver clear, intelligible speech. No shimmer or fainting beauties here, just articulate, punchy upper mids that cut right through noise and dense mixes. Either one could be better or worse, depending on what you're after. Sensitivity and dynamics response work differently when recording sources of differing volume. Some mics (like the SM57) tend to "flatten and fatten" when pushed hard, giving a kind of mechanical compression that can sound artificial and "recorded," although potentially in a good way, especially for stuff like explosive snare, lead guitars, or screaming indie-rock vocals. Other mics overload in rich, firey ways or simply crap out when pushed too hard. This last is particularly common among ribbon mics and cheap Chinese-capsule condensers, which sometimes sound great right up to the point where they sound outright bad. Once again, careful listening is the key.

The very best (and most expensive) mics deliver predictable, intuitive, and usable dynamics, proximity effect, sensitivity and pickup patterns in a wide variety of applications, as well as very consistent manufacturing quality that assures consistent frequency response and output levels from one mic to the next. Cheaper mics are often much better at one thing than another, or are hard to match up pairs (one mic outputs 3dB higher than another, or has slightly different frequency response or proximity effect, etc).

Inexpensive mics are not necessarily bad-sounding, especially these days. There is a tidal wave of inexpensive Chinese condenser capsules that are modeled on (i.e. ripped off of) the hard work that went into making the legendary mics of the studio world. There is a lot of trial-and-error that goes into designing world-class mics, and a lot of R&D cost that is reflected in the price. For this reason and others, top-tier mics tend to be made with uncompromising manufacturing, workmanship, and materials standards, all of which cost money.

Moral issues of supporting dedicated craftsmanship aside, whether it is worthwhile to pay for that extra percent of quality when you can buy a dozen similar Chinese mics for the money becomes almost philosophical past a certain point. If you're building a home addition, professional-grade power tools will make the job a lot easier and go a lot faster, but flimsy discount-store hand tools can still get the job done if you're willing to deal with more time and frustration. If you've ever tried a building project or worked a trade, you'll understand immediately what I'm talking about.

But since most musos are work-averse layabouts when it comes to practical arts, these can be hard distinctions to draw. If you've ever read reviews of the modern wave of cheap condenser mics, they almost all read the same: "surprisingly good for the money! Not quite as good as (fill in vintage mic here), but a useful studio backup."

By that measure, the average starving-artist-type could have a closet full of backup mics backing up nothing. The reality is that these second-tier mics CAN be used to make first-class recordings, but they often require a little more work, a little more time spent on placement, a few more compromises, a little more willingness to work with the sounds you can get as opposed to getting the sound you want, and so on.

A commercial studio has to be able to set up and go. If the first mic on the stand in the iso booth isn't quite the right sound, they swap it out for the next one. Three mics later and they're ready to roll tape.

In the home studio world of fewer and more compromised mics, it might take trying the mics in different places, in different rooms, at different angles. Some cheap mics might sound great but have terrible sibilance unless they're angled just so. That might mean an extra four takes, or it might mean recording different sections of the vocal with the mic placed slightly differently, which might in turn mean more processing work to the get the vocal to sound seamless.

These are the tradeoffs when you're a self-produced musician. The gear in professional studios is not magic (well, maybe one or two pieces are, but most of it is ordinary iron and copper). The engineer is not superhuman. The wood and the acoustics are not made by gods. But the tools, experience, versatility, and professional expertise are all, at the very least, great time-savers, and time is worth money.

If you have more time than money, or if you prefer the satisfaction or flexibility of doing it yourself, you can absolutely do so. You just have to trust your ears, and keep at it until it sounds right.

I want to double back to this notion of "all you need is ears." If you have read through these first few posts, I hope that it is becoming clear that this principle does not denigrate the work or the value of recording professionals. On the contrary, it is ordinary civilian ears that distinguish the work of great recordists. And there are some great ones, people who deliver recorded works that are beautiful in their own right, like photographers or painters who make gorgeous pictures of everything from old shoes to pretty girls.

But it is also those same ordinary civilian ears that allow us to hear when our own recordings are substandard.

I am taking it for granted that anyone reading this thread has already, at some point or another, made good-sounding music. There was a time when all that recordings aspired to was *accurate* recordings of good-sounding music. This objective is preposterously easy these days. I recently tried a $50 Earthworks knockoff mic made by Behringer that is absolutely fool-the-ear accurate. Throw it in a room and record a conversation with this mic and play it back through decent speakers and the people in the room will start replying to the recorded conversation.

But that is not usually what people are looking for in modern popular music recordings. These days, everything is supposed to be larger-than-life, realer-than-real, hyped and firey without sounding "distorted." We are no longer creating accurate recordings of live performances, we are creating artificial soundscapes that the live concerts will later try to duplicate with studio tricks.

You have whispered vocals over a full metal band backed a symphony orchestra, with a delicate finger-picked acoustic guitar on stage right. And it's all supposed to sound real, and big, and natural. And when the singer goes from a whisper to a scream, the scream is supposed to *sound* 20dB louder without actually *being* any louder than the whisper. Both of which are supposed to sound clear and natural over the backing band, which is of course supposed to sound loud as hell, louder than the philharmonic behind it. And everything is supposed to sound clearly articulated and distinct, including the chimey little arpeggiated guitar. And by the way, can we squeeze in this low-fi record loop and make it sound proportionate like an old record player but also clearly audible.

And the answer is yes, we can do all this. We can make conversation-level hip-hop lyrics sound bigger than explosions, we can make acoustic folk duos blend seamlessly with industrial drum machines, we can make punk rock bands that sound indie and badass while singing autotuned barbershop quartet harmonies with forty tracks of rhythm guitar. We can make country-western singers sound like heavy metal and heavy metal bands sound like new age and we can make "authentic audiophile" jazz recordings where the cymbals sound twenty feet wide and fifty feet overhead.

All these things we can do. But these are no "captured" sounds, any more than a Vegas hotel is an "authentic" reproduction of an Egyptian pyramid or a Parisian street. These are manufactured illusions. Unlike a Vegas hotel, the construction costs are almost nil. Reaper and programs like it have practically everything you need to create almost any soundscape you can imagine. All you need is ears.

This might sound like a rant, but my point is a very specific and practical one. Sound is at your disposal. Modern technology has made its capture, generation, and manipulation incredibly cheap. You can twist it and bend it and break it and re-shape it in any way you imagine. The power at your fingertips is huge. There is no excuse for dull, noisy, bland recordings except user error.

There is a lot more ground to cover, but no way to cover it all, or even most of it. Your ears are a far better guide than I or anyone else. Anything I or anyone can describe about sound, you can hear better.

With any instrument or sound source, the biggest single recording decision to be made is whether is to record in the nearfield or the farfield. These are not just arbitrary words for subjective distances from the source.

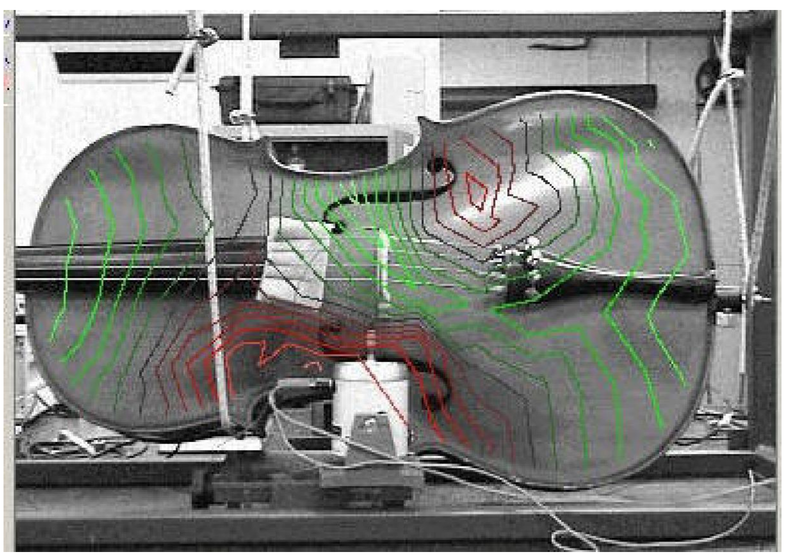
The "nearfield" is the radius within which the sound of the instrument is markedly different depending on the location and angle of the mic or listener. The "farfield" is everything outside that radius. The nearfield of most instruments usually ends at a distance about the size of the main body of the instrument itself. So an acoustic guitar's nearfield extends maybe about 3 feet away from the body of the guitar. A drum kit's nearfield extends maybe five or six feet away, and a grand piano's is even bigger.

This distinction is obvious to visualize with a drum kit. If you put a mic right next to the floor tom, it's obviously going to record a lot more floor tom than hi-hat. It is also going to record the other kit pieces disproportionately, according to their distance from the mic. This is "nearfield" or "close" miking. Anywhere we put the mic inside this "nearfield" is going to make a very big difference in the recorded sound, nut just in subtle ways, but in very specific and acute alterations.

In order to get to the "farfield," we have to move the mic far enough away from the kit so that all the drums are heard more or less proportionately, no matter where we angle or place the mic. The mic has to be *at least* as far away from the closest kit piece as the closest kit piece is from the furthest kit piece (e.g. if the outside edge of the floor tom is 4 feet from the outside edge of the crash cymbal, then we should be at least 4 feet away from either one). Changing the mic position or angle in the farfield can still affect the sound, but small changes will not have the same drastic impact on the overall balance as they do in the nearfield. We have crossed the critical line where the individual kit pieces begin to sound like a unified whole.

The drummer's head and ears are in the nearfield, and as it happens, putting all the drums in easy reach has the effect of creating a pretty good balance of sound, so that they are also all about equi-distant from the drummer's head. Nevertheless, the sound that the audience in the front row hears is apt to be quite different from what the drummer himself hears.

This distinction becomes a little harder to wrap your head around (but no less important) when we get into single-body instruments like acoustic guitar. The guitar is shaped the way it is to produce certain resonances from different parts of the body and soundboard. Here's a resonant image overlay showing the vibrations of a violin soundboard at a particular frequency:



As you can see, different physical parts of the instrument are producing different parts of the sound, the same way that individual kit pieces in a drum produce different parts of the overall kit sound. If there were a way to "watch" this happening, you'd see different parts of the instrument's body "lighting up" and moving across the body as different oscillations as various notes and chords sounded and decayed.

So if we point a close mic at one part of a guitar body, we will be picking up a disproportionate amount of the particular resonance of that square inch of the body. Not until we get a few feet away do we get a full, unified, consistent image of the entirety of the guitar sound.

This can work for us or against us. Moving the mic around inside the instrument's nearfield can allow us to highlight certain aspects of the sound, or downplay unflattering aspects of a cheap instrument.

I want to try and stay away from specific "recipes" for now, but one thing that bears mentioning by way of illustration is the common mistake made by beginners of trying to record a guitar or string instrument by pointing the mic right in the soundhole or f-hole. If you want to think of a guitar top as a "speaker," the soundhole is like the woofy "bass port" that extends the low end and increases efficiency. It is not usually the most satisfying or flattering place to record.

The most versatile "catch-all" generic starting positions for nearfield single-mic acoustic guitar are usually ones that fire *across* the soundboard, not right at it. The old standby BBC technique was to put a mic close to the strings near the body fret and aim the mic across the top of the soundboard (i.e. parallel), giving a bright, stringy, but fairly balanced sound. Moving the mic closer or further to the strings, or tilting it so that it fires across the soundhole or "misses" it offer quick-and-easy adjustments to the tonal balance. An alternative approach (some might say more natural or full-bodied) is the "below the guitar" position, where you put the mic near the seated player's knee, again firing across the top of the soundboard, angled to taste.

These are starting points, not ending points for finished studio recordings. In fact, they are actually designed to try and "defeat" the most prominent nearfield effects. The point of the example is not to tell you how to mic an acoustic guitar (there are a billion threads for that), the point is to illustrate the *reasons* why certain approaches achieve different results.

An informed understanding is not a substitute for listening and experimentation, it's just an accelerant that speeds up the digestive process. Like the eq example above, this is not stuff that you can just "think through," but understanding the whys and wherefores can help you to understand the connection between the approach and the results attained, which can in turn help you to make better, more systematic, and more purpose-driven evaluations.

With that in mind, note now that the acoustic guitar player's head, like drummer's head, is **also** in the instrument's nearfield. But unlike the drummer, the guitar player is not situated in anything close to a representative position-- the audience in row one is typically getting a totally different sonic profile then the guitar player, whose head is to the side of and almost "behind" the guitar, and whose hearing is supplemented by direct coupling through the chest.

This presents a couple of interesting considerations. One is that the guitar player might be quite taken aback by the recorded sound of the guitar, and might feel like nothing sounds right or "feels" right (more in a minute). Another is that monitoring, e.g. through headphones, could be a challenge, especially if you are recording yourself and trying to evaluate sounds while you're playing the instrument.

The headphone mix is one of the most powerful tools that a recording engineer can use to direct and control a performance. This is going to be a very big deal when we get into vocals, but it's worth touching here. You need to know what you're listening TO and what you're listening FOR.

Guitar players are often very finicky about the sound of their instrument, and rightly so. One of the things that makes guitar such a compelling instrument is the remarkable sonic expressiveness of the direct manipulation of the strings by the player. If the player is not hearing what they want, sound-wise, they are apt to change their playing technique to compensate. This can either be a virtuous cycle or a vicious one. For instance, a player who is accustomed to pounding on the strings to get that extra "bite" might start to back off if they have an stringy-sounding headphone mix.

This is what good guitar players do, after all-- they use miniscule and subconscious variations in pick position and fret pressure and picking technique and so on to get just the right balance of chirp and thwack and thump and strum and sing and moan and so on from every note and chord. Whether the subconscious adjustments made for the headphone mix are a good thing or a bad thing is totally subjective and conditional. From a purely practical standpoint, having the guitar player perform "for the mic" is theoretically a good thing.

But whatever we feed to the headphones, the player is always going to hear something a little different simply because the instrument itself is acoustically coupled to his or her body. This is not usually that big a deal, until the player himself is the one making sonic evaluations of the mic position in real-time.

To put it another way, the process of mic placement is essentially self-correcting when it is directed by a dedicated engineer in the control room. The combination of playing technique and captured sonics interact until the engineer is satisfied that she's getting the best or most appropriate overall sound. If you hearken back to the stuff we said about the importance of accurate monitoring at the start of this thread, and then imagine the engineer trying to make decisions with one extra speaker or resonating guitar pressed against his body, then you start to get the idea.

This is not insurmountable. Once again, the careful application of trial-and-error and critical listening can level the playing field, but sadly there is no simple eq recipe or plugin that eliminate this effect.

My point is not to discourage anyone, but to get back to the thread title. You can play good guitar music (or whatever). You can play it so you like the sound of it. Chances are, you have even played it with headphones and have been totally "into" the sound you were getting, maybe even more so than usual. If you have then played it back and been disappointed, it might have something to do with the principles at work here. Maybe that sound that you were "into" while playing was not actually the sound being recorded, but a combination of captured and un-captured sounds. The headphones were not telling you what was "going to tape," they were just supplementing and hyping up the sound of the guitar resonating against your chest. And if you recall what we said about level-matching and louder always sounding better, you can start to see where this kind of monitoring can be misleading, especially if the headphones are giving you a louder overall perception of the sound while you're playing, but not when you're just listening to the playback.

If you've ever been through the above scenario and have been tempted to blame your mic or your soundcard or you preamps, stop and think for a moment-- if they were really the culprit, then why did it sound so good while you were tracking, and only sound worse on playback? Are some lightbulbs going off yet?

Nearfield vs farfield continued.

Getting back on track, it may seem almost pointless to talk about farfield miking these days, since almost nobody does it anymore, at least not so far as home-produced multitrack recordings go. But at the risk of wasting oxygen on forgotten lore, there is a lot to be said for farfield recording when it can be made to work, and the principles are still valuable to understand as we get into mixing, acoustics, and sound transmission.

In the olden days, the way to get a "big sound" was to get a shitload of musicians in a room all playing together-- lots of guitars, two pianos, two drumkits, horns, strings, woodwinds, shakers, tambourines, background singers, vibes, xylophone, whatever. Then let a big, natural reverberation fuse it all together. If you listen to those old Phil Spector "wall of sound" or "one mic over everything" records, it's hard to make out any particular instrument, or sometimes even the lead vocals. The sound could be huge, but every single instrument is small, just a little bit of texture in the overall effect. This is like that symphonic synth patch referenced above, favorite of heavy-metal intros.

But a lot of things were different in those days. One of the biggest differences was that the musicians were basically considered anonymous, disposable role players. These were the days of house bands and label contracts and separate in-house songwriting and arrangement teams and salaried stars and so on. Pre-Beatles, in other words, the days before guitar gods walked the earth.

Nowadays every musician is supposed to sound like a sonic super-hero. The bass player who earns his living as a professional octave pedal with tattoos and who occasionally plays a leading seventh must be clearly heard, for all to appreciate his seventh-playing prowess in all its glory. The punk guitarist palm-muting quarter-notes in the key of the fretboard dots has to have sixteen tracks lest the chunka-chunka fail to overwhelm and subdue any aspect of the listener's central nervous system. The DJ whose sheer artistry allows him to hold a headphone with a single shoulder while simultaneously operating a fader and playing records must not be made to feel like a second-class citizen by having his performance obscured by more pedantic forms of music.

In other words, putting the band in a room with thirty other musicians and capturing a massive sonic vibe of creative energy is not likely to please the client. Unless of course it is overlayed with double-tracked, close-miked, compressed and hyped-up versions of the "named member" performances.

Even if you eschew the old ways of doing things, it is useful to consider some of the potential of farfield recording, and some of the implications of doing everything nearfield.

One immediate and often overlooked effect of recording nearfield is that reverb applied to a nearfield recording does not sound the same as an actual recording of the performance of the room. People go searching high and low and spending fortunes trying to replicate the old plate and chamber reverbs of yore, trying to get that big, rich, warm, natural sound. All without stopping to think that a reverberated nearfield recording of a guitar does not sound like an old recording of a guitar in a room **BECAUSE THE NEARFIELD RECORDING DID NOT RECORD THE WHOLE SOUND OF THE GUITAR.**

So when you throw some fancy plugin or all-tube spring reverb on a close-miked guitar sound or drum overhead and it sounds splashy and brittle and artificial, that is at least in part because **IT'S NOT PROCESSING THE SOUND OF THE INSTRUMENT IN THE ROOM.** It's processing the sound of a surgical capture of an exaggerated microscopic part of the sound.

You cannot make a dehydrated steak taste like real steak by adding water. You cannot do it with vintage water or all-tube water or water with ceramic capacitors or water salvaged from an early session at Sun studios, because the dehydration process changes the chemistry and texture of the steak and alters more than just the water content.

Similarly, and this is neither a good thing nor a bad thing, just a thing, nearfield recording is not the same thing as recording in an anechoic chamber. It's not just "instrument sound minus room sound," it's a distorted and selective recording of particular parts of the sound. "Just add reverb to reconstitute" does not necessarily bring it back to life in the same state it was. If you put a recording of a telephone call through reverb, it is not going to produce a convincing illusion of a person speaking in a room, it's going to sound like a reverberated telephone call. Even if you have the best reverb in the world.

Now, this is not to say that you can't achieve great results with reverberated nearfield recordings, and it's not to say that you even need reverb. And nearfield recordings can often sound better than the actual sound of the instrument in the room, especially if you have a bad room.

But a lot of the double- and triple- and quadruple-tracking of instruments and finicky use of delays and general obsession with "fattening" and "thickening" that goes on these days is part of a complex effort to try and restore the sense of size, volume, and richness that is lost when we strip away the fully-developed sense of sound pressure moving air molecules by close-miking everything.

Something that I am certain exacerbates this process is failure to understand the effects of level-matching. During mic placement, when we pull the mic back away from the source, it gets quieter. Remember what that does to our perception of the sound?

This is very hard to compensate for in real-time. Even if you adjust the gain after re-positioning the mic, the immediate effect of the transition (before you compensate for the level change) is of a sound that gets bigger and hyper when you nudge the mic closer, and smaller and weaker when you back the mic off. That immediate impression is hard to shake off, even if you're on the lookout for it (which a lot of people are not, even professionals who should know better).

This creates a highway to hell for the well-meaning recordist who wants a "big" but "natural" sound. When they back the mic off, the snap reaction is that they lost some "big." When they push the mic in, they get big back but lose some "natural." So they try a little reverb to put back the natural. This increases the signal gain and gives even more "big," but doesn't quite sound as "natural" as it should. So they fiddle with delays and compression and try adding more doubled-up tracks and whatnot to try and "smooth" out the sound and "fatten" it up and so on. Which will, of course, add more signal strength and push the whole thing a little closer to clipping, at which point they have to back off the signal level and end up deciding that they need a 12-foot plate reverb or an Otari machine to get "natural" tape delay (both of which of course add a little more signal gain).

Repeat this process for eight months and spend an extra $83,000 of the starving band's advance money and eventually you end up with a quarter-million-dollar, radio-ready commercial recording of a clipped, phase-smeared, hundred-and-eighty-tracked, fatigue-inducing mix of a three-piece folk-rock group that is ready to be sent to mastering for further limiting.

To their credit, most home studios usually give up a lot earlier in the process, but they are still desperate to know the "secrets" of how the pros work.

- Instruments that are panned dead center are identical to instruments cloned and panned both hard right and hard left. On a good, properly-positioned speaker setup, there should be three specifically identifiable "cardinal points": hard left, hard right, and the "phantom center." Everything else tends to be a blurry and variable no-man's land, which is fine, it just is what it is. But you should be able to hear instruments or content coming from those three distinct locations if you close your eyes-- it should basically sound like there are three speakers, with stuff in-between (this is the system setup, not necessarily the pan position).

  Assuming you have a good monitor setup where you can hear the three cardinal points using test tones or reference CDs or whatever, why is it that some instruments panned center seem offset, or shifty, or seem to come from that vague no man's land? One common reason is different masking effects on the left and right. E.g., if you have a guitar in the right speaker and a piano in the left and the bass dead center, the guitar is going to be masking and covering some parts of the bass sound, and the piano is going to be masking and covering some other parts. If you have something else dead-center (like a full-spectrum rock vocal or lead part), then that is going to be masking some other parts of the bass sound, maybe most of the upper-midrange articulation. So different parts of the bass sound are going to poke through wherever they can find room and the whole effect might be a somewhat de-localized sound, which is neither good nor bad, just a thing to deal with. Everything affects everything, and frequency management of different instruments and different parts of the stereo spectrum is huge.

- Playing technique. Some of the most highly-valued studio musicians in the world are bass players who can generate "hit bass," which usually has almost nothing to do with the kinds of acrobatic technical virtuosity required of guitar players or session vocalists. These hitmakers frequently play pretty simple lines, but they control the dynamics, note duration, and tonal quality to get just the right "feeling" that beds the song and complements the drums.

One of the biggest differences between a really good bassist and a guitar player playing bass is that the bass player will tend to play with a much lighter touch while still controlling the dynamics. Guitar, especially electric guitar, is an instrument that was made to played loud. Even with "clean" guitar sounds, the amplification is typically a very crude, primitive, soviet-era system that is meant to overload and saturate on the input stages, the output stages, and at the speaker itself. This is what gives that rich harmonic "fire" and expressiveness to electric guitar. It also compresses the signal and delivers articulate, emotional "oomph" that stays at a fairly consistent level but just "sounds" louder when you pick harder.

If you take the same approach to bass, and pound the hell out of the strings, playing with the kind of expressive, loosey-goosey timing that many guitar players do, the sound is apt to overload the pickups, the input stages (preamps), and everything else, producing the same kind of dull, farty, obnoxious-sounding lows that come from overloading cheap speakers.

Bass needs a lot of headroom and power. It requires high-wattage amplification (ever notice how a 50-watt guitar needs a 1,000-watt bass amp to keep up?), which translates into good, adequately-powered monitors so that you can hear what you're playing clearly and powerfully without saturating the signal, and it requires lots of clean input amplification, which means playing with a lighter touch and rolling off your preamp input levels to insure that you're not pushing them too hard. Just because your soundcard's "clip" LED doesn't come on until you pin the peak meters doesn't mean that it has adequately-sized transformers to handle massive steady-state basslines right up to 0dBFS. The AD converters might not "clip" until long after the analog preamp has become voltage-starved and starts to fart out from current overload (Notice how everything seems to come back to level-matched listening comparisons, EVERY STEP OF THE WAY, including how you set your input levels? **Golden ears in one easy step**). If you've been recording bass with hard-picked notes on an inexpensive starter bass plugged into an inexpensive prosumer interface, trying backing the gain down and playing the notes very lightly and see if clarity, focus, and power doesn't improve dramatically. Gain-staging is a big topic for a later post, but like everything else, all you really need is ears.

- This might sound obvious, but use fresh strings and a good instrument. Bass strings sadly wear out quickly, and unless you're James Jamerson (the greatest bass player who ever lived, but not someone most people are equipped to emulate), old strings are even worse for bass than guitar, while also being more expensive. You can boil old strings in water with a little white vinegar to restore some life if cash is tight. A decent bass doesn't have to be all that expensive, but the pickup configuration and general sound of the instrument should complement the kind of music you do. A fat, funky, burpy-sounding P-bass is not going to sound appropriate in a nu-metal band, and a deep, clackety, growly, heavy-body bass with EMGs might have a hard time fitting into mellow blues-rock ballads.

-Arrangement and performance. This is a topic for another thread, but a bass is not just a four-string guitar. Whatever instrument is playing the lowest note sets the tonal foundation for the whole song. If the bass plays a fast run up to the seventh, then the whole band sounds like it just played a fast run up to the seventh. That's not necessarily a good thing or a bad thing, just something to be aware of. If the bass plays with a loose, expressive timing, the whole band can sound lurchy and out-of-step. If the bass plays tight, sensitive timing in synch with the drums, then it sets the solid foundation that frees up the lead instruments to play expressively. The bass is the most powerful instrument, literally, and with great power comes great responsibility, in the words of the famous audio engineer Uncle Ben (from Spider-man, not the rice guy). If the bass line is "off" (which is a purely subjective judgment), then the whole thing just doesn't sound or feel right. This is purely a "feel" thing, it does not necessarily mean that every note is plucked right on a drum beat. In fact, the nature of the bass is such that slightly dragging or pushing the beat often produces the best results, because bass waves are slower to develop and interact in funny ways. But it has a big effect on gluing the whole sound together.

Let's talk a little more about farfield vs nearfield recording and how the concepts interact with some of the stuff from earlier.

As a quick aside, if you have followed the thread so far, one of the biggest reasons to purchase actual dedicated-purpose nearfield monitors is because they are designed for even response at close-up listening, as opposed to the Bose tagline of "room-filling sound," whatever that means (it probably doesn't mean perfectly linear mids at a distance of two feet from the speaker). I will leave the advantages of monitoring in the nearfield to the acoustics thread, but the short version is that you're generally better off listening to monitors that are too close than too far.

Do you play electric guitar? If so, do you play with the speaker a centimeter away from your ear? If you do, you should probably stop. But if you are like most players, you have probably spent significant effort on dialing in an amp sound that sounds good from, say, 1.5 meters or 6 feet away (I'm trying to incorporate metrics for readers who don't live in this alternate universe known as USA). So why do we commonly record guitar amps with the mic shoved right up in the speaker grill?

For that matter, why do we record string bass with a mic under the bridge, or piano with mics under the soundboard, or drums with mics right up against every kit piece?

The answer is complicated in theory, but the short version is because it often sounds better.

In the real world, we are making records for people to listen to on a variety of playback systems, in a variety of listening environments. And ideally, we want the records to sound good in all of them. A "purist" approach might be to simply set up the ensemble in a concert hall and record them from row 3, center with zero processing. This is all well and good for re-creating the ideal listening experience in a dedicated audiophile listening room, but an immediate problem presents itself in proletarian real-world playback.

In a loud car, or as shopping mall background music capped at 60dB SPL, or in a noisy bar's jukebox, the playback is not going to be a philosophically pure listening experience. We have no control over the playback volume or acoustics. We have no control over the background noise.

But an interesting solution presents itself if we consider the ways in which human hearing automatically adjusts for surrounding acoustics (if you haven't already read through the acoustics stick in this forum please do so). If we simply recreate the SOURCES (i.e. the individual instruments) proportionately, then we can theoretically create a virtual concert hall in whatever space the listener is in. I.e. we don't actually have to re-create the "ideal listening experience," we can just reproduce all the instrument sounds, balance them out, and let whatever environment the listener is in take care of the rest. And the obvious way to do this is with direct recording and close-miking.

BUT, that leads to some pretty significant complexities. For instance your electric guitar sound that was developed for listening six feet (or 1.5m) away is going to sound a lot different on studio monitors with the mic shoved in the grill. Especially if you are trying to make records that might be played back at a different (lower) volume than you usually play guitar.

The fact is that volume makes a big difference. For example, let's take gunshots. If you've ever shot a gun, you know what I'm talking about. If you haven't shot a gun, imagine something really loud and then make it a lot louder.

Now, with that in mind, I want you to think about TV and movie gunshot effects. The fact is that an authentic recording of a gunshot, when played back at sane living-room listening levels, sounds like a wimpy little "pop" or hand clap. You have probably heard this kind of gunshot recording before in documentaries or newsreels or some such and thought "how wimpy." But that's what a gunshot sounds like, unless it is at ear-blasting, speaker-rupturing SPL levels.

So what happens in *most* TV and movie soundtracks is that they compress, saturate, stretch out, and "hype up" the sound of gunshots to create the *impression* of loudness within safe, reproducible playback levels. This is particularly pronounced if you watch a movie or TV show where there are massive-sounding handguns interspersed with smaller ratatat-sounding high-caliber machine guns. In reality, the machine guns are just as loud and powerful as the sidearms on every round, if not more so, but there is no way to fit the explosive "decay" into every machine-gun round, so the mixer is forced to compromise. In real life, machine guns are not abruptly treblier and smaller-sounding than handguns. Real-life machine guns are a great way to go deaf quick, but in the movies, the action hero's voice sounds just as loud and powerful as the high-caliber assault rifle, which is yet another illusion.

The fact is that we can, within limits, create a whole lot of sonic illusions. Where these are most useful in the studio is in creating the right sense volume, space, and size that will fool the ear on playback. In other words, we can make gunshots *sound* deafening, even at perfectly safe listening levels, within limits.

Facts about the rock band AC/DC that you might not have known:

-The singer from AC/DC usually sings whisper-quiet.
-The guitar players from AC/DC usually use quite low gain settings for heavy rock guitar, older Marshall amps with the knobs turned up about halfway (no distortion pedals).

Both of these fly in the face of impressions that most casual listeners would have about AC/DC, which is a band that has been releasing some of the loudest-sounding records in rock for decades. The reality is that the moderate amp gain settings actually sound louder and bigger than super high-gain settings, which are prone to sound nasal and shrill at low volumes.

The singer, like TV gunshots, is creating the *impression* of loudness without straining his voice by only pushing and exerting the upper harmonics that are strained while screaming. IOW, he's singing not from the diaphragm, as most vocal coaches teach, but from the throat and sinuses. Instead of screaming, he's skipping the vocal chord damage, and only exercising the parts of the voice that are *unique* to the scream. He's using parts of the voice that normally never get used except when we're screaming our head off, and the result is that it sounds like someone screaming his head off, even though he's barely whispering. Because nobody walks around talking like that, the effect is of a "super-scream," something that sounds louder than any mortal human could ever scream, because the normal sound of a human voice is completely overwhelmed by the effects that are usually only heard during screaming.

My point is not to endorse AC/DC, nor to say that you should try to emulate them, only to cite a commonly-heard example as a way to illustrate how perceived loudness, size, and impact can be crafted as a studio or performance illusion.

Nearfield close-miking opens up a world of opportunities in this respect. We can zero in on the sharp "thump" of a kick drum and make it feel like a punch in the chest for an uptempo club track, or we can stretch it and compress it to sound like distant thunder for a slow mournful ballad. We can take a poppy, bouncy snare and turn it into a gated, white-noisy industrial explosion or we can subtly lift up the decay to get a sharp, expressive, woody crack. We can flatten out the guitars and shove the Celestion greenbacks right into your ears. We can get the bass to pump the speakers and we can make the piano plunk and plink a whole new backbeat.

But for the reasons mentioned above, we still run into trouble with trying to get "natural" sounds from close-miking. This might be something of a lost cause, but listen to modern-day records on the radio and see how many of them actually sound anything like a band in a room. Not many. Whether this is a good thing or a bad thing is not for me to say, but I will go out on a limb and venture that increasingly artificial-sounding productions lend an increasingly disposable quality to popular music.

How many of today's records will people still be listening to in 30 years? Will some balding middle-aged insurance salesman be telling his kids that they don't understand rap metal and that their stuff is just "crap metal" and go home to watch Limp Bizkit's PBS special at the Pops while sipping iced Chablis?

Okay, this is probably a bit premature, but I might not have much posting time before '09, and I promised this in an earlier post:

A short **buying guide** to recording gear...

First rule is do *not* go into debt over a hobby (even if it is a hobby that you are certain will be your lifelong ticket to fame and fortune).

Second rule is do not buy anything that is not on your afore-mentioned **pad of paper.** The way to avoid sucker buys is to wait until you have actually needed something in one or more actual recording projects. There will *always* be stuff that you need.

Once you have saved up a significant sum to upgrade your studio, the absolute best way to shop for recording gear is to book a few hours at a well-equipped commercial studio and try out their gear. Be up-front about what you are doing, and you will find the people there very helpful. All recording studios these days are well-accustomed to dealing with home studio operators. For a few hundred bucks you can sit down with someone who has recorded actual rock stars and see how they would record you, try out the different gear, and see how they actually use it. Bring your MXL mics or whatever along and hear for yourself the differences that preamps make on your voice and your instruments. The knowledge is worth more than you spend, and any good studio will be happy to help, knowing that the biggest thing you will take away from the experience is the understanding of how valuable their gear and expertise is.

That said, here are some tips for approaching reviews:

-Professional studio operators and engineers are very likely to be unfamiliar with the low-end of the recording market. Very few top-flight engineers and producers have much exposure to a wide cross-section of $100 Chinese condenser mics or freeware plugins. They spend their days recording with established name gear, not scouring the web for freebie synth patches. So when a pro says that a certain plugin has finally broken the barrier to compete with hardware compressors or whatever, it might be only one of a half-dozen plugins he's ever seriously tried. Same with cheapo mics, preamps, and the rest of it. They may have no idea how much the bottom of the market has improved in the last 5-10 or even 20 years. And this is especially true of the big-name super-legendary types. HOWEVER, if they say that something sounds good, chances are very high that it does sound good.

- On the other hand, many amateur forum-goers have never had much exposure to top-flight gear. When someone on a forum says that X is the best mic they've ever tried, it is quite possible that they have never tried any other serious studio mics. And consensus opinions can emerge on individual forums and message boards with little connection to reality. Somebody asks about the best headphones, and one or two posters who have only otherwise used ipod earbuds rave about one particular model, and before you know it, some totally mediocre headphone pick gets a dozen rave reviews anytime anyone asks about headphones on that forum. HOWEVER, what these kinds of forum reviews are collectively *awesome* at is sussing out technical, durability, and compatibility problems. Professional reviewers often get better support and/or optimized test samples (especially with computer-based stuff), but a real-world survey of amateur forums can give a very good sense of the kinds of problems people are having with a particular model on big-box laptops and wal-mart computers not optimized for audio work.

- Professional reviewers are another conundrum altogether. The resume criteria for this position is often almost nil, and the accountability is even lower. Everything is "a useful addition" to an otherwise well-equipped studio. Which is useless info if you're trying to build a well-equipped studio in the first place. On a scale of 1-10, they rate everything a seven. Look for multiple 10s.

Down to the meat-and-potatoes:

Avoid intermediate upgrades. What the audio industry wants you to do is to upgrade a $100 soundcard to a $300 soundcard to a $700 soundcard to a $1,500 soundcard and so on. By this point you will have spent $2,600 to end up with a $1,500 soundcard, and the old ones will be close to worthless. And the next step is to upgrade to dedicated converters and a selection of preamps which will render the previous generation worthless.

Once you have functionally adequate gear, save up, and make your upgrades count. Buy the expensive, primo gear, not the incrementally "better" prosumer upgrade. Bona-fide professional gear holds its value and can be easily re-sold. A used $1500 Neumann mic can be sold tomorrow for the same $1500, and may even go up in value. But put $1500 worth of used prosumer mics on eBay and you're lucky to get $500 for them, and it will take a lot more work, hassle, and postage.

The price-performance knee has been pushed a lot lower in recent years, and there is a ton of cheap gear that compares sonically with stuff costing several times the purchase price. This means that the best deals are on the very low-end and the very high-end of the price spectrum. There are very cheap alternatives to mid-range gear on the one hand, and the heirloom-timeless stuff on the high end will hold its value on the other hand.

The next couple years will be a very good time to buy. The cost of old gear has been driven up exponentially in the past 15 years, even as the quality of low-end gear has shot up. A lot of pro studios have been closing their doors, but an ever-increasing number of hobbyist studios were driving up prices for heirloom gear in the days of easy credit and exploding home equity in the western world. You may have heard that those sources of personal wealth are collapsing. High-end studio gear has become a sort of "luxury good," and is very likely to start to lose value as buyers dry up and as lavish hobbyist studios get sold off in a tough economy.

There was a time maybe 15 or 20 years ago when you could just keep a sharp lookout for college radio stations and such that abruptly decided to "upgrade" to digital and you could get vintage tube preamps and such for practically or literally nothing. As stuff like ADAT and later ProTools allowed people to set up a "professional" home studio for sums of $20,000 or so, people began to look for ways to re-analogize their sound. And as the explosion of extremely cheap DAW studios came into being, prices for the old junk exploded, even as a newfound reverence for all things analog and "vintage" usurped the previous love of digital. This going to start to sound like a rant, but I promise it's going somewhere.

The explosion in prices for "vintage" and "boutique" gear was not driven by professional studios. Even before the home-studio boom, the arrival of cheap, high-quality digital and better broadcast technologies made a whole lot of local recording and broadcast studios redundant. There was a small increase in inexpensive project studios, fueled by the rise of punk, hip-hop, and "indie" music, but for the most part, the emergence of the ADAT and Mackie mixers spelled the beginning of the end for mid-market commercial recording studios, and began to turn broadcast studios into cheap, commodity workplaces devoid of the old-school audio "engineers" (who actually wore lab coats in the old days of calibrating cutting lathes and using occilloscopes to measure DC offset and so on).

The irony is that the explosion of cheap, high-quality digital fostered a massive cottage industry of extremely small home and project studios, that rapidly began to develop a keen interest in high-end studio gear. Even as broadcast and small commercial jingle studios and local TV stations (of which there were a LOT, back then) were dumping their clunky mixing consoles and old-fashioned ribbon mics and so on, there was a massive rise in layperson interest in high-end studio gear.

As the price of entry has gotten lower and lower, interest in and demand for truly "pro quality" sound has increased exponentially, and superstition and reverential awe has grown up around anything that pre-exists the digital age.

Some of this reverence is unwarranted. But there is no doubt that things were made to a higher standard in the old days, when studio equipment was bought on industrial and not personal budgets, and when consoles were hand-built to contract by genuine engineers who built only a handful of them per year, to order. Things were over-built, with heavier-gauge wires and components that were tested by sonic trial-and-error, and had oversized power supplies and artist-perfect solder joints and military-grade, noise-free precision knobs and so on.

There are still manufacturers working to this level of quality today. Whether and to what degree this stuff actually produces better sound quality is a bit like asking whether heirloom antique furniture is more comfortable than Bob's discount sofas. The answer is usually yes, and even when it's unclear, the difference in build quality and longevity itself usually has value.

The long and short is that genuine super-primo gear has intrinsic value that is likely to hold steady or increase as more and more of the world becomes interested in small-scale recording, even while cheaper, more disposable gear based on stamped PC boards and chips and flimsy knobs and so on continues to improve in quality, while simultaneously losing resale value.

The next year or two are likely to see a significant selloff by lavish home studios that were financed by home equity and easy credit in the western world. This is likely to lead to some very good deals for buyers. But in the long run, developing countries and increased interest in home recording is likely to sustain or increase the value of top-flight gear, even as the cost of low-end consumer stuff continues to decrease.