Why do your recordings sound like ass?

By **yep** / thread started 12-02-2008

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Introduction

Nothing personal, if the title does not apply, please ignore. But if you have ever asked yourself some variant of this, or if you have ever tried to figure out the answer on web forums, I'm here to help. This is in part a spin-off of some of the ideas explored in the acoustics thread, so there is some overlap.

Here's the scenario: Joe Blow, proud owner of a Squier Strat, an SM57, and a Peavy amp, buys an Mbox so that he, too, can "produce professional-sounding recordings on his computer", just as it says on the box. He makes recordings. They do not sound professional. He goes to the makeprofessionalrecordingsonyourcomputer.com forum and asks why. Responses include:

- Mbox sucks and you can't make good recordings on an Mbox
- I make recordings on Mbox and they sound pretty good
- You need a tube amp to record guitar
- You need a POD to record guitar
- You need an API preamp to record guitar
- You need two mikes to record guitar
- You need to get waves plugins to make good recordings
- Waves suck, you need UAD plugins to make good recordings
- I like Peavy amps
- I used a firepod and it sounds good
- What kind of speaker cables are you using?
- I use an all analog boutique amp emulator pedal and it sounds just like Slash
- Strats suck, you need a vintage Gretsch guitar
- Pros use mastering to get good sound
- I also have an Mbox but it doesn't play MIDI, please help
- Copy protection is evil.

Just in case those answers didn't clear things up for ol' Joe, I am endeavoring to create a thread of specific, practical, gear-generic methods for evaluating recording techniques and approaches, and yes, making purchasing decisions, all with an eye towards identifying weak links in terms of gear, acoustics, techniques, and methods.

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.

Before we get started, I'm going to make a request the participants try to avoid recommending or debating specific pieces of gear. There a billion threads all over the web for that. What there is less of is specific focus on principles and practical approaches. And at any budget, there are principles that can be used to make good-sounding recordings.

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.

Speakers

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.

One note bass

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.

Crossover frequency

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.

Subwoofers

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.

Room acoustics

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).

Your recording chain

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.

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

* Please note that none of this is to say that preamps or converters or mics don't matter.

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.

> Originally Posted by jplanet
>
> I agree that quality monitors are essential to mixing, but not necessary for good tracking.
>
> If you are in a scenario, as many are, where you record at home, but send your projects out to be mixed, I would say that you can get spectacular results with a $100 pair of AKG headphones...and your neighbors will thank you!
>
> If you're recording with a guitar amp mic'ed with an SM57, your neighbors will also thank you for using an amp sim VST...That also gives the mixing engineer the option to re-amp your sound...

Even though I'm going to disagree with your premise, I thank you for bringing the topic up.

You gotta do what you gotta do, and if it works, go with it. But my experience is that it is very difficult to make primary decisions with headphones, whether tracking or mixing, especially on stuff like electric guitar.

Headphones obviously exaggerate the soundstage, but they also tend to deliver exaggerated fletcher-munson effects, even at low-ish volume levels. Things that sound rich, full-bodied, and “big” on headphones have a way of sounding tinny and muffled on playback with regular speakers. Detail and presence evaporates, and electric guitars (for example) often sound excessively over-driven and nasally when you play back the tracks in the car or on a stereo.

There is nothing wrong with monitoring at conversation-level volume or below, in fact it is often desirable to do so. If you live in a circumstance where even conversation-level sound is too loud, then it's going to be hard to make a serious go of recording, but people have done it all with headphones.

In any case, this leads perfectly into my next post, which is all about level-matching...

Level-matching

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).

Peak levels and average levels

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.

* 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.

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.

Monitoring levels

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

> Quote:
>
> Originally Posted by junioreq
>
> I am Joe Blow, wow! Hours and hours looping that same progression.

You are not alone, Mr. Blow.

This whole idea of steady-state vs peak level and the effects of frequency thereupon has MASSIVE implications throughout the entire processes. If you can swing a simple SPL meter from Radio Shack it's a worthwhile expenditure of $30 or so, not that it has a lot of direct application to the recording process, but it's very useful to start to quantify and analyze the ways in which we perceive sound, and to have a sense of, for instance, how loud your car is, and how loud you like to listen to movies, and so on.

It's getting late here in Boston, and I'm taking phone calls and such, but I'm going to try and get in one more post tonight since it might be awhile before I can continue. Anyone else with something to say is free to jump in.

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.

More later.

> Quote:
>
> Originally Posted by Fabian
>
> 83dB sound pressure, measured where?

At the listening position, like Lawrence said.

Thanks to all for the kind words, I have never written any books and have no immediate plans to do so, but I do plan to get back to this thread when I have time. There are an awful lot of basic principles that hardly ever get talked about with this stuff. The people who know them tend to take them for granted, and the people who don't know them don't know enough to ask.

Cheers.

[begin page 2]

> LARRY GATES COMMENT
>
> Simple Addition to all Above
>
> Noise is truthfully not your friend. Learn some simple techniques about Noise Reduction.
>
> Even with some of the best recording techniques, mix leveling techniques, Masking techniques, additive and subtractive EQ, great limiting / compression . . .
>
> If there is lot's of low level / mid level background noise on your lead vox, bkg vox, guitar tracks, samples of drums, any source for that matter, it will multiply, compound itself making ones recording or song suffer. Nothing surgical, but a good idea of minimal noise reduction can go a long way for a lot of people.
>
> This suggestion will not help at all, if one doesn't take the time to read through this thread and take advantage of the free knowledge given. But I can assure you, and any older people here will agree (as there was a time when Noise reduction wasn't even a question as it wasn't even a requirement, it was just THERE).
>
> I'm sure there are free plugins that can get most people there, I would not suggest anyone go out and buy any analog or outboard noise reduction gear, as that won't really help much since our medium is pretty much IN = OUT now.
>
> Also, learn how to use an Expander it's the small cousin of sophisticated noise reduction and it does wonders in our world of Uber Compression.
>
> There is my addition for the world.
>
> That no one cares about.

Larry Gates touched on a very important topic that I plan to get into more detail later. When someone like him says something is important, it's good to listen. But for myself, I still have some very un-glamorous ground to cover before we get into the juicy details of actual recording and processing techniques.

Finished vs. perfect

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.

Setting specific goals

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-guess-

* 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.

ing 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.

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.

Pad of paper

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.

Storage, furnishings, accessories

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.

A place for everything

I'm late for a show, but I forgot something important.

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.

> Quote:
>
> Originally Posted by Lawrence
>
> Hehe... I often wonder why people almost always decide to "re-produce" their music in my studio on the clock. Seriously.
>
> It happens regularly. Go figure.

Yeah, arguably the best reason to record in a professional studio is the organization and division of labor. Partly having someone knowledgeable to deal with the technical stuff, but also just having someone experienced, who can say, "yeah, this will sound good in the final mix", or who can nip in the bud approaches that are going to be problematic.

But of course that doesn't fit into the the tagline "make professional recordings on your computer."

Starting to talk about sound

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.

The value (and problems) of presets

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).

The disappearing bass line

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.

Microphones

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

The best microphone?

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...

What makes a good microphone?

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

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.

"Natural" vs. "produced" sound

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.

[begin page 3]

> Quote:
>
> Originally Posted by Lawrence
>
> Often this is dead true. There are exceptions to that in some genres like folk and classical where the objective is to just capture the performance in a pure form. But yeah, point taken.
>
> Pop music recordings are often like movies, partly an illusion. It's entertainment. Those actors in the movies aren't really doing some of that stuff either, it's part editing and part fakery.
>
> As opposed to a "concert", a stage play.
>
> There's movies and then there are plays. There's pop and then there are live classical recordings. Unfortunately in music, many people (the listening audience) don't always realize how much of an illusion it really is sometimes.

Yeah. I'm trying hard to avoid value judgments, here, because so many of these kinds of threads turn into philosophical debates. If punk rock bands that sound like a barbershop quartet are your thing, then you can still do it better or worse, regardless of whether I think it is a worthwhile endeavor.

And even in "purer" music, or music that does not immediately announce itself as "produced", there is often a lot of illusion at work. Some famous arranger or composer once said something like, "there's no sound in the world as small as a philharmonic." It was said in the context of making arrangement decisions, that if you wanted a big, in-your-face, dramatic sound, the way to get it was with fewer instruments playing better-defined parts. If you wanted a "soft", distant, less-personal sound, the best way to get it was with the wash of a hundred strings. This was someone who really understood the concept of level-matching, whether he knew it or not.

Careful listening bears this out. A close-miked cello or viola can actually have a very aggressive, throaty, ferocious sound that gives electric guitar a run for its money as king of the "power" instruments. In order to get the same kind of power from an orchestral patch, you have to overlay timpanis and cymbal crashes and horn stabs to get the whole orchestra playing one giant power power chord. Which makes a nifty preset on a Yamaha keyboard, but is a completely unrealistic and fairly silly use of an orchestra.

Get a good acoustic guitar player and singer in a room and try to reproduce the performance on "black horse and the cherry tree" by KT Tunstall. Unless you also have a very capable delay or looper running, it's not gonna happen. Which also means that this apparently intimate, authentic-sounding folk track is actually dependent on amplification, i.e. There is no way to "capture" this sound as a pure performance, because it doesn't exist as soundwaves in open air until it's already been recorded, processed, and amplified.

There are some beautiful records that have been made with minimalist far-field miking techniques (and this is still the norm in orchestral and choral recordings), but they do not produce the sparkling, 20-foot-tall acoustic guitars and massive "voice of God" vocals that have become the norm even in a lot of jazz and modern folk recordings. And speaking of far-field...

Nearfield and farfield

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.

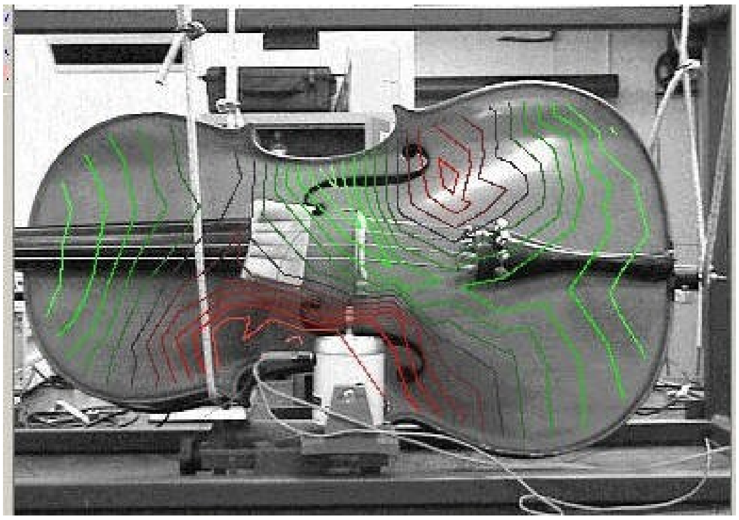
The "nearfield" is the radius within which the sound of the instrument is markedly different depending on the location and angle of the mic or listener. The "farfield" is everything outside that radius. The nearfield of most instruments usually ends at a distance about the size of the main body of the 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?

More to come.

PS: I appreciate all the stuff about writing a book or whatever. For now it's all I can spare to post about this stuff now and then as time allows, and I actually like the back-and forth of a forum, even though there have not been too many questions so far. I suspect Cockos technically owns the copyright to stuff published on the forum, but it certainly doesn't bother me if anyone wants to copy and paste into a word doc or whatever for future reference. In fact it is flattering to be asked. In fact I would love to have a copy, if anyone wants to do the work and send it to me, then maybe someday I can clean it up and put in some diagrams and get the thoughts a little better-organized.

But for now there is still a lot more ground to cover, and I have a feeling that there are some more people with good insights as we get into more nitty-gritty stuff.

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.

Reverb as a farfield substitute?

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.

> Quote:
>
> Originally Posted by junioreq
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> The thing I notice most about pro recordings is that instruments have their own space in the stereo spectrum. You had been talking about bass guitar. One thing I haven't seemed to get yet is how to get the bass to take up a narrower slice of the pie in that field. I hear these recordings where the bass is quite powerful and yet sits in such a small area dead center, almost coming from above.
>
> As I lay a bass, it seems a little wider and unfocused. Guess that would be a good word.
>
> Unfocused. Kinda blurred. What is actually giving these instruments this pinpoint position in the whole stereo field?

I don't want to get too far ahead of myself, but here are some things to think about for now:

- 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?

Anyway, stuff to think about. More to come.

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.

Reviews

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.

Upgrading

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 overbuilt, 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.

Reverb

> Quote:
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> Originally Posted by junioreq
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> I'm broke, no questions on gear. But as far as effects, reverb is killing me. If you listen to this Dokken song http://search.playlist.com/tracks/don%20dokken you hear so many reverbs, I believe. At this point its hard to tell what is delay, what is verb on individual instruments or what is reverb on the whole mix. Seems like I'm having a hell of a time getting all the instruments to sound like they are in the same "space".
>
> ~Rob.

Ah, reverb is a big topic. (isn't everything?)

In normal, everyday life, you almost never "hear" reverb, unless you're in a parking garage or a stairwell. But it's everywhere, and it affects everything you hear on a subconscious level. Even outdoors, the sound is not the same as a close-miked instrument.

Here is an experiment to try. Put on a pair of headphones and listen to the radio. Now, keeping the headphones on and playing, tune a another radio with actual speakers to the same station and turn it on. Turn it off, and then on again. Listen to the difference in sound quality when the speakers are on vs. When it's just headphones. If you're paying attention to it, it's obvious, but it is extremely hard to describe or to put your finger on. I could say it sounds bigger or richer or more natural, but these are clumsy descriptions.

Reverb should not jump out of the speakers as sounding "reverberated." Even massive, lush, 80's reverb doesn't have the splashy, murky, tinny "effect" sound, most of the time. Reverb should be subliminal. Sometimes this is simply matter of turning the reverb down just below the level where you can actually "hear" it (but if you mute it, it still makes a huge difference). But just as often, it is a matter of "tuning" the settings to get a sound that blends in and complements with the dry sound, rather than overwhelming it.

I would encourage anyone interested in audio to listen closely to the Dusty Springfield song "Son of a Preacher Man." You've probably heard this track a million times, but might never have noticed that the only instrument panned center is the vocal (maybe the horns, too, I haven't listened to it in a while). All the drums are hard right, all the backing vocals are hard-panned, and so on. Everything is either hard left, hard right, or center, like a lot of early stereo recordings (believe it or not, the original stereo consoles did not have pan knobs, only switches that went L-C-R).

It's a great mix, featuring a fantastic performance and really good instrumentation and engineering. One really interesting effect that they achieved is that the guitar is panned to one side, but it's reverb is panned to the other. And the reverb is gorgeous, and perfectly-sculpted.

If you listen to the recording closely, The guitar's reverb is nearly as loud as the guitar, but has an extremely muted, "soft" quality that doesn't smear or dilute the guitar at all, it just reinforces it and makes it bigger and richer. In fact the guitar still sounds quite punchy and articulate and "dry." The highs and lows to the reverb are rolled off, so that just the "note" portion of the sound resonates. The decay is "timed" to the tempo of the song, and to the feel of the guitar. This was not achieved with presets.

You really need to dig into the settings of reverb to understand it. A bigger predelay makes a bigger-sounding reverb without smearing the effect. Low-and High-frequency damping make the reverb less conspicuous. Decay times that are "tuned" to the tempo of the song (by ear, not by calculator) fill out the sound without sounding like an "effect." In fact, real musicians in real acoustical space do this instinctively, and adjust what they play and the tempo to suit the real resonance of the space that they are in. People play differently in a bathroom than they do in a cathedral, and they "compensate" for the sound of the space they're in by playing "harder" or "softer."

Setting reverb parameters

Reverb effects in the real world are subliminal.

Predelay conveys a sense of how close to the instrument we are. If we're sitting right next to the instrument in a big venue, we will hear the direct sound immediately, and the reveberated sound a little later (long predelay). This gives us a lot of instrument articulation and sense of immediacy. If we're sitting in the back of a long, narrow cathedral, we might be hearing the early reverb from up front right along with the direct sound (short predelay). This might give a bigger, more "washed-out" or faraway sound.

Decay time tells us something about the size and nature of the space we are in, and also gives information about the volume of the instrument. Very soft sounds decay quickly, but very loud, dynamic sounds can also appear to decay quickly, because the direct sound tapers off quicker.

High-and Low-frequency damping tell us something about the kind of room we're in. An empty cathedral will sound very "splashy" and also muddy with low-frequency resonance. But a cathedral full of people will have a lot more highs and extreme lows absorbed. A living room or soft-furnished nightclub will sound even more muted, regardless of the actual decay time.

"Size" and "Density" controls give us some degree of control over the ratio of "early reflections" or distinct echoes, compared with more "washed out" reverberant sound. In an empty cathedral with lots of stone pillars and hard wooden pews, we are likely to hear a lot of broadly mixed-up, diffuse reverberation (higher density). On the flipside, in a small cinderblock room full of people, a lot of the reverb we hear is likely to be from direct refections off the nearby walls and ceiling (lower density). Again, this exists independent of the decay time or predelay.

For instance, somebody sitting onstage in a basement party with a lot of people might hear a long predelay, very little density, lots of high damping and medium low damping, and a long decay. Someone sitting in the back of a plush nightclub might hear almost zero predelay, lots of low-and high-damping, short decays, and medium density. Somebody sitting in the middle of a massive arena concert might hear medium-long predelay, very low density, and very short decays (because of the surrounding crowd absorbing all the weaker sounds).

This last example leads to possibility of using distinct delays (or echoes) in place of or in addition to more diffuse reverberation. It's harder to find a better example than the stadium rock staple of Gary Glitter's "Rock and Roll Part 2" (which is a bizzare phenomenon unto itself in a whole lot of ways).

In all cases, the above illustrations are not "rules" or "recipes", they're things that have to be tuned by ear. The biggest mistake that beginners make is to flip through presets and stick with whatever one sounds least offensive, or most masking of a mediocre sound or performance.

Monitors revisited

> Quote:
>
> Originally Posted by munge
>
> Great stuff. My own recordings can be described as boomy colored mud embedded in hiss, with ocasional hard-limiting noise. A few items.
>
> “The two most common speakers used in the history of studio recording are certainly Yamaha NS10s and little single-driver Auratones.”
>
> Aren't reference monitors, and all little boxes, seriously unfaithful (you're playing bass through something that no bassist would ever play through)? Aren't they just overpriced imitations of bad speakers that the audience uses? And I'm paying for what, the manufacturer's R&D-ing just how /uniform/ they can deliver the mediocrity? Any problem substituting mediocre old KEF or newer Sony bookshelves? Just like an OK soundcard, they too can convey some of the innovative brilliance of a good recording.

First off, great questions from a first-time poster. And my guess is that for everyone who actually posts a question, there are probably a hundred others wondering. And it's hard for me to tell whether I'm moving too fast or too slow without feedback, so kudos.

If you find a set of bookshelf speakers that work well as monitors, go for it. The proof is in the pudding, as they say, not in the price tag nor in the label or brand designation. The pudding, in this case, is NOT your ability to make good-sounding records on that set of speakers, nor in the speaker's ability to convey the innovative brilliance of the recorded music (the brilliance or lack thereof is in the performance, not the speaker). The pudding is when you are making records that sound consistent, balanced, and essentially the same on every other speaker system.

When you listen to a commercial recording, it pretty much sounds the same whatever speaker system you play it back on – in the car, in a bar, on headphones or at Redbone's. That does not mean that the sound quality is not affected by the speakers, just that the mix and the underlying recorded material itself sounds like the same material, just played through different speakers, and ideally it sounds pretty good on everything. But if you have ever mixed a record on headphones or on a home hifi system, I bet you have experienced the effect of popping the test CD into a friend's car or your girlfriend's home stereo and hearing something that sounds totally different from what you mixed at home.

The bass is way off, the balance of instruments is all screwed up, you can't hear the vocal (or it's way too loud), the cymbals either sound pingy or like white-noisy trash – in short, nothing sounds right. It sounds like a totally different mix from what you had at home.

The reason for this is that most home systems these days are designed to alter and flatter the sound in frequency-, dynamics-, and phase-dependent ways. An obvious analogy is the kinds of "SRS WOW" effects and sonic maximizers/aural enhancers that are built into a lot of mp3 players and consumer electronics to hype the sound in various ways. Speakers are very often built the same way, and frankly this is actually worse for reference monitoring than simple "bad speakers." If you luck out on a set of inexpensive consumer bookshelf speakers, it will very likely be something pre-1990, from before CDs ushered in the new wave of inexpensive hi-fi, or else something specialized at the low-end of the dedicated "audiophile" market.

My experience is that Sonys and the like (even in the $300+ range) are going to be chock full of one-note-bass, big directional distortions that interfere with near-field listening, crossover-frequency-related distortions, inconsistent frequency response at low volume, and smiley-curve "hype."

It wouldn't be my first choice, but I'd be okay with doing a record on Tivoli audio speakers if I had to. And Wharfedale Diamonds are supposed to work well. But those are already in the price range where you could just buy a set of Behringer Truths or something. I don't have a lot of exposure to low-end monitors, but they are probably made with at least a minimum level of faithful reproduction as a design goal, and for most people, buying an inexpensive set of dedicated-purpose reference monitors is probably cheaper and a lot faster than buying a dozen different sets of cheap bookshelf speakers and doing test mixes to see which if any work well as monitors.

You can of course try anything, and it's always better to get busy with whatever you have available than to stress and second-guess your gear, but If you find that your recordings are not sounding as good on other speakers as they sounded at home, or that you are having a hard time hearing the effects of subtle eq or compression, then monitors are the first thing to put on your shopping list.

With specific respect to NS10s and auratones, obviously the ideal monitors are probably better speakers than these, and if you can afford ADAMs or custom soffit-mounted $30,000 monsters, then go for it. But my guess is that most of those reading this thread are probably on a tighter budget. My point with the NS10s and auratones is that "great-sounding" speakers are not necessarily even desirable for reference monitoring. NS10s sound like "perfect" cheap speakers. And that means that they sound the same at low volume or high, they deliver consistent nearfield frequency dispersion, they do not compress or "hype" the sound, they deliver bass response that is focused and tonal down to the cutoff frequency, and they have a clear, even midrange.

None of the above applies to most consumer bookshelf speakers, even "good" ones, which are apt to have sloppy dispersion, "loose" bass response, very different frequency and dynamics response at different volume levels, and a midrange that is designed not for accuracy but to compensate for crossover distortion. It is really important to understand that none of this necessarily translates into "bad sound." In fact, for home listening, any of these might actually be desirable "features." But they're not good for reference monitoring.

As an aside, I'm going to touch on your example of "something that no bassist would ever play through", since it raises a great point. The surprising reality is that a majority, or at least a significant plurality of bass players play through exactly these speakers when it comes to modern studio recordings. The whole idea is that we are making records suitable for living-room listening or something similar, and standard practice is for the bass player to sit in the control room and either plug straight into the board or to hear the miked bass cab through the control room monitors. For purposes of the recording, this is exactly the sound that we care about. But even if the bass player is out in the live room playing with the band and hearing her amp sound, what you care about as the recordist is the sound as it's being captured, and how it translates in real-world playback.

Level-matching revisited

> Quote:
>
> "Level-matching" does NOT mean making it so that everything hits the peak meters at the same level."
>
> That's what the red lights on analog meters are for. I get the advice of, don't overdrive an input, and analog was more forgiving, within limits. But what are you really saying to do with this information? How and when do we do the balancing act? Some combination of gaining up the dry-ish strat and/or dialing down the overdriven Les Paul, yes? Limit and compress the high peak-to-average channels, like the dry strat? If so, when? When capturing the performance? At mixdown? Somewhere in between? Dial down the low peak-to-average channels, such as the overdriven Les Paul? Again, at what stage? Which brings us to...

NonononononoNO.

This "level-matching" that I'm talking about has nothing to do with any console or DAW meters, analog or digital, clip, peak or RMS. It is totally about the volume of sound in open air at the listening position. Neither REAPER nor any other DAW or mixing console has any meter for this, and they could not. I am talking about the actual perceived volume level after the sound has left the speakers. I'm talking about the sound pressure changes in your ear canal, not in the recording system.

When you have that band in a room with the clean Strat and the dirt Les Paul that I described above, the Strat player is turning up his amp and the Les Paul is turning up his amp until they both sound about right compared to he drum kit and everything else.

Nobody is looking at meters or thinking about peak level or clip lights or any of it. And NOBODY is compressing or limiting the sound to make it fit with preconceived notions of what the recording meters are supposed to look like. That is the OPPOSITE of where good sound comes from. Real musicians play at varying volume levels and have sounds and instruments that are dynamic and exciting and that do not fit into a preconceived 12dB window, and nor should they.

So how do you mix this record? Easy. TURN THE LES PAUL DOWN. There is NOTHING wrong with starting out with the Les Paul peaking at -15dB. FORGET THE METERS. If it sounds too quiet, turn up the volume on your SPEAKERS. HEADROOM IS YOUR FRIEND. It is what makes the Strat sound punchy and dynamic.

I haven't even begun to talk about compression, and nobody who is unclear on any of this should be TOUCHING a compressor yet. Start your mix like a band in a room. If it's a rock combo, the loudest fixed-volume instrument is drums. So pull up those faders first, and set the drums so they they are peaking at say -6. Now turn up the guitars NOT according to the meters, but according to the SOUND relative to the drums and to each other. TURN UP YOUR MONITORS if you need more volume. Really. It's EASY. DO NOT OVER-THINK THIS. Just do it.

Compression comes AFTER. And it is a huge topic. But for now, just record good signal, and then mix it to taste. Just mix it. They are sounds. Mix them together. If one instrument is too loud, turn it down. If another is too quiet, turn it up. If the signal is clipping, pull back your faders, and start over WITH YOUR SPEAKERS TURNED UP LOUDER.

Erase the parts of your brain that think of compression and limiting as a way of making things louder. Now re-write those parts of your brain to think of compression as a way of making things QUIETER, because that's what it does. When it comes to compression, start loud, and then see how much quieter you can make it before it sounds bad. NOT THE OTHER WAY AROUND. Compression does not make anything louder, it makes things quieter.

If the above does not make perfect sense, then just leave compression alone for now. If your records sound quiet, turn up your volume knob.

This thread could go for 100 pages and years, and there is a lot more to come. As I said earlier, there is a lot of back-and-forth to this stuff. Gain-staging is a big topic that we've barely touched on. Compression is a HUGE topic that affects everything, but all in good time.

Sound waves

Before we get into compression and gain-staging (both closely inter-related topics), it is important to understand some basic concepts of audio and acoustical wave forms.

Sound waves are "AC" or "alternating current." In electrical terms, this is similar to a battery with a switch that rapidly changes the polarity from positive to negative. In an ocean, it is like waves coming in and out, pushing and pulling. "DC" or "direct current" has no sound. Acoustically, it's just static air pressure. Unless the pressure is disrupted, we don't hear anything. A very sharp "DC" displacement of air pressure such as a hand clap creates ripples similar to throwing a pebble in a pond. Those AC "ripples" are what we hear. Like ripples passing a fixed spot on the surface of a pond, they pass right by us and dissipate into the ether, and we only hear the quick passing as a sharp pop. But most of the musical sounds we are interested in are more steady, fluctuating changes in air pressure.

You can perform a simple experiment to generate low-frequency changes in air pressure by simply waving your hand up and down very close to your ear. If you wave your hand very quickly (say 20 times per second or more), you'll hear a very low-frequency tone or rumble. You have to keep the amplitude (up and down distance) fairly small, or you will start to generate actual wind or puffs of air which will mask the tone, but if you just wiggle your hand over a very short distance close to your ear, you'll generate something like a 20Hz tone without creating actual wind or moving air currents, just changes in air pressure.

This is essentially how the human voice and all other instruments work. When we sing, we are passing some air out of our lungs, but that's not what is actual generating the "sound", it's just carrying it out into the world. The actual changes in sound pressure that create tonality are from vibrations in our vocal chords, which fluctuate very rapidly. This modulates the "wind" as we exhale, and the current of air carries a steady-state alternating pressure that those around us hear as mellifluous song (or as wretched screeching, depending on our skill and their tastes).

Electrical audio works the same way, except the current carried is positive and negative electrical current instead of air pressure. If you could connect a wire to ground and somehow switch a battery's terminals from positive to negative 20 times a second you could generate a 20Hz audio signal the same way you created a 20Hz acoustical signal above. (the distiction between "audio" and "acoustical" is that "acoustical" is what happens in open air, while "audio" refers to captured or processed sound signals in electrical or digital systems. Make sense?)

In a very simple transducer system such as a guitar pickup, you have a coiled, magnetized wire (inside the pickup) next to a vibrating metal string. The vibrating string pulls the magnetic field, which causes electrons to move back-and-forth across the coiled wire. The coiled wire is connected by leads in the guitar cable to the preamp, and the faint electrical current caused by the disruptions in the magnetic field is sent down the lead wires to the preamp where a transformer increases the signal voltage to something usable called "line level."

This amplification process is like a second pickup. An oversimplification would be to imagine a strong DC current (like the air from a singer's lungs) being modulated by a weaker AC current that modulates the stronger current, amplifying it. If we imagine weak ripples in a pond being used to wiggle a floating paddle, and that paddle connected to a lever that makes bigger waves in a nearby river, you can start to get the idea.

A dynamic microphone capsule works the same way. Instead of a pick vibrating a string, acoustical sound pressure changes are caught by a disc-shaped "diaphragm" that moves in and out. The diaphragm is connected to a magnet that is suspended inside a coiled-up wire. As the magnet is pushed in and out by alternating pressure on the diaphragm, a small current is generated, just like a tiny electrical generator, powered by air pressure. This is fed to an amplifier, and if we pretend that there is only a single amplification stage, the tiny current from the mic cable creates the same kind of electro-magnetic disruption in a much bigger coil of wire powered by bigger current, which drives speakers, which are much bigger transducers that have the exact same design as the microphone. Only in this case, instead of being moved by air pressure, the magnet in the coil is moved by the powerful current in the coils, and speaker cone is pushed in and out, creating alternating sound pressure waves.

Having a rudimentary understanding of the basic mechanics of sound will become valuable as we start to talk about some of the technical details of modern studio recording.

Monitors revisited

> Quote:
>
> Originally Posted by drybij
>
> yep – since the job of a recording engineer is to make a recording sound good on a wide range of speakers, and my impression is that the main difference between speaker enclosures is the frequency curve, I've pictured the mastering process as a a sort of "averaging" or "balancing" of the recording so that it's in the "sweet spot" of all these different frequency curves. Is that a somewhat accurate statement? If so, then by disregarding commercial appeal is it possible to get a pristine, killer reproduction of a recording if we custom-master the recording for a specific set of speakers?
>
> Not meaning to derail the thread, just curious.

Uh, sort of. "Custom-mastering for the speakers" is, in a sense, what happens when you mix on inaccurate speakers. But it's not just a question of frequency, it's also got a lot to do with things like the speaker gating or compressing certain frequencies.

Example 1: If the speaker is built with a tight woofer suspension, this can give a much thumpier, tighter low end, which sounds good for listening. But it also disguises any sloppyness or mud in the underlying mix, and it may lead you to crank up the low end just to excite that cool "thump" from the speakers.

Example 2: If the speakers are built with tweeters that are very sensitive but that limit excursion (volume) to avoid damage, then any highs might be "sexed up" and compressed on playback. So a pingy, clangy, uneven ride cymbal comes out of the speaker sounding like splashy sizzle and you don't know what's really going on behind there until you take the mix to a different set of speakers.

Example 3: Let's say your Sony system has a crossover at 1.5kHz (a very common place for it). This is an extremely sensitive range of human hearing, and any ugliness around it is going to sound bad. So the speaker designer bypasses the problem of crossover distortion by simply designing a crossover that depresses all frequencies around 1.5k, like an eq cut. The neat thing about this approach is that cutting the mids like that is like a "loudness" circuit, and not many customers are going to complain about too much highs and lows. Let's further imagine that Sony thoughtfully included a free stereo widener circuit to make this little bookshelf system sound bigger and more dramatic, so not only are frequencies around 1.5k depressed, but so is anything in the center of the stereo spread. Now, what might be panned center with important content at around 1.5k, hmm? Maybe vocals? Snare? Kick? Bass? Only the most important instruments in the whole mix.

So you end up "mastering" the hell out of these critical instruments at critical frequencies, and then play it back on another system and the whole mix is totally out of whack.

The important thing to understand is that NONE of those effects are necessarily going to interfere with anyone's enjoyment of material that was well-mixed to begin with. Take any commercial record and play it back through a system that delivers thumpy lows and sizzly highs and a wide stereo spread with scooped mids, and almost nobody's going to complain. But it's like one-way glass – good sound can still get OUT of the speakers, but you can't see IN to tell what's going on with the underlying audio.

It's perfectly okay to listen to music on a system that adds thump and sizzle and size, and the music you listen to does not have to be mastered specifically for that speaker – the speaker is basically "re-mastering" everything that goes through it: gating the lows, compressing the highs, depressing the mids and center. Decision-making becomes a crapshoot on a system like this. You just can't tell what's going on.

There is no such thing as perfect speakers. You might get 90% of the way there for $200 or whatever, but getting closer and closer to perfection drives up the costs exponentially. There is a small market for speakers and other kinds of audio gear that are overbuilt and over-designed in every way, and there are people and businesses who will pay whatever it costs to get as close to perfection as possible, even if that extra 1/10th of 1% means a hundredfold increase in cost. The fact that this market is small and that the producers of ultra high-end equipment are small boutique manufacturers means that the market and the production does not benefit from the economies of scale that drive down the cost of humdrum consumer goods. Plus there is a fair amount of fluff, superstition, and nerd cachet at work.

I don't want to get too far off-track, but the very best speakers **are** more expensive to design and produce in a whole lot of ways. Whether the difference is "worth it" sonically or otherwise is a separate question.

[begin page 4]

Quick note before proceeding...

As I mentioned earlier, there is a lot of back-and-forth and inter-dependence to this stuff. As much as we try and isolate different aspects for discussion and analysis, ALL real-world sound has dynamics, noise, reverberation, standing waves (even just the ones in our eardrums), absorption, harmonics, and so on. And all real-world audio similarly has some of everything that we might talk about.

There is no way to talk about one aspect at a time without glossing over or assuming a lot of other relevant stuff. So whether you get it from this forum, or a book, or magazines, or independent research, it is usually most valuable to work through the same concepts multiple times. The "aha!" moments often come when re-visiting one topic after having picked up a smattering of others.

So read, think, and most all LISTEN to everything around you, and then be prepared to read, think, and listen some more.

This is specifically prompted by uncertainty on part over whether to talk about dynamics or gain-staging first, but with the idea of "begin at the beginning" in mind, we'll start with gain-staging.

Gain-staging and noise

"Gain staging" is a super-critical concept that unfortunately gets short shrift in the digital era, which leads to a lot of frustrations among young recordists who do not realize the effects it can have.

Let's set aside digital for the moment and pretend that we still live in an all-analog world. When you walk into or see pictures of an old-school professional recording studio, there are thousands, maybe millions of knobs, switches, faders, meters, and blinking lights. Almost every single one of those corresponds to some kind of signal amplification. In a typical commercial mix there may be literally thousands of stages of amplification or "gain" captured in the final mixdown, when you count all the preamps, processors, instrument amplifiers, and mix decisions. And still pretending to be in an analog world, EVERY SINGLE ONE OF THOSE AMPLIFICATION STAGES HAS A "SOUND." And whether you got them all right or wrong is going to have a big deal to do with the quality of your recording.

For example, let's imagine a super-accurate, extremely sensitive preamp designed for sparkling, dazzling, likelike headroom. Big transformers and power rails for massive headroom means slightly higher internal noise, but whatever. We'll use that as our default preamp. We added a tiny bit of hiss, but otherwise have fairly pristine, unaltered capture. Let's call this preamp the "CRYSTAL PALACE" when we talk about it later.

Now we want to EQ the track a little, maybe subtractive EQ with makeup gain from our warm, chunky-sounding vintage mixing console. This hypothertical gain stage is very low noise, but part of that is because it fattens and flattens the sound a little. That's the "warm" part. The "chunky" part comes from having a slightly slower response and slew rate than the super-accurate preamp used in stage one.

Overall, this gain stage has a neat effect of ever so slightly gating and compressing the sound, which might even slightly reduce the hiss from above, but probably won't increase it any (unlike if we had used an additional stage of gain from the first preamp). Let's call this one the "FATBACK." Next, we add some compression to tame the peaks and even out the overall level a bit.

Here we might decide to use a tube-based "character" compressor, one that adds a litle harmonic "fire" to the signal, to up the growl and presence a notch. This stage of amplification uses extremely high internal votages to power the tubes, and is likely to introduce a smidgen more hiss, and it also a more reactive and non-linear approach to dynamics. In fact the output of such a compressor might actually have HIGHER peaks than the input, because of slow attack times and makeup gain. But that's okay, we're going by ear, not by the meters. Let's call this guy the "INFERNO." Next we're going to send the signal to tape, which is effectively yet another stage of gain.

How hard we hit the tape can have a big efect on the sound. Tape is about the hissiest thing in the studio, so we want to stay above the noise floor as much as possible, which is one of the reasons why it was so common in those days to compress BEFORE tracking, because any compression after tracking will reduce the signal-to-noise ratio.

Another aspect of recording to tape is that the higher in signal level we go, the more peaks become compressed and saturated. At extremely high signal levels, it sounds like the direct out of a guitar distortion pedal (in fact you can make a great distortion effect from the guts of a cassette player). At moderately strong signal levels, you get a very smooth, natural, musical compression – that infamous "tape warmth." But simple "warmth" is not all there is to it – we cannot undo anything done previously to the signal, and HOW we hit the tape counts just as much as HOW HARD we hit the tape. It is very probable that putting a little low-shelf cut BEFORE we record to tape and then a corresponding low BOOST AFTER tape will come out sounding different than if we just left everything flat. The tape saturaton would be embedded in the highs and the midrange, without causing the low end to "fart out" as might happen if we sent the whole signal through unaltered. So we could get a little fire and saturation in the presence range without losing clarity and impact in the lows.

And we could apply this to any eq, compression, reverb, or other processing that we did before or after ANY gain stage. THIS REALLY MATTERS, so re-read or ask questions if it's not making sense.

Continuing on, let's say that mixdown time comes around and the engineer just decides to go crazy on this track and try and get that kind of lo-fi, band-limited, telephoney crunch of listening to something really loud on a cheap cassette walk-man. He finds some device to fit the bill that just overloads the hell out of itself. It hardly matters whether it's an eq, a compressor, a preamp, a stompbox, or whatever, because it's pretty much doing all of it whether it means to or not. We'll call this one the "CRAPOMETER." He probably would not run the entire mix through this device, but for one instrument that's having a hard time fitting in the mix it might be just the ticket.

So far so good. Now, let's talk about how each of those gain stages are inter-related. Think about the characeristics of "CRYSTAL PALACE" and see if this makes sense: There would never be any reason to use crystal palace AFTER any other of the devices in the example above. It can never restore clarity or lost dynamics, it can only capture what was already there, plus hiss.

Placing FATBACK after the INFERNO would probably not achieve the results we were after, unless our intent was to subdue the effects of INFERNO (i.e. We realized we made a mistake and overdid it). If INFERNO hypes the sound, FATBACK mellows it. I.e. Fatback kind of undoes the effect of inferno, but the reverse is not true. This could lead to some frustration if you got it wrong before recording, because simply-rerunning it through the INFERNO might not restore the same result – it might just give a more strangled, fizzy version of the duller FATBACK'ed sound. And the CRAPOMETER simply cannot be undone.

The signal chain we described above makes a kind of sense: take a pristine signal, chunk it up and fatten it a little, then fire it up to maybe restore a little impression of clarity and "cut." But rearranging the components doesn't work the same way. This is NOT a recipe where the order of ingredients doesn't matter.

Similarly, HOW we use each of those gain stages matters A LOT.

The CRYSTAL PALACE preamp, with its super-sensitive modern transformers and massive power rails might well offer tons of crystal-clean headroom, but if we push them to the point of actual overload, they might actually crap out pretty badly, like digital clipping.

On the other hand, the preamps on the FATBACK console, with their slow, burly, heavy-wired Soviet-era transformers might be nigh-impossible to overload. They might just get fatter and chunkier the harder the harder you push them. At some point they might get TOO fat, but they won't give the crackly nastiness of outright clipping, they just round off the edges of the sound.

Similarly, the INFERNO and the CRAPOMETER are likely to change sound radically depending on how hard they are pushed. Both of these are heavy "character" devices that have a lot of subjective middle ground, like tape saturation.

Analog circuits have electrons moving across copper wire, or across a vacuum, or jumping across coils in transformers, or getting stored and discharged in capacitors, or squeezing forcefully through resistors, and so on. These processes result in phase-, dynamics-, and frequency-dependent alterations in the output signal (distortions, in short). Small amounts of inevitable randomness in the movement of electrons produces hiss, and induced magnetic and electrical disturbances produce hum and radio static and other kinds of noise.

And the copper (or whatever) conductors themselves have capacitance, resistance, reactance, and all the rest of it. There is no free lunch. If we used massive industrial transformer like the power company does, we could have essentially infinite headroom, but the self-noise of such a system would be off the charts, or else it woud have to be a system the size of a house with every component shielded in a lead box. And even if money and size are no object, the length of wire runs in such a system would cause losses in regular line-level signal, unless we specially constructed a system that ran with 200 volt signal, in which case we're right back where we started because now our 1,000 volt transformers can only handle 6.2dB of headroom. So now we're upgrading to 20,000 volt transformers and much heavier (more resistive) wire, and back to the signal attenuation problem.

Everything is a tradeoff. This is why top-flight hardware is so expensive. The closer you get to "perfect", the more you run up against the laws of physics. The great wizards of hardware design, the wild-eyed, chain-smoking, sleepless, obsessive electrical engineers who labor away in basement workshops building gear for mail-order so esoteric that even the wife and kids don't know what dad is up until the day comes when some marquis producer decides to outfit her entire studio with the stuff this guy is producing... these people are constantly threading the needle between noise and headroom, between accuracy and flattery, between fidelity and desirability.

It is all well and good to speak of a "straight wire with gain" as the ideal preamp design, until we consider that it is impossible, and that a straight wire itself has a sound, and that gain itself has a sound, and that virtually zero popular music recordings are intended to have the "neutral" sound that "straight wire with gain" theoretically employs. And this is where the magical, "musical", sound of the best analog equipment comes into play. The very best devices are forgiving, intuitive, natural-sounding, well-suited to downstream processing, and whatever personality they have hits a "just so" note that seems to work great for all kinds of stuff.

More affordable, second-tier gear might also be very good, but might be for instance a little more limited in application. For example a second-tier prosumer "FATBACK" preamp might be just the ticket for drums, but all wrong for overheads. An "INFERNO" might be awesome for synths, bass, and power vocals but totally out of place for orchestral recordings or soft crooning ballads. A "CRYSTAL PALACE" might be brilliant for small jazz and acoustic combos but hard to process and unforgiving for garage rock or hip-hop vocals.

Suss out your gear

Bringing this all back to home-studio applications...

Every single analog process in your studio has a "best" setting. Even if you consider yourself to be "all digital", your preamps, mics, speakers, amplifiers, and instruments are still analog. Even your converters have an analog front-end with a bona-fide copper circuit that handles analog signal.

I want you to go dig out the documentation that came with your preamps, soundcards, hardware effects, mics, and so on (which will be easy if you have organized your studio, as above). Now get an exacto knife or razor blade, scissors at least. Got all that? Good. Now with the exacto knife, carefully cut out all the portions that talk about frequency reponse and THD+N and every other spec, file them all in alphabetical order, staple or paper clip them together, and throw them in the trash.

Now that you have documentation that talks accurately about what your gear is capable of, it is time to suss out your gear. Your preamps will sound different at different gain settings. So will everything else. Mics will sound different when recording louder or quieter signals, from closer or further away. And the type of signal you are putting through them matters.

Especially if you are working with inexpensive preamps, it is almost a certainly that some will sound better than others, or at least different on different gain settings (even in the same physical box). Maybe the ones closer to the transformer sound different. Maybe one that has a slightly off-spec capacitor or resistor sounds different. Maybe the first ones to tap off the power rails sound different when you're recording multiple channels. It is very possible that some channels on some instruments will sound best when you set them well below the threshold that would be indicated by your digital clip or peak meters. This is especially true of low-frequency instruments and highly dynamic instruments, and especially true if you are using more than one channel at a time.

If all of this sounds hopelessly complicated, it's not. Take deep breaths, close your eyes, forget about what you paid for anything, and repeat ten times "all you need is ears (and level-matched listening)."

Here's a specific and very relevant tip: any active instrument (e.g. a bass with active pickups, or an outboard synth) is apt to sound very different when plugged into line inputs vs "instrument" inputs, or when used with a DI box. Try them all.

Professional studios with loads of gear have long-since gotten over brand anxieties. In one recent session a cheapo behringer mixer was selected for preamps over a very lush, well-respected tube preamp on a piano recording. It just sounded more appropriate. Well-equipped engineer often have favorite channels to plug into on the mixing console, and they have the massive gear selection not because more expensive is invariably better, but because different gear sounds different, and a restaurant needs to have all the ingredients.

The point is not a clinical evaluation producing detailed charts that you have to look up or think through, the point is to LISTEN to what you are recording and fix it until it sounds right, or at least as good as you can get it EVERY STEP OF THE WAY. This process is actually a lot faster and easier than trying to fix it later.

None of this means that you have to try every mic through every preamp on every gain seting on every track you record. I think the soul-suckingness of such an approach would actually be counter-productive. What it means is to take nothing for granted and to let your ears guide you, not your preconceptions. Trust your instincts, not your documentation. If something isn't sounding right, try something else, even if it seems stupid. Actually, nothing should seem stupid in music.

Some of the stupidest things have been the most successful in history. And not just commercially for teenyboppers, either – think about the foundational melody from "Ode to joy", probably the single greatest piece of music in history. A lot of graduate students in composition would be embarassed to build a piece on such a singsong, rudimentary melody. If that's not your cup of tea, think about the real essence of say, "A Love Supreme" or even "Love Me Do."

The relationship between conception and execution, between inspiration and perspiration is often vastly different from what we imagine. Genius is in the details as often as it is in the big ideas. Maybe more so. But it is the works that ignore the details and focus solely on the conceptual ideas that come out clumsy and sophomoric. And the cool thing about the details is that they are relatively easy. All you need is ears.

> Quote:
>
> Originally Posted by junioreq
>
> Just a quick Q. Would you consider guitar pickup position and height to be important to the staging? I usually run my pickups as high as I can get...
>
> ~Rob.

Uh, yeah. Extremely so. Probably just as important as the kind of amp you use. And exactly the right kind of question to be asking yourself.

And on the topic of electric guitar, do not take your tone or volume knobs for granted. The onboard electronics on a guitar are VERY reactive. For example, the classic "woman tone" of a guitar on the neck pickup with the tone knob rolled all the way down (see Clapton, Slash) sounds vastly different through an amp with the treble cranked and the bass knob way down than a guitar set to a treble pickup with the amp at even eq settings. The difference is NOT subtle.

This is EXACTLY the kind of stuff I'm talking about. In the analog world, turning a signal way up and then way down in a later stage ALWAYS sounds different from turning it way down and then way up. And whether it is eq'ed or reverb'ed or compressed or whatever before, after, or in-between this process matters.

Coming back to digital...

WITHIN a modern DAW like Reaper, gain itself is essentially pure, clean, and soundless.

You could mix all your tracks so that the individual track meters are like +50dB and totally redlined, and as long as the master output is turned down so that your DA converters don't clip, it will sound basically exactly the same as if you had mixed everything at -50dB and then turned up the master out to compensate. There IS a limit to this, but in a 64-bit mix engine, it is so far outside the realm of sane real-world work practices that you can basically pretend it doesn't exist. But it is probably better practice to keep your tracks in normal ranges, if for no other reason than that the controls and meters are much more useful and intelligable when you're working with tracks that are running around -20dB steady-state or so.

HOWEVER, when we get to plugins and processing, the same principles are still very much in effect. EQ before a compressor sounds different from eq after a compressor. Maybe only slightly, maybe not. Compression after reverb sounds a LOT different than compression before reverb. And the more you work with analog-style "saturation" effects, the more these things are true.

The big thing is that stuff that happens earlier in a signal chain cannot be undone later in a signal chain. Going back to the "ideal preamp" discussion above, one of the things I mentioned was a "forgiving" sound that is easy to process. It is very hard to add back clarity and depth to an overly "FATBACK" sound. Turning up the highs is likely to bring up steady-state hissy fizz if the high-end dynamics are dead to begin with. Turning up the lows just increases mud if the deep dynamics have already been squashed. Attempting to use reverb to smooth out a harsh sound might just result in metallic splashies.

One of the ironies of this stuff is that sometimes the only solution to "too much" is to dial in "too little." For example, if you recorded a vocal with a shrill, brittle, essy high-end, your only solution might be to dial in a duller, flatter, sound than if you had simply recorded a smooth, midrangey vocal to begin with and then shelved up the highs. If you recorded an overloaded, farty bass in an over-enthusiasm to get big lows, you might end up having to roll off all the lows in order to get the bass to fit in the mix.

This is what we mean by "don't plan to fix it in the mix." It doesn't necessarily mean to try and hype up all your sounds at tracking, it means to get GOOD sounds, FORGIVING sounds, WORKABLE sounds. Sounds that are a smooth and natural representation of the source, without any ugliness.

Trust your ears, and LEVEL-MATCH your AB comparisons. Make sure you are focusing on better and not louder, EVERY STEP OF THE WAY (golden ears in one easy step, really).

One more post on this topic before we get into noise, especially for the home recordist...

It is often hard for the beginner (or even the old pro) to distinguish between "good" saturation/distortion and bad. This is especially true on full-frequency stuff like electric guitar, snare, organ, bass, massive synths, and rock power vocals. If you're recording a cranked Marshall stack it can be hard to hear the effect of the mic diaphragm flattening or the preamp overloading in the vortex of steady-state tube distortion that you are TRYING to record.

But it really fucking matters. Because the full-throated Marshall roar is NOT the same as the strangled, clipped sound of a flattened mic diaphragm or the buzzy nasal fizz of an overloaded transistor preamp. And these things WILL make themselves known in the mix, even if your Marshall-deafened ears couldn't hear them while playing the thing.

Similarly, if you are recording yourself singing through headphones then what you are hearing is likely to be the smooth, dull, bassy, inarticulate sound of your own voice, with your ears blocked, PLUS whatever is coming through the headphones. This may lead you to record an overly hype, brittle, saturated, presence-rangey sound of your own voice.

I plan to post some specific approaches for specific instruments and voice later, but for now the most important thing is to be aware of these effets, and on the lookout for them.

You don't actually need my approaches or tips (all you need is ears, remember), but you should be taking it slower and listening more critically and giving your ears frequent breaks if you are both the performer and engineer.

Noise

As Larry Gates put it in an earlier post, noise is seriously not your friend. Noise is anything that you DON'T want in a signal, but the most common culprits are 50/60 cycle hum, hiss, and low-end rumble.

Hiss is the most common and least egregious kind of noise. In fact, tape hiss can be a little soothing to listen to, at low levels. But let the listener put on their own hiss machine if that's what they like.

Hum is the most obvious and offensive kind of noise, and the leading culprit is single-coil guitar pickups, followed by unbalanced mics and a handful of older keyboard instruments that lack balanced connections. The last two are so uncommon that I'm not even going to address them. Hum that comes across anything else is a whole nother topic.

Low-frequency rumble is nasty and devious stuff that is often inaudible on conventional monitors but that devours headroom and causes dynamics processors to work in unexpected and often unpleasant ways.

Taking the above in reverse order, from most specific to most general solutions...

Rumble

Rumble is usually noise picked up by mics and/or electrical signals that is below or almost below the threshold of audibility. Passing trucks, handling a mic, appliances running in the basement, people walking on nearby floors, planes flying far overhead... all of these things can produce very low-frequency soundwaves that are practically inaudible and often too low to be reproduced by your speakers. But they still eat up headroom. Even very quiet sounds at 20Hz can use up a LOT of energy, and can cause inexplicable clipping when you try to turn up affected tracks that sound too quiet.

The simplest solution to rumble is to use high-pass filters on every track. As I mentioned in an above post, frequencies lower than what your monitors can produce are often not all that necessary or desirable to have in a finished recording anyway. And a gradual high-pass filter set to say 40dB actually DOES still allow a significant amount of content down to 20Hz and even below. You could do a lot worse than to simply get in the habit of high-passing until a track sounds bad, then backing off just a smidge. Especially for anything that is not a bass instrument. Not only will this clear up rumble, but it will also clear up mud and undertones on non-bass instruments, giving you more room for a clean, tight, punchy low-end, and more headroom so you can make a “hotter” mix without compressing and limiting everything to death.

An even easier solution to rumble that is also generally good practice is to decouple your mics. This means shock mounts, floor pads under mic stands, anything that keeps sound from being transmitted through anything other than mic diaphragm vibrating in open air.

That way what you hear is what you get and the water boiler in the basement doesn't rumble up through the floorboards and mic stand. Padded carpet works great.

Hum

Hum is a very ugly kind of noise. A little "hum up" in the intro of a track to give a "garage" feel to the lead-in of a song is one thing, but incessant, droning hum is off-putting and unpleasant to listen to and makes for a very bad-sounding recording. Especially if you have lots of stacked tracks of guitar. Everybody hates it. Guitar players who have become deaf to it or who think it's just "part of the sound" frankly need to pull their head out. It sucks.

Fender guitars can be shielded pretty easily with either copper foil or even heavy-duty household aluminum foil. If you're comfortable working on your guitar, just unscrew the pickup cover, take the whole thing apart, and glue a bunch of foil into the entire body cavity and over the whole inside of the pickup plate, making sure the two will overlap the screw holes when you put the cover back on (ie the guts of the guitar will be totally enclosed by metal). Connect it via another strip of foil or wire to the ground pin of the guitar jack and viola! Massive hum reduction. Why they don't come this way is beyond me.

Google for more detailed instructions, I'm sure. I disclaim all responsibility if you damage or discolor your vintage strat with bad glue or a hack job, so do your homework first.

Passive or humbucking pickups obviously offer a more direct solution, but they also change the sound.

Other hum-producers are flourescent lights, lighting dimmer switches, and motors of any sort, including fans, air conditioners, refrigerators, and anything else that hums or buzzes while running. It may not be enough to simply have these turned off in the recording room, because any that are running on shared circuits will still send hum along the ground lines that your gear uses for reference. If they are on the same fuse or circuit breaker, they should be turned off while recording. Also, as much as possible, mic and signal cables should be kept away from power cords, and/or should cross at 90 degree angles (should not run parallel).

Hum from electrical can also be reduced by what is called "star grounding", or using the same ground point for everything that shares a signal path. In simple terms, this means clever use of power strips to make sure that everything that is physically connected in a signal path (i.e. Guitar amp and effects rack, but not necessarily mic preamp and computer) are ultimately plugged into the same outlet. Please use UL-listed surge-suppressing power strips for this purpose. Do not use “ground lift” adapters or cut the third prong off your plugs. They are there for a reason, namely to keep your studio/home from burning down. If the place does burn down because you lifted grounds or cut off prongs, insurance will not pay the claim. I'm not kidding.

But the worst hum producer in most home studios is CRT monitors (and TVs). If you don't exclusively use LCD flat-panels, now is the time to switch. They use a lot less energy, are much lighter and smaller, and cheap. And they don't hum. If you cannot afford a new monitor right now, put it on your wish list and turn off the CRT monitor while recording. Down goes the hum.

Hiss

Hiss is the sound of random electrons moving around electrical circuits. Better-designed stuff has less hiss, but hiss is the most treatable and least offensive kind of noise. A little expansion works wonders. Egregious hiss is usually the result of of either bad gain-staging, or having something plugged into the signal path that doesn't need to be there. For example if you leave your entire effects rack plugged into the aux loop even when you're not using it, or incorrect bussing on an external mixer, or something like that.

Minimize your signal chain for the shortest possible path from mic to preamp to converters, and use decent-quality cables (not monster).

Unwanted background noises

Having said all of the above, let's move on to the embarassing truths of home recording: Your neighbor's lawn mower, the family TV in the next room, the upstairs neighbors walking around on creaky floorboards, sirens and traffic. These are all sounds that are commonly heard in the homes of musicians the world over. They should not be captured on your recordings. Notice I did not say they should not be distincly AUDIBLE on FINISHED MIXES. I said THEY SHOULD NOT BE CAPTURED in the first place.

Unwanted background noises will usually end up masked in the finished mix, but that does not prevent them from muddying up the sound, limiting your options vis-a-vis processing, and generally making your record sound worse than it should. Moreover, and I think this is one of the dirty secrets of a lot of home recordists: anytime you can hear your neighbors, they can hear you. And unless you are profoundly confident and un-self-conscious, that awareness is likely to affect your performance, which is vastly more important than your audio quality. Your ability to get 40 takes of singing "let me lick you up and down" should not be affected by fear of the elderly landlord couple downstairs.

It is very important to have a quiet place to record. If you don't, move. I'm serious. Forget soundproofing. Legitimately soundproofing a typical residential room (one room) STARTS at $10,000. And it involves the kind of heavy construction that most landlords forbid and that reduces rather than improves property value. A windowless, double-doored room is not a legal bedroom in most developed countries. And taking a foot off of the floor-to-ceiling height by floating a room-within-a-room is not a selling point nor a subtle modification for most buyers. And that is where soundproofing STARTS. Do not waste money on foam or egg-crates. That way lies madness.

The only exception is if your problem is a single door or window that you can realistically block or replace. If you can buy an industrial solid door or block off a window with an extra mattress or something, and actually SOLVE THE PROBLEM, then go for it. But be realistic, and don't waste valuable recording time on piecemeal non-solutions.

Fortunately, working with samples, direct recording, and other such studio trickery offers a LOT of high-quality solutions for modern computer-based recordists. A multi-input soundcard, a midi keyboard, and an inexpensive electric drum kit triggering a good VST sampler offers everything you need to record a typical rock combo at headphone volume these days, and you can get great results that way. Take a weekday when nobody is home off to record vocals and you can solve a lot problems. You can even get wind controllers for the horn players.

This is not necessecarily the ideal approach, though. And it requires some degree of "scheduling" inspiration, which is an approach that I am pretty skeptical of. Moreover, this approach assumes that all the material has been thoroughly written and rehearsed in advance, which implies the existence of a rehearsal space. And if there is a rehearsal space, why not record there? (quick aside – the ambient noise in a lot of commercial practice spaces is actually worse than a typical apartment. Given the choice between recording below people watching TV and below a live jam-rock band, well...) Unless there is some "all-headphone" band that I don't know about. Which sounds pretty lame, but who knows?

I cannot solve all of these realities for any particular individual in any particular situation. But if you do not have a space in which you can realistically record the kind of music you create on a reasonably flexible schedule that coincides with your realistic free time, then you need to decide whther your music or your current residence is more important. Maybe you can rent a barn somewhere. It's a good time for real estate deals.

> Quote:
>
> Originally Posted by Heartfelt
>
> Yep,
>
> ...In regards to tracking, I am becoming aware of distance in my tracks. When a mix is assembled, the distance migrates into smearing and a lack of dynamic punch. My primary pre is a Daking which is known to be the opposite of that. What would you look to as a culprit or accomplices?...

I might need you to clarify what you mean mean by "distance". Are you asking about something specifically related to reverbed or far-field recordings? When you talk about tracking, I assume you're talking about something you can hear immediately when the track is captured – i.e. This is a problem that makes itself known before you go to mix. Is that right?

Without commenting on any specific mic pres at this stage, I think it's safe to say that the brand of preamp is probably not your main problem, assuming you are using it correctly. The first thing to start with is the source itself. For instance, if you're recording a cheaper, mushy-sounding piano with really old strings and subpar construction, then no mic or preamp is going to make it sound like a steinway, any more than a different preamp is going to make a tambourine sound like a splash cymbal.

This gets back to the very first posts in this thread, about level-matched critical listening.

You need to start with fairly assessing the real sound in the room and then work one step at a time. Doing this methodically will yield much bigger dividends much faster than randomly experimenting with different "recipes" or gear. In other words, if you're starting with an old, mushy-sounding piano (or a great piano in a mushy room), then you need to be fair and realistic in terms of what you can expect from the sound. This doesn't mean that there is no way to get a good sound from this piano, it just means that you can't squeeze blood from a turnip. If the piano itself plays the song in a way that sounds pleasing in the room, but that lacks plink, clarity, and dynamic punch that you ultimately want in the finished recording, then maybe it's time to think about, for example, doubling up the piano part with some midi samples. Or maybe you could add a low-level spanky guitar track behind the piano to make the track bounce a little more.

There are things you can do with gated reverb, compression with slow attack times, and noise gates/expanders which can exaggerate the sense of punch while still keeping a semblance of spaciousness, but they can't squeeze blood from a turnip. We can selectively flatter or exaggerate stuff that is already in the sound, but we can't necessarily make it sound different from its nature.

Listen very closely to some records that have the kinds of sounds you're after, and really isolate what the individual instruments sound like. I think a lot of people would be surprised at how “small” and undramatic a lot of their favorite instruments really sound in isolation. Sometimes, a huge, roaring rock guitar record actually has guitar sounds that are fairly small, low in the mix, and not very dynamic or dramatic. But when you add in really loud, punchy drums and a deep, powerful bass track and some shakers or whatever, the whole thing jumps to life. We hear the impact of the drums, the power of the bass, the motion and excitement of the shaker, and the guitar is just there in the upper mids adding some sustain and thickening it out.

But because the guitar saturates the range where our hearing is most sensitive, and because it is the most sustained element, the whole mix “fuses” in our mind's ear into one massive, punchy, powerful, exciting guitar track, alongside which our own guitar sounds seem wimpy or lifeless. The problem with this breakdown in critical listening is that it may lead us into trying to make guitar sounds that compete with whole-band recordings, which produces a worst-of-all-worlds result. The guitar is simply not going to “do it all” and trying to make it so produces something that muddys up the lows, masks the drums, and results in a weak, strangled midrange because everything is built up in the high and low corners.

I can't tell you what kind of sound you should be after, and I can't tell you what your expectations should be, but I can tell you that the most important element in the sound are basically as follows: **source** > **mic placement and type of mic** > **preamp** > **converters**. So if you start from the beginning, you can figure out for yourself where the problem is coming from. If the source sounds great but playback sounds bad at the same playback level, then try fiddling with your mics to get them to sound the way it actually sounds in the room. But be honest and make your AB comparisons at the same volume level. You can't expect the same clarity, punch, and size from a 60dB playback that you heard from a 90dB piano while sitting at the bench.

If you want to try and clarify what you meant about distance a little more, I might be able to offer better help.

Clear, punchy, and balanced sound

> Quote:
>
> Originally Posted by Heartfelt
>
> Yep,
>
> maybe instead of bogging down in my stuff, how about this: What contributes to an album sounding clear, well balanced and punchy? If this is putting the cart ahead of the horse, I am content to wait... please continue.
>
> Rob

First, no fear of carts before horses here – it's all just a big jumble of carts and horses and we're trying to fit them all into a pair of 5” speakers.

Moreover, I guarantee that your specific questions are more valuable to more people than my vague and unguided ramblings. If one person dares to post a question, that means that a thousand others were wondering the same thing. So no worries at all about “bogging down” or any of it. The stupider you think a question is, the more people are probably thinking the same thing. The worst part about most recording books is that they are all written either with the idea that the reader doesn't understand the documentation that came with their compressor, or that they already know what different compressors sound like.

You might know something that I don't, and I might know something that you don't, but if neither of us asks and we both defer to the other out of courtesy or humility, then neither of us learns anything. So the stupider the better, when it comes to questions. Frankly it's the stupid stuff that most often gets left out.

More specifically, "clear, punchy, and balanced" are all inter-related. It might be time to talk about arrangements, but I'm not ready to go there quite yet (there is SO MUCH to cover!).

The first thing is that all of these goals are easier and more obvious than you think. “Punchy” is the effect of sharp dynamics that are sustained **just** enough to momentarily raise the AVERAGE perceived volume level above the baseline volume level. Clap your hands. Do it. That's punchy. Want to add punch to a track? Record some hand claps, or cowbell, or wood block, or a xylophone (really –listen to old Benny Goodman records).

Don't fear the reaper, nor his cowbell.

Want to bring out the “punch” in a track without adding handclaps or cowbell? Turn up the backbeat (kick and snare) relative to the rest of the song.

Want to “punch up” a particular instrument? Create a bigger difference between the level of the first few milliseconds of the instrument attack versus the steady-state portion of the sound. A compressor with a low thresh, heavy ratio, slow attack (50 ms or more), and quick release will actually exaggerate rather than compress your dynamics.

"Punch" is the sound of instrument dynamics. A plucked string or a hammered drum sounds louder in the first instant than it does a few milliseconds later. That's all there is to it. There is no way to sidestep this. YOU MUST HAVE HEADROOM TO HAVE REAL PUNCH.

Modern digital look-ahead, frequency-variable limiters have a few tricks that emulate some advanced mastering techniques for limiting dynamics while preserving the impression of "punch", but they are so inferior, unnecessary, and extreme that trying to employ them without having a very sophisticated understanding of what you are doing is like asking how to do a power slide in a Hyundai Sonata so you can shorten your commute to work by power-sliding off the exit ramp of the highway. The short answer is that this is a great way to get in a massive wreck, and a very poor way to try and improve your everyday life.

"Clarity" is all about creating space, and it is closely related to "punch". It is a process of stripping away. If the low end is cluttered and muddy, try using a high-pass filter or a shelving filter to get rid of everything except the kick and bass. If it still sounds murky, start filtering those instruments. Especially in the low end, clarity and punch are all about definition. A thumping bass part plus a thumping kick drum equals LESS overall thump, not more.

You cannot create clarity in the upper midrange by hyping everything up there. You have to strip away. One of the golden rules of the great arrangers in days past was to never have any instrument playing in the same range as the lead vocal. When the vocal dropped out, that's when the clarinet, or the sax, or the guitar would play a little fill or riff.

Nowadays, the tendency is to have everything hammering on the upper midrange – wild organs, blasting horns, fizzy synths, clackety bass, clicky kick, explosive snare, and of course, roaring guitars (at least four tracks of them, no less). All fighting for the articulation range.

There are some ways of dealing with this. Frequency-limited/multiband sidechain ducking is one obvious starting point. But I am easing into that stuff deliberately, because it is not easy to do right until you understand the essential problems that you're trying to fix. And frankly because it is better to not have the problem than to try and fix it in the mix.

So let's begin at the very beginning. Let's say you have a straightforward jazz/blues combo onstage. Drummer starts with a backbeat. Kick, snare,kick,snare... (can you hear this? Bump, CRACK, bump, CRACK... maybe some hi-hat eighth notes or whatever...) No Problems with Clarity or Punch so far. (I'm going to abbreviate that last sentence as NPCP from here on – with me?)

So the string bass comes in (or P-bass, whatever), with a walking line that hits the backbeat accents. The bass player is in the groove, the bass notes are just giving tonality to the drum hits. The bass player, onstage with the drummer, is playing just loud enough to complement the drums. NPCP. With me?

Singer starts in, alto, let's say. She's singing, nice and mellow melodic lines over the punchy backbeat and the mellow bass sustain and tonality. NPCP. Any questions? Singer breaks for the pre-chorus. Guitar player comes in with a little melodic fill, echoing the vocal line, then switches to a spanky backbeat pattern that reinforces the snare drum as the singer delivers the chorus. With me so far? NPCP, right? Second verse. Singer. Guitar now continuing the backbeat pattern, just muted chord stabs over the snare. Tenor Sax comes in low and mellow, an octave below the singer, fattening up the melody and providing a tonal bed. NPCP, right? Second chorus. Singer delivers full-throated, lots of harmonics, sounding almost an octave higher as the tenor sax continues and as a Hammond organ jumps in, reinforcing the tenor sax part an octave lower with the left hand, and playing some fat upper-register echoes of the guitar part with the right hand. Band now sounds huge, but everything still has its own space. NPCP, right.

Third verse. Guitar now switches to a funky chunka-chunka part that hits the chords on the backbeat but also chugs the hit-hat. Singer picks up her tambourine and the whole band starts to shimmer and shake with the jingle-jingle-THWACK-jingle-jingle-jingle-THWACK-THWACK! Organ still jabbing the right-hand chords and echoing the sax on the lows, sax now playing fills between the vocal lines (there is a reason why they are called “fills”), bass and drums still pounding out the backbeat, singer still in full control of the alto range with full-throated harmonics competing with the organ jabs for the soprano range.

NPCP like a motherfucker, and this is just the first song of the set. Nothing to do but put up a mic and step out for a smoke. Even if you don't smoke. The band mixes itself.

Now let's contrast the above with a typical amateur garage band. For one thing, the drummer is never playing bump, CRACK, bump, CRACK – he's playing a drum solo the whole time, whether he's any good at it or not – cymbals crashing, toms rolling, kick and snare playing all around the beat but never on it, with no attention paid or the decay of the drums or how the drum sustain fits with the tempo...

Next, the bass player is not reinforcing the drum beat (there is none), the bass player is playing her own lead part, complete with loosey-goosey timing, an overloaded, clackety, stringy, midrangery sound that can barely keep up with the steady atonal crush of overloaded mud in the lows as she strives to prove that she's really just another guitar player...

The guitar player(s), meanwhile, are stomping all over the vocal range, thoroughly convinced that the only reason anyone listens to music is to hear guitar riffs and "solos", which are of course guitar parts played in the presence range whenever the guitar player feels like playing them, without regard to whether any other instrument including the singer have actually dropped out...

Meanwhile the singer is probably also cluelessly strumming chords on an overdriven electric guitar, with little sense of punch or clarity, just trying to be heard above the cacophony, often as not playing the wrong chords for the key of the song, but determined to strum them on EVERY VOCAL NOTE and somehow you are supposed to make that fit into the rhythm and tempo of the rest of the band (which has no rhythm or tempo to begin with). On top of that, concepts such as "range" and "melody" are lost on this singer who switches octaves constantly (badly) and who makes up for inability to create melodic tension by howling tunelessly (which you are somehow supposed to make sound "soulful" or "passionate")...

Meanwhile the keyboard player is in her own little world (and who can blame her), playing some kind of late-80's rearrangement of the whole song that is completely disconnected from the rest of the band (and also totally saturating the upper mids)...

Our poor soon-to-be fired horn player is left trying to play fills in no particular key (cue sad horns wah-WAHHHH)....

Okay, so let me take off my jaded audio guy glasses for a sec and stipulate that the second example might actually NOT be a bad band. They might actually have good songs, and an impassioned, energetic delivery and good musical and personal charisma. They might be the next Nirvana. But this is not going to be a "set up a mic and go out for a smoke" recording project.

The trick here is going to be to divide the sound not up as INSTRUMENTAL PARTS, which the first band did for us, but as SONIC ELEMENTS. In other words, It is totally possible that the best results might come from trying to isolate and clone some kind of kick/snare pattern from the non-stop drum solo, and reinforce that, either through some triggering and sample-replacement or clever mixing, just to get some rythmic punch back into the record.

It is also a certainty that the upper mids are going to be a carefully-threaded minefield of making sure that every instrument can be clearly and articulately heard. This is going to require a lot of careful back-and-forth listening and adjustment to find the least un-flattering aspects of each sound that can be made to fit in with the overall band.

How can we isolate some of the lows from the bass to reinforce the beat we sculpted out of the constant drum solo? How can we still fit in a little growl and string from the bass to keep the bass performance intact wihtout rocking the whole boat every time the bass plays a leading tone?

How can we best scoop the guitars during the vocal parts so that the riff doesn't drown the vocal, without making the guitars sound wimpy? How can we scoop out the mids of the singer's guitar so that the sound becomes jangly and atonal and so that the wrong chords don't jump out of the mix?

What should the relationship be between the keyboard melody and the vocal? How can the left hand of the keys be made to complement the bass and drums instead of fighting the guitar?

How can we make the singer sound like a badass instead of a strangled lamb on the "pasionate" parts?

If we look at the mix critically in these kinds of ways, the punch and clarity have a way of falling into place. The more you get back to fundamentals, the more the details take care of themselves.

Advanced mixing techniques are really arrangement techniques. Except instead of designing roles for certain instruments, you're coming in after the fact, hearing the instrument parts, and then deciding which kinds of roles to assign them.

In a sense, this is just another kind of organization – a place for everything and everything in its place. The real work is always in finding the "place for everything." Recipes work great with the first band, same as generic home organization tips work great for the couple with two kids, a spare bedroom, and standard-issue hobbies and home-office requirements. But what happens when the wife does marble sculpture, or the husband does hair styling in the home? What if one of the kids is learning bagpipes? The recipes break down when the assumptions change. A "music corner" in the dining room means something very different if we're talking about bagpipes instead of violin (if you ever lived with someone who had to practice bagpipes, you know what I mean. If not, count your lucky stars – they are loud as hell and there is no way to "stop" playing bagpipes, you just have to keep sounding notes until the air runs out).

The point is that both organization and multitrack recording become more difficult as the requirements shift from the conventional to the unusual. And any kind of "recipes" break down when you are cooking with new ingredients.

More to come. Questions and criticisms are good.

Cheers.

Mixing

> Quote:
>
> Originally Posted by Brad
>
> ...Will you be covering the nuts and bolts of the questions you asked in post #150? I would like to learn more in this area.....where you asked....”How can we best scoop the guitars during the vocal parts so that the riff doesn't drown the vocal, without making the guitars sound wimpy?”
>
> Along the same lines....fitting the vocal around a couple of fingerpicking guitars.....without killing off the nice fingerpicking....
>
> Thanks Again.

So far I have not talked too much about mixing. Not because mixing is not a hugely important part of the overall production, but because there is this rampant tendency on the web to say, “don't plan to fix it in the mix. Now, how can we fix this problem in the mix?”

There are a ton of mixing guides out there (nicholas' ReaMix is among the very best). I plan to talk about mixing later in this thread, but to skip over a lot of the lists of important eq frequencies, sample compressor settings, and so on. Partly because there are so many examples out there already, and partly because by the time you've gone through all the possibilities, you've negated the point of the presets and recipes in the first place. Any frequency is potentially a boost or a cut.

So with that said, let's talk about your specific questions: Why do you want two fingerpicked guitars if you can't clearly hear them both? Why is the guitar playing in ways that obscure the vocal? Is that what you want from the track? Is that want the guitar player is trying to achieve? If the musicians are not playing what they mean to play, if their sounds are not what they think they are or what they are supposed to be, then the problem is not a mixing problem (even if there are things we can do in the mix to address it). These are serious questions. There ARE a lot of ways to polish turds and “fix it in the mix”, but why start from that proposition?

Can the two fingerpickers alternate, or break up the figure so that one or the other is popping through the gaps in the vocal? Can you do that by simply muting or editing the parts? (first rule of mixing: Just because it's recorded doesn't mean it belongs in the mix) Can you take the rythm guitar and re-amp a cleaner, less obtrusive sound to use during the vocal? Better yet, can the guitar player back off and play a more muted figure instead of full-bore open chords during the vocal? (This would actually make the open guitar riff sound bigger and more dramatic when it does kick in.) You can use a compressor with the vocal plugged into the sidechain to duck the guitars when the singer is singing. You can get even more specific with a multiband. You can strip away all possible frequencies and gate the parts to make the conflicting fingerpicking as narrow and defined as possible, in the hopes of finding a little place for it to pop through.

You can get creative with panning to try and improve isolation and definition. You can use delays instead of reverb to try and minimize wash and smear. But why START from these propositions?

If you already know there is a conflict and what it is, why start by asking how to fix it after the fact? It's a little like saying, "I'll be crashing my car tomorrow, what is the easiest way to do bodywork myself?" If that's the way it must be, then so be it, but my first inclination is to look for ways to avoid the problem in the first place. I think there is internet-wide presumption that plugins and recipes and preamps are the secrets to great recordings, which leads people to overlook the obvious.

I don't know how much help this post is, but the more specifically we get into specifics, the more specific we have to get. IOW, there is no quick-and-easy "make a bunch of poorly-thought-out instruments in a bad arrangement fit together" preset. I wish I could just tell you to cut track one by 6Db at 2k and boost track 3 by the same at 1k and compress track 2 by a certain amount, but I can't. For the record, there are lots of other threads and articles that DO give those kinds of answers, if you prefer them. But I am not optimistic that the results will be as neatly satisfying as the instructions.

There is a lot of ground to cover yet. In the meantime, if you would like more specific advice, I and others might be able to help with more specific questions. Hope some of that helps.

> Quote:
>
> Originally Posted by shemp
>
> Same here. I'm having trouble in the low to mid range. Trouble getting bass, kick/snare and heavy guitars to sound decent together.

You need to decide which of those instruments is supposed to dominate the low midrange, and then the other instruments need to make room for it. (Here's a hint: one of those instruments might be called "bass").

I bet that if you turn down the "bass" knob on your guitar amp in acknowledgment of the fact that there is a whole instrument doing that job all by itself, you suddenly get a lot more clarity and power in that range, have the ability to crank the guitars higher in the mix for even more impressive power, and generally solve a lot of problems. It's like, "hello Mr. Guitar, now we have a bass, so why not take a load off? No need to try and do everything yourself anymore." (Alternatively, if the track is already recorded, you could drag a shelving filter up into the mids with a 3-12dB cut and see how much you can shelve off the lows before it starts to sound bad. But I like starting with a less bass-heavy guitar sound better) Goodbye mud, hello headroom.

I also bet that if you find a snare/mic/position combination that does not try to compete with the kick drum but instead just gives a nice midrange pop or crack, then you will create a lot more space for the kick to thump, and less need for the kick to try and compete in the midrange, since the listener will more clearly feel the distinct low-end.

Instead of trying to make every drum be all things to all people, focus on a kick/snare combination that is complementary, with good up-and-down motion (like, the way they call them "up" beats and "down" beats). Usually better than the common beginner approach of trying to make every drum sound like a bass drum, in my experience. With that last in mind, I bet the kick drum doesn't need much in the lower-mids at all. In fact, a tight "thump" down in the 40-120Hz range or so might be exactly what the track needs to complement and reinforce the newly-audible bass.

The thing is to think about every instrument, and to listen without preconceptions. Like, what is the role of this instrument? What does this instrument actually sound like, in real-time, in the real world, in the room where the band is playing? What are the dominant and most important aspects of its sound?

The danger is to just listen to every instrument as a solo'd thing and get caught up in trying to make each solo'd track as big and dramatic and complete as possible, and only after, try to find a way to fit the pieces together.

(I like chef analogies): If you are going to be serving more than one food item on a plate, then it is not necessary or even desirable for each item to be a complete, satisfying meal in itself. If you've got a steak and mashed potatoes and wilted spinach, then it is okay for the potatoes to be starchy, it's okay for the steak to be strongly flavored, it's okay for the spinach to be light – the meal is the the whole thing, how everything complements the other. Individual elements can and SHOULD be unbalanced or incomplete on their own, because they are SUPPOSED to go with and fit together with something else. Unless, of course, you are making a solo recording of a snare drum.

A couple of clarifying points related to the last few posts...

I'm not here to tell you what your guitar or snare should sound like, nor what kind of mix or arrangement you should aim for. My questions are genuine ones, not rhetorical. When I ask whether X is supposed to sound this way, the answer might be yes, or it might not be.

The point is not to tell you how to do it, but to think through what you're looking for.

By way of for instance, some heavy rock recordings in particular make use of very guitar-heavy soundscapes that are harder to work around. The old 80's metallica records for instance (pre-black album) had lots of layered tracks of very bass-heavy guitar sounds that soak up the entire frequency spectrum. The approach on these records was to have excruciatingly little bass, an almost inaudible little wub-wub, and quite “pointy”, papery-sounding drums. All of the meat of the track was guitar. The vocals were also heavily multitracked and also compressed and saturated, with most of the lows subtracted, and just kind of “soaked in” to the dominant guitar riffs. This was a very unconventional approach to mixing, but at the time and for what it was, it worked.

Other guitar-heavy rock albums, such as a lot of modern punk and nu-metal, use a very clackety, stringy, higher-frequency bass sound to “cut” through the wall of saturated, bass-heavy guitars. The “base” is really coming from the guitar chugs, and the four-string is almost kind of a special effect “third guitar.” Papery drums and trebly, delay-and multitrack-thickened vocals are again the norm, since there is almost no room for anything with any sustain to fit in the gigantic crush of guitars. These kinds of records are a nightmare to record and mix, but it IS possible.

Most sounds are, to some degree, either “fat” or “pointy.” The ever-popular “Punchy” is kind of a hybrid, like a “fat point”, if you will. And a lot of sounds are different things in different frequency ranges. A kick drum might be “pointy” in the upper-midrange click of the beater head, “punchy” in the low-end thump, and “fat” in the lower-mid “note.” And we might make a seperate category for clear, even, full-wash sounds in the midrange and up that we could call “transparent.” (Think Enya vocals).

It is very hard to fit two overlapping “fat” sounds together in the same frequency range. It is usually fairly easy to fit in more “pointy” sounds (wood block, spanky guitars, hi-hat or ride, etc). “transparent” sounds are also fairly easy to overlay on top of other sounds, but's hard to have more than one. “Punchy” sounds are prone to lose a lot of their punch if they overlap other “fat” or “punchy” elements in the same frequency range. It's all about changes in sound level, real dynamics. There is no magic secret to it – a sound that fills up and stays full sounds fat, a sound that fills right up and then drops right off sounds punchy.

This is why it is important to really listen to and think about how all these sounds fit together before we start setting up mics. Ideally, a real band who sorts out and rehearses their real material together, in a room, over time, will evolve orgnaically and will play with taste and sensitivity, adjusting their approach, attack, and note duration according to the instruments in real time.

Note that in reality, a lot the time, if anyone plays louder, it just makes everyone else play louder, too. Instead of giving each other space, the whole band is fighting for dominance. C'est la vie. This kind of approach is actually not all that bad to work with, and frankly any kind of performance dynamics is a breath of fresh air these days, even if it's just the whole band piling on top of the chorus. Any change in texture and intensity provides more drama and emotion than a click-synched 5 minutes of static volume.

Moreover, in the isolated, one-track-at-a-time world of home recording and loop-based productions that have never actually been performed, much less rehearsed in a real room, the above kind of organic back-and-forth is a pipe dream. But this just makes it all the more important to think through what role each element is actually playing.

If the guitar sound needs to pound on the low E and A strings, and extend way down into the bottom octaves, why is there is a bass player, seriously? (guitar is technically a bass instrument, and the bass only goes one octave lower). And if you've got a dropped-D or baritone-tuned guitar, then how many speakers are actually going to reproduce the two or three notes lower than that? Do they really matter? And if the guitar is furthermore a super-saturated modern high-gain sound that takes up the whole frequency spectrum, what room is there for other instruments, other than for papery drums to add a smidgen of attack to the overloaded guitar riff?

These are not rhetorical questions. These get back to some of the earliest posts about the kinds of soundscape we're trying to create. And maybe we ARE trying to create a super-aggressive soundtrack for space marine battles or whatever. But we're not going to get that AND get fat, pounding hip-hop drums that suck the whole air out of the room between beats, because leaving enough air to do that means turning down those massive guitars until they are whiny fizz behind the 808 stomp. In order for something to be big, something else has to be small. A mountain next to a tall mountain looks like a small mountain. 6'2” people in pictures next to NBA players look like midgets. Scale is relative.

So if we want to have a fat, punchy bass, then we need to leave room in the lows for the bass to breathe and punch. There has to be an empty space between the notes. If we also want to have a punchy kick drum then we have to find a place for the kick drum to punch that is't simply eating headroom from the bass. Good luck. So maybe we're better off just getting “fat” from the bass, and getting “punch” from the kick. Or vice-vesra (this can work great, actually). But neither of them are going to happen if the guitar is soaking up the whole low end, at least not without some very fancy trickery with multiband compression and look-ahead limiters that frankly is a fast track to unpleasant, fatiguing, unnatural, and generally bad recordings.

We'll get into to some of the mix techniques later, but the less your recordings depend on mixing magic, the better they will be (and the better the mix will be able to work its magic).

[begin page 5]

Marah Mag wrote in post #161

Hey Yep. Thanks again for this super thread. A few snippets from your last post, with some emphasis added :

Quote:

Originally Posted by yep

By way of for instance, some heavy rock recordings in particular make use of very guitar-heavy soundscapes that are harder to work around. The old 80's metallica records for instance (pre-black album) had lots of layered tracks of very bass-heavy guitar sounds that soak up the entire frequency spectrum. The approach on these records was to have excruciatingly little bass, an almost inaudible little wub-wub, and quite "pointy", papery-sounding drums. All of the meat of the track was guitar. The vocals were also heavily multitracked and also compressed and saturated, with most of the lows subtracted, and just kind of "soaked in" to the dominant guitar riffs.

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Originally Posted by yep

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Quote:

So maybe we're better off just getting "fat" from the bass, and getting "punch" from the kick. Or vice-vesra (this can work great, actually). But neither of them are going to happen if the guitar is soaking up the whole low end...

> Here's what I'm getting from this, and what I've found to be true while DAWing and also just from careful listening.
>
> Every instrument has its range where it "normally" belongs. But the actual range the instrument is capable of producing almost always exceeds its "normal" position in a mix and its function in a particular arrangement.
>
> What's important from the POV of a total mix is that there be enough frequency distribution to fill the ear in a satisfying way, but it doesn't necessarily matter WHAT instrument is producing any particular frequency range so long as the total mix is balanced relative to genre-expectations.
>
> That's why you can get away with papery drums that when soloed sound like nothing to be proud of and a pointy high-end bass that is barely "bass" at all, like your 80's Metallica example. (Aside: It's when you can successfully pull-off new balances that defy genre-expectations that new sub-genres are born... or at least novelty hits.) The idea is, when listening to – and actually enjoying – a well-made record, you don't immediately notice that the drums are tiny and thin, because they're still doing their job **as drums** in a mix that is overall satisfying your expectations of "heavy" or "full" or "punk" or whatever it is you wanna hear.
>
> In context of the mix, the fullness of the guitars will "lend" fullness to the papery drums and the pointy bass, just as the drums are lending rhythmic dynamics to what might be a just a wash of wide-spectrum guitar slosh. This is why in a typical mix you can lop off low end on the bass (even going up into its fundamentals) to let the kick through, or vice versa, because each of them "borrow" characteristics from the other. That's why it's called a "mix."
>
> Plus, the ear fills in what's missing, which is also what lets you high-pass into fundamentals; overtones always imply the pitch, and define instrument character.
>
> It's all an illusion. You don't really notice what's actually going on until you get "out from under" the full wash of the mix and look/listen closely at what's actually there. What's actually there is often quite surprising, and less than you would imagine or how you remember it.
>
> That getting "out from under" is one of the advantages of listening and tracking and mixing at sub-conversation levels, because it puts you more "on top" of the sound where you're less susceptible to the power of mere volume.
>
> Does that make sense?

Wow, Marah Mag. I think you just said in one post what it took me five pages to say. Exactly.

Sidebar: It's the performance that makes the song

Trivia question: what band recorded more number 1 hits than any other? More than the Beatles, Elvis, The Stones, and the Beach Boys combined?

A: The Funk Brothers, the then-anonymous house band/songwriting/arranging team behind Motown.

Home recordists take heart: all of the Detroit-era Motown records were made in the small (originally dirt floor) basement of Berry Gordy's humble Detroit home. I am paraphrasing from the film "Standing in the Shadows of Motown" when I say: "people always wanted to know where that ‚Motown Sound' came from. They thought it was the wood, the microphones, the floor, the food, but they never asked about the musicians." I am paraphrasing again when I say that it was widely thought that it didn't matter who the singer was, anything that came out of "Hitsville USA" (namely, that dirt-floor basement) was made of "hit." Smokey Robinson, Diana Ross, the Temptations, The Four Tops, the Jackson 5, Stevie Wonder, Mary Wells, and so on were basically just rotating front people for the greatest band in popular music history.

I don't care what kind of party you're throwing or what the crowd is like, if you put on "Bernadette" or "Uptight Everything's Alright" or "standing in the shadows of love" or "WAR" or any of those old Motown numbers, people will get out of their seats and start dancing and clapping (maybe on the wrong beats, but whatever). Nobody knows the lyrics, nobody can hum the guitar riff, and it has nothing to with the production. The music bypasses the higher cognition functions and directly communicates with the hips and the hairs on the back of your neck.

The guitars are indistinct, the keys are hard to make out, the horns and winds vanish into the background, James Jamerson's incomparable bass symphonies are the definition of "muddy", but the unified whole is impossible not to respond to. One cannot be human and not react to "Heard it through the grapevine", "Heatwave", "Tracks of my Tears", "Shotgun", and so on.

This is American-style popular music at its apex, and unlike nostalgic hippie music or punk purists, all you have to do is to throw it in the CD changer to hear its real power and musical accomplishment. No explanation or cultural context required. My point is not that everyone should aspire to sound like Motown. In fact I do not think it is possible or desirable to re-capture such a sound with any kind of production techniques. And my point is definitely not to argue that they were "good for their time" or anything like that. Throw it in the CD changer and see if it isn't just as good today. If you think it sounds "old" or doesn't hold up, ignore what I'm saying.

My point is that you could not MAKE a bad recording of this band. The recordings ARE bad – they are muddy, overloaded, indistinct, midrangey, all of it. And you could put those recordings into a cassette player and record the output of an old 6x9 car speaker through a cheap mic and then replay it at a wedding and it would STILL get more people dancing than anything on the top 40 from any era.

The production does not make the song. The preamps DEFINITELY don't make the song. Hell, the SONG doesn't even make the song, in modern popular music.

It's the performance. The rest is just flash and sizzle.

End sidebar. More to follow.

> Quote:
> Originally Posted by Heartfelt
> ...Pros care to add?

I should probably say that I am NOT a "pro". I was, once upon a time, a "pro" in the sense of somebody who earned his daily bread by twisting knobs on mixing consoles, but not anybody of note. Audio engineering is a cruel life, fraught with the acute anxieties of borderline homelessness in the company of grossly overpaid musos, and I could not hack it.

I am now just a hobbyist, who occasionally does recording projects, mostly for love, rarely for money, and never for more than break-even rates. I have made records that have been played on commercial radio, but such playings are few and far-between and I am not some million-dollar producer in disguise.

If my advice is helpful, then take it for what it's worth, if it's not, then ignore. In any case, do not mistake me for any kind of "authority" in the biz, and don't trust anything I or anyone else says unless it actually works to help you make better-sounding recordings. It is your ears that count.

The ringing phone effect: Hyped high-end

One more thing as you start to listen more closely to the production and the mix...

If you have one of those random/everything radio stations that plays all kinds of songs from all different eras, that can be a great resource for hearing a wide variety of juxtaposed approaches, and especially for hearing how skilled recordists in different genres may approach things.

Rolling off the lows is a common "oh, wow" moment when you first hit upon it, but do not overlook doing the same for the highs. High-end buildup is not always so obviously degrading and unsatisfying as low-end mud, but getting into the habit of rolling off the highs can also work wonders.

If I had to pick a single least favorite aspect of modern "loudness war" recordings, it would be the distinctive effect of having a big, flat wash of highs fed into a look-ahead limiter that modulates the extreme highs of the whole song in response to the actual dynamics that were once there. The effect is like having a constant ringing phone buried in the mix, and it only gets worse when the mix is fed through broadcast processing at the radio station. This is especially common in over-produced alternative rock bands, where you have strings, hyper-compressed splashy cymbals, multi-layered vocals with hyped highs, saturated, trebly guitars, and what-have-you all piled up in the highs. Listen for this "ringing phone" and you'll start to hear it everywhere, and it's not pleasant. This is the kind of thing that we mean when we talk about records that are "fatiguing" to listen to. They're loaded with essiness, seasick dynamics, and weird artifacts. And mp3 conversion and cheap DA converters only worsen these problems in real-world playback, especially when you have a huge stereo spread with lots of highs from different sources.

NOBODY who was actually using level-matched listening would actually PREFER such a sound. The reason people do it is to try and get the record "hotter." The engineer (or more likely, an A&R mook) hears the extra 3dB increase in signal level as sounding "better" for all the reasons we talked about earlier in this thread, so that's what stays. The problem with this is that you cannot use these techniques to reach through the listener's speaker and turn up the volume knob. In fact, these are exactly the kinds of recordings that customers are likely to turn DOWN, completely defeating the point of the degradation.

So once again, if it doesn't sound loud enough, use the volume knob on your speakers. And match levels every step of the way. Your ears will guide you, as long as you're not confusing them with hype and volume effects. The reason why so many people are inclined to record sources and then mix in ways that have over-hyped lows and highs is the whole "loudness switch" effect – it sounds louder, and louder sounds better. But it's a self-defeating cycle when you just keep piling on more loud and more hype and then turning down the mix to prevent clipping, and then adding more hype and more loud, and then turning down the mix to prevent clipping, and so on. And this is not just a mix thing, it's every step of the way, from setup to instrument selection to mic placement to gain-staging to tracking and so on.

It's not some super-magical thing requiring golden ears and magical gear, it's just careful listening and not deceiving yourself. And it's not actually that hard when you strip away the confusing superstitions and mumbo-jumbo and anxieties and TRUST WHAT YOU HEAR, without getting caught up in trying to guess at where the "hit magic" or whatever comes from. Just take ten deep breaths, and repeat to yourself "all you need is ears."

> Quote:
>
> Originally Posted by ringing phone
>
> I'm gonna go out on a limb here and say yep's ‚ringing phone' comment is a metaphor for ‚bad sound' ‚annoying sound'....not literally a ringing phone...

No, literally. That's the best way I can describe it – it sounds like a phone buried somewhere deep in the mix, as though there were a phone ringing far in the background when they recorded the tracks.

The effect comes from having really saturated highs that get rapidly modulated (pumped up and down in level) by aggressive digital look-ahead limiters and multiband compression. This is an ugly process in a lot of ways, but when it starts tracking really fast-moving signal such as the individual cycles of low-frequency content (yes, this happens), then it starts to modulate more delicate and sensitive parts of the sound.

Listen to some modern rock stations for a little while (like, ten minutes) and you are bound to hear examples of it. You might describe it differently, but I think "ringing phone" is a pretty good analogy, and egregious examples could certainly cause someone listening to loud music to reach for a phone with an old-style ringer or ringtone. If you take some high-passed white noise and sharply modulate it very quickly up and down in level, that's a pretty good way to synthesize a ringing phone, and that is exactly the effect going on here.

The technical causes for this are a little more complicated than we need to get into right now, but the cool thing about using your ears is that the technical causes really don't even matter all that much. If you level-match your monitoring decisions you would never apply the kind of processing that produces this effect, because it sounds bad. The only reason people do it is because it makes the signal hotter, which fools them into thinking it's an improvement.

> Quote:
>
> Originally Posted by Bubbagump
>
> ...Listen to any of the Hinder/Nickle Back like bands... they all have this cloud of high end in their sound. It sounds very big and 3D for about 5 seconds, then it is just tiring as you realize other definition is totally gone.

Even that big and 3d effect is an illusion created by loudness. The songs are mastered 6~12dB hotter, so the immediate effect when it comes on the CD changer or ipod shuffle is of a sound that "blooms." But if you actually level-match it against a pre-digital recording, the badness is immediate and obvious. It doesn't even have to be a particularly good alternate recording – some 70's disco or whatever.

And my point in this thread is not to rail against the modern "loudness race", it's just to point out how easy it is to fool oneself into making bad-sounding recordings, regardless of whether you ultimately decide to master them hot.

If anyone decides that they want or need to ultimately try and compete with modern hyper-limited records by squeezing the song at mastering, that's their business. But even still, you will get much better results if you are starting with good tracks and a good mix than if you go through the whole process trying to hype the hell out of everything every step of the way.

You can try to fool your listeners with "loudness race" mastering if you want, but for heaven's sake don't fool yourself during the recording process.

> Originally Posted by drybij
>
> yep – i apologize if this is off-topic, but i was wondering if you could comment on when it's appropriate to eq or compress a signal prior to recording on a DAW versus applying eq or compression after recording...

Great question, not sure if I have time to answer in full but here are some thoughts...

First I would refer you to all the stuff about gain staging above. The more analog you have, the more it matters.

Second, there are some situations where there is a technical advantage to certain kinds of eq and compression before the AD conversion. If you can remove rumble and clamp down on obvious and egregious spikes before converting to digital, then you will be able to have more bits of resolution for the stuff you actually want to keep. This is becoming an almost academic point with good 24-bit converters in modern multitrack recordings, but there is no reason not to use high-pass filters on stuff like female vocals, for instance. And if you're recording something like a shaker or metallic percussion or a clean electric guitar on the bridge pickup straight in, then chances are it's going to have a lot more dynamic swing than you really need or want, so there is little danger to knocking a few dB off the attack, especially if it's a wild player who is prone to clip the input.

Third, there is a lot to be said for analog. Analog compression in particular may be easier to get a smooth, natural sound out of than digital compressors. This depends a lot on the particular kinds of effects available to you.

Fourth, there is a lot to be said for working fast and committing to sounds while you are still inspired, as opposed to second-guessing and pushing off decisions until later. This depends a lot on how you like to work and how prone to OCD and ADD you are, but sometimes just doing the obvious thing as soon as it's obvious gives better overall results than obsessing over every little aspect of fidelity or theoretical "best practice." This consideration can cut either way – maybe it's faster and easier for you to just plug in the mics and hit record and then clean up the sounds later, or maybe you can focus better and keep up inspiration by getting the sounds closer to where you want them with a couple of quick eq rips before you hit the record button.

Personally, I have a really hard time feeling good about drum tracks in particular until they are at least approximately the sound I'm looking for – sometimes that means real-time monitoring with plugins, but if there's a decent channel strip on the input, why not put it to use?

Lastly, and with specific respect to typical bedroom studios, there is nothing at all wrong with just recording everything clean and then doing all of your processing "in the box", especially if the quality and usability of your plugins exceeds that of affordable analog gear. ESPECIALLY if you're not quite sure what you're doing with a compressor (I will get around to that topic, I promise).

If you have good, clean preamps and respectable 24-bit converters (see test from page 1 if you're not sure), then there is nothing wrong with just doing it all in the box. People can and do debate endlessly about whether analog sounds better and how important resolution is and so on, and some aspects of those debates have merit, but in practice there are a lot of very high-quality plugins that make it easy and cheap to get great sound. If you have the time and money you can buy the full complement of analog processors and experiment to find which are your favorites and how they compare with plugins, but IMO a good all-digital recording is not going to prevent you from getting signed or prevent your record from being a hit.

Resolution and conversion

Another couple of words on resolution and conversion, and why it matters.

Very low-bit converters do not sound as good as higher resolution converters. Modern 24-bit converters actually exceed the technical capabilities of the technology (they really only get about 19 or 20 bits of meaningful resolution, but whatever). The point is that reasonable recording levels, there is as much resolution as anyone could realistically hear, more than any real-world speakers could produce, and a little extra.

HOWEVER, any converter loses resolution as the signal gets quieter. If you record at like – 50dBFS, then you are basically recording 16 bits of resolution plus 8 bits of silence. (16 bits is actually perfectly adequate for real-world music, but it's useful to have the extra headroom and "insurance" of recording at higher bit depths). If you were to record at say – 100dB, then you would effectively have an 8bit recording with 16 bits of silence. (speaking in round numbers here). This is getting into territory where we are starting to hear noticeably degraded signal in the form of grainy tails and general “digititis”, particularly pronounced in the highs and in quiet passages. But of course you would have to deliberately go very far out of your way to make such recordings, and no sane person would ever set their record levels that low. (In practice it would actually be noisy as all hell and probably much worse than an actual recording through 8-bit converters, but whatever).

So without over-stating the case, it's generally desirable to keep the input levels to the AD converters reasonably close to 0dB on the digital peak meter, within the parameters of careful gain-staging above. And generally speaking, that's about all there is to it as far as the modern recordist is concerned. Easy as cake.

BUT, there IS a slight possibility of extreme scenarios where resolution is needlessly lost due to sloppy work practices. For example, and going back to some of the stuff talked about above, if you close-mic everything and get that “big” proximity effect on every track, and then go back in with a digital eq and pull down all your lows by 12dB (ala TedR, above), then in theory, your converters devoted a lot of their available headroom and resolution to capturing some heavy bass that you did not need, at the expense of the more delicate and sensitive highs. IF you ALSO then boost those highs by an aggressive 12dB or so, then you are turning up any grainyness or other undesireableness that you maybe could have avoided by either:

- Using less proximity effect through better mic placement, or;
- Rolling off the lows BEFORE converting to digital.

This is especially true if you also apply heavy digital compression – you're turning up more and more of the highs and quiet passages that are most susceptible to low-resolution degradation, because you dedicated so much of your available resolution to capturing big, powerful, headroom-devouring low-end that you didn't even need.

This is MOSTLY academic, and would only ever become a noticable problem in pretty extreme cases. But it never hurts to use best practices when it is easy to do so, and it's always better to work in ways that are sensible in the first place than to try and push the limits needlessly.

Continuing...

Analog magic

A lot of the stuff about analog “magic” is a hard-to-parse-out tangle of theory, personal preference, superstition, gear chauvinism, and genuine technical differences. And maybe even a little bit of “magic.”

Undoubtedly one of the reasons why many people prefer to track stuff like drums to tape before importing into ProTools or whatever is just because they have developed and found ways of working that revolve around the peculiarities of analog signal. For example: Engineer tracks drums to tape, doing his eq rips and basic compression and gating right on the console, hitting the tape in just the right way that he's used to doing to get the drums to fatten up and punch just so. When he comes back the next day to mix, the drums are already "seated" – they're warm, sculpted, well-placed, and "glued" together from the combination of tape compression and the little bit of harmonic fire and spaciousness that this process adds to the sounds (so far, this is just from bringing up the decay and room sound by compressing, plus harmonic distortion – no need to infer any "magic" at all yet). He then dumps it into Protools or whatever for editing and it still sounds good, so he decides to give digital a little more investigation.

Same engineer, on the next project, tracks drums straight to digital. Comes back the next day to mix, and finds that the drums (which have not been saturated, compressed, and distorted) sound cold, isolated, and disconnected compared to what he is used to. It takes him a lot longer to get the drums to sound the way he wants them to, and he finds it a slower, more cerebral, and less-satisfying process compared to the inspired familiarity of tape.

Being that this engineer spends his days actually making records instead of prowling the internet for flame wars and gear debates, he makes the simple decision that recording to tape sounds better, and says as much whenever he is asked. He also feels that at least compressing and eq'ing in analog is preferable to digital. For obvious reasons he does not bother to spend weeks looking for freeware tape emulators and AB'ing them with his real Otari deck or whatever, he just tracks to tape first.

This perfectly legitimate opinion based on real and non-imaginary experience leads to a widespread misunderstanding that digital is somehow flawed or incapable of capturing the tiny details or nuance or warmth of real instruments. Theories spring up left and right that this is due to quantization or superharmonics or nyquist filters or what-have you. Boutique manufacturers bring to market expensive modules and processors of every sort intended to restore that "analog warmth." Preposterously high sample rates are proposed to try and capture the ultrasonic harmonics that digital is missing. Analog fever grips millions of home recordists who believe that this must be the magic that is missing from their late-night sessions of boosting every frequency to clipping.

Well, magic there may be, and then again maybe not, and maybe superharmonics or quantization irreversibly affect sound and maybe they don't, but we don't actually need any of that to explain why this engineer prefers working with tape. Occam's razor says that tape provides him with an intuitive, famillIar, and easily-controllable form of processing that he's become used to. And the most obvious technical aspects of that processing are things that we can reproduce or at least approximate with other kinds of processing (including digital), so there is no reason to ipso facto conclude that there is anything supernatural about analog nor intrinsically inferior about digital.

And here is the kicker – when you record to digital you are already recording an **analog** signal. The mics, preamps, and input circuits ARE analog. So whatever "magic" supposedly exists in analog should theoretically exist in EVERY digtial front-end already! When he's taking his "analog" recording and then dumping it into protools after it's got that "analog magic", you're doing the same thing when you plug into the preamp on your firepod or whatever and then converting it to digital!

Now, it may very well be the case that some processors sound better than others, and it is entirely possible that some or all of the best-sounding ones are analog, but a lot of the analog crowd is trying to have it both ways when it comes to the theories they propose. If digital is bad because it chops the waveform into quantized slices, then why is it acceptable to record to analog and then chop it into slices in ProTools for editing, or for playback on CD? If analog is better because it retains ultrasonic harmonics, then why do low-passed vinyl records still sound good?

EVERY digital recording is analog first, then digital, then restored to analog on playback.

This applies to recordings that are recorded straight into an onboard soundcard, as well as recordings that were tracked and mixed entirely in analog and then passed through a single digital processor at mastering. If there has ever been a single good-sounding CD or DVD, then digital is capable of good sound (and there have been, I've heard them).

This doesn't mean that all freebie compressor plugins are just as good as a Fairchild, and it does not preclude a certain "magic" in the way that certain kinds of well-designed circuits react to varying signal voltage in ways that mimic human hearing and the mechanical reactance of sound in open air, but it does mean that digital is **capable**. And occam's razor suggests that the electrical processes that happen in analog circuits are subject to being analyzed and reproduced by clever makers of digital processors, at least theoretically, and that those processes do not require exotic theories of human hearing or spiritual resonance to explain.

I do not claim to have the answer to all questions and debates, just offering some food for thought next time your heart sinks when your favorite producer says he prefers the sound of tape.

[begin page 6]

16 bit vs 24 bit

> Quote:
>
> Originally Posted by stupeT
>
> Given my "real world poor man's studio"...Shall I print in 24 bit or is 16 bit enough and I will have no lose what so ever, but better performance of my DAW? Cheers
>
> stupeT

I think it is unlikely that an otherwise reasonably capable DAW computer would bottleneck due to recording at 24-bit instead of 16-bit. Reaper and all modern DAWs use high-precision audio engines over 24-bit, so your samples are being processed at high bit depths even if they are low-resolution samples. A second fast hard drive is pretty cheap in the scheme of things and almost a requirement for high-track-count audio, it seems to me.

Moreover, 24-bit is stupidly cheap and easy insurance against the single biggest headache of digital recording, namely trying to set the record levels high enough without clipping. With 16 bit, if you need to leave 24dB headroom above the average level for a singer with no mic technique, then you're really only recording at about 12 bits resolution on average.

The whole point of 24 bit is that you no longer have to record close to zero, you could record with peak levels of like -50 and still have CD-quality resolution. So you can leave plenty of headroom and just turn down the input gain as low as you want – no fear of clipping, and no worries of lost resolution, no matter how "wild" the singer.

Sample rate is a whole different thing, OTOH. Working at higher sample rates definitely affects performance.

> Quote:
>
> Originally Posted by BoxOfSnoo
>
> First of all, I love this thread... but a reminder to please keep this phrase in mind, or elevate it (in this context) to supreme importance! We want to know if it's possible to get fabulous results from our "real world poor man's studio"! Some of the tips at the beginning (uh, furniture?) are a bit "blue sky" for most home recordists...

If you can be more specific, I'll try and revise/advise. Even if you have to shop at junk shops or thrift stores I imagine you must put your computer on something?

(Now that i think about it I once had a four-track, a reverb box, and a little 8-channel mixer sitting on top of an old door suspended between two folding chairs in the basement of a house I rented with like 9 other people. That was a long time ago. The arrangement was suboptimal.)

> Quote:
>
> Originally Posted by stupeT
>
> Yep,
>
> not to be missunderstood: I benefitted SO MUCH from the way you explained things and gave tips so far. So its unfortunate for me to step in and slightly have to disagree in just that minor point:
>
> Loading 24 bit per sample instead of 16 bit does give just 50% more load to the part of the operating subsystem which is loading takes from hard drive. Either USB driver or PCI or whatever. ...
>
> stupeT

I stand corrected. I should say obviously it does affect hard disk performance, assuming that is even a meaningful issue (and I suppose it might well be for people who use a laptop with only a single 5400rpm or slower drive).

I'll amend the error, thanks for pointing out.

> Quote:
>
> Originally Posted by stupeT
>
> For me that one is answered by yep's explanations already with a plain: YES. The question is more: how? **Ggg**
>
> I state: a today's poor man DAW studio with some OK but not great mics and converters is way superior in everything – but studio acoustics – to what the top producers had in the 60s. And still they made great recordings the old days and most of us do not. So it must be us. Our skills, our experience, the way we do it.
>
> That's why I am keenly waiting for more input, pleeeeez...

Before this gets too far out of hand...

This is not and never was intended to be a "how to sound like a million-dollar studio for $100 and a computer" thread. I do not personally subscribe to the theory that an inexpensive computer-based studio is equal to an expensive analog studio. But my intent IS to describe some of the experience and knowledge that slips "between the cracks" of a lot of how-to guides, and to focus on basic techniques and approaches that work on ANY budget. And in keeping with that, a little PS to BoxOfSnoo's comment above about some of this being a little "blue sky"...

The reason I started with a lot of boring stuff about organization is because it is really important, and it is exactly the kind of stuff that many musos ignore for years and years. When I said that organization is more important than preamps I wasn't kidding.

I cannot tell you how many times I have been to some home studio or another where nothing is ready to record, nothing can be found, there are four name-brand guitars and not one of them has fresh strings or a good setup (and there are no complete sets of strings, just random-gauge loose ones), the only mic cable the guy can find crackles and hums when touched, the desk rattles and buzzes whenever anyone makes a sound, and one of the guitar amp tubes is blown. It takes the guy 45 minutes to turn on the computer, find "his pick" ("I think I left it in the kitchen..."), shut down all the junkware, stick a mic randomly in front of the amp with the blown speaker that sits under the buzzing desk next to the wheezing computer because that was an easy place to put it, and start playing some chords on a guitar with bad intonation, fret buzz and completely inappropriate gain settings. Then he realizes it's not tuned to standard pitch.

While he's tuning, he turns to me and says, "I've been thinking I should really just bite the bullet and get one of those Avalon preamps, because yours sounds really good and it seems like you can just set up and record with it." Or he asks if I can email him the settings I used to mix his songs when he recorded at my studio because they sounded "really professional."

And you know what? My Avalon DID sound better than his preamps. You know what else? A properly set-up el cheapo guitar with fresh strings in a quiet room with a well-placed amp and mic that were set up and ready to go would make a vastly bigger difference than a $2,500 class-A tube preamp. In fact, at his gain settings, you might not even be able to tell much difference at all between a $3,000 preamp and a $30 ART Tube MP.

What he is attributing to the preamp or to the effects settings was actually just basic good practice and an organized, sane approach to recording that was based on the SOUND instead of based on BRAND NAMES and "HOT TIPS." If you are that guy, then you need to sell one of the guitars and use the proceeds to buy a dozen sets of strings, some good-quality cables, a huge fistful of picks, new tubes for the amp, a thrift-store desk to replace the buzz machine, and a setup and re-fret on the other three guitars. If one guitar won't cover it, then sell two.

Even if your desk is a door on top of two folding chairs, put some cushions on the chairs if the door is rattling (I've been there). If you can't afford drawers and shelves, then save up coffee cans and shoeboxes to put stuff in. If you have an office chair that squeaks and rattles, then replace it with a $5 plastic lawn chair.

Instead of spending time on the internet reading gear reviews and plugins and hot tips, learn how to properly set up a guitar. Make test recordings in different parts of your house to figure out which rooms and corners sound better than others (this is probably the single best investment of time you can make). Keep your instruments set up and ready to record at all times. Pick up your cables and hang them on hooks so that they don't develop crackly humming partial shorts from stepping on them. And for the love of all that is holy, put some bass traps in your monitoring room. It's easy.

Apologies to BoxOfSnoo, it just occurred to me that there might be people out there who were thinking I wasn't serious with all that organizational stuff, or that it was for rich people or some kind of perihperal thing before we got into rolling off the lows.

Edit: In any case, if I have said anything in this thread that seems out of anyone's league expense-wise or skill-wise or anything else, please do raise your hand. Obviously some of the stuff on gain-staging or whatever will have less immediate applicability to someone recording straight into an onboard soundcard, but I'm trying to stick to principles that are relevant at any (and I mean ANY) budget and skill level.

> Quote:
>
> Originally Posted by Marah Mag
>
> Re: recording to analog
>
> Seems to me that tape compression and harmonic distortion were initially technical artifacts, that came to be appreciated as intentional effects, which eventually became part of an aesthetic...

Partly, and partly also that dedicated "boutique" analog designers have long since given up the idea of trying to design perfect equipment “on paper”, and have tended to focus on real-world trial-and-error tests of various components and designs to create circuits that are forgiving, intuitive, and “just so” in terms of response curves and slew rates and frequency-dependent variations in dynamics and so on.

The controls on something like a Fairchild or LA-2A are not what we would design a technically ideal compressor around. They are very specifically designed to “sound good”, much like a typical guitar amplifier is not made for fidelity but for tone.

A perfectly accurate recording of an electric guitar would be a reference mic in front of the strings, and it would not be a very satisfying sound for most guitar players. The shortcomings of the magnetic pickup system and primitive amplification technology of the early days of guitar have been harnessed, exploited, and carefully refined by obsessive tone addicts over the decades to produce an offshoot of audio that cannot be judged on normal scales of “quality.”

The best and most “analog” of analog gear has a similar quality, maybe like impressionist painting, if you'll forgive a crude analogy. It exploits and exaggerates the inadequacies and idiosyncrasies of the medium for deliberate effect, and at its best produces results that sound realer than real, and better than perfect. The current analog fetish is almost certainly overblown and over-romanticized in many respects, but that doesn't mean that there is not a kernel of truth in it.

That said, a lot of plugin makers have been creating digital processors that do a very good job of either trying to emulate the salient characteristics of the best analog gear, or of coming up with entirely new ways to create processors that are “musical” and creative in their approach to sound-sculpting, and that aim for something different from the rigid technical goals that gave early digital effects a reputation for being sterile, cold, and “too perfect.”

Good and bad are subjective judgments, and ears can be easy things to fool in strange ways. We can measure accuracy pretty well, but measuring “good” can be a bit trickier.

> Quote:
>
> Originally Posted by Colin_D
>
> ....I'm noticing something that sounds like a flanger from time to time. There's nothing but EQ on any given track and I can't ever hear it on any solo'd track so I think it's several tracks interacting in a goofy way. Is this an indication that everything's still fighting for the same space? How do I go about discovering which tracks are causing the problem? Colin

The cause is almost certainly nothing other than the most common. "phaser" and "flanger" effects are created by having two identical (or almost identical) signals playing simultaneously, where one of them is delayed ever so slightly. This creates the "whooshing" or phasey sound.

So... it is extremely likely that you have the same sound being slightly delayed somewhere. This could be from a routing issue, or from a duplicated track, or from some kind of signal that is somehow being re-routed back into the project, or it could very easily be from some situation where you have two mics picking up the same source, or from two midi tracks or duplicated midi notes feeding the same plugin instrument, or from a bounced version of the whole mix playing along with the individual tracks. It almost certainly has nothing to do with eq. When you happened to first notice it might have nothing to do with the cause.

I would encourage you to break off a new thread and post a copy of the project on stashbox or some such if you need more info.

> Quote:
>
> Originally Posted by routine
>
> I didn't really get that. i've read here and elsewhere that "hot" is not the best way to track and that we should check the meters to peak around -12.
>
> So i sillyly check the meters in my DAW assuming they are my converters meter but i'm beggining to think i was assuming wrong. So my question is how do you keep the input close to 0???

First of all, during tracking (and pretty much all the time, for that matter), the only purpose of digital meters is to tell you when you're clipping the signal.

So the first rule is don't clip. Which is very easy to do, just turn the input gain down so that the signal is not clipping, then turn it down some more in case you hit a loud note or some such. 10-12dB below full-scale is a pretty safe target for most kinds of material. Lower if your source is prone to big spikes.

The second rule has nothing to do with the meters. It is to figure out where your signal sounds best (using level-matched listening). All the gain-staging stuff above. Sometimes, with a very linear and quiet preamp, it doesn't make any difference. Sometimes it makes a big difference. If you have a crappy preamp or even some very good preamps, it is possible that the best-sounding gain setting might be well below or above the ideal “no clipping” target. Your meters cannot tell you what sounds good, they only tell you what is clipping. So stop trying to use them to decide what sounds good, and start using your ears. Make sense?

AFAIK, Asio sound cards should report accurate input level at the converters to your recording software, i.e. REAPER. So Reaper's meters should tell you accurately whether the signal is clipping. If you are using non-asio sound or an onboard soundcard, it might be possible that the soundcard itself has some sort of gain or volume control that happens in between the converters and the software. I'm not really sure about that – maybe someone smart can jump in?

But in any case it IS really important to make sure that you have a reliable clip indicator of some sort, since it is sometimes easy to miss clipping in the heat of battle and then discover a bunch of ruined tracks the next day.

Hope that helps.

PS – I am trying to cover this stuff in more less sequential order of most basic to most complex. So if something from an early post doesn't compute, please don't just skip over it. Ask questions. This stuff is going to get more complicated and will involve more synthesis of the early concepts as we progress, and runs the risk of turning into just another thread of meaningless, de-contextualized “tips n' tricks” if we are skipping over the basics.

So please, please ask questions if something doesn't add up or make sense. And feel free to criticize or disagree, too. I'm amazed that I've been able to rant this long without much real disagreement, but I am sure that will change once we get into signal processing and mixing and treatment of particular instruments.

Compression part 1

Okay, so I am going to do this completely backwards from how most guides would do it. I'm going to explain how compression works later. The first thing I want to do is to demonstrate what compression SOUNDS LIKE, because this is very often difficult for beginners to hear.

In practice, with strictly technical compression, the whole idea is that it's not SUPPOSED to sound like anything. Theoretically “perfect” mastering compression simply reduces the dynamic range in imperceptible ways. In other words, if you can HEAR it, then you're doing it wrong. This is very different from effects like reverb or eq, which may be subtle, but which are still audible as changes in the sound.

However, theoretically perfect mastering compression (aka "technical compression") is often a vastly different thing from the kind of compression that recording engineers get all wet in the pants about. Where compression really makes recordings come alive is in its ability to create a sense of power, fatness, size, and dynamic impact. Compression can change the whole vibe of a recording and make the performance dynamics come alive.

Compression Example Files

Attached to this post is a zip file of a reaper project consisting of two measures of a generic bassline. The exact same bass line is duplicated across two tracks, each with very different compressor settings and nothing else. Go ahead and download and open it. (pay no mind to the recording quality – this is just a bass plugged right into my internet laptop).

Now, forget about the compressor settings, and just alternate between the two tracks, toggling the FX button on and off (everything should be approximately the same output level, volume-wise).

Both of these tracks are set with fairly extreme but not completely improbable compression settings, and no other processing. Either, with some eq and gating could conceivably be close to a real-world application. My point with the examples is not offer "recipes" but to illustrate the ways in which compression alone can vastly alter the way a track "feels."

As you listen to the different tracks, pay attention to the following:

- Changes in the way the track breathes and pulses – not how it sounds, but how it "feels"
- Differences in how one version or another might fit in with either a very tight, snappy drum sound, or with a more "vintage" boomy, rickety, drum sound
- The fact that the post-compression versions are not less dynamic than the pre-compression version, they're just dynamic in different ways
- How the different compression settings alter the sense of timing in the track – how the bass pushes and pulls the beat differently
- How the frequency profile changes quite a bit, even without eq -How inconsistencies evolve and change organically, and musically, and affect the performance dynamics

- Each measure of the bass line is played slightly differently. On one, there is a slight "flam" as my fingernail hits the string right after the pad of my finger, and on the other, my fingernails don't touch the string. There are also differences in the way grace notes are voiced. The difference between the performance dynamic of the first measure and the second measure is pretty pronounced on the unprocessed track and could make for a track that would be hard to "seat" in a mix, because of the difference in attack from the fingernail vs non-fingernail versions. But BOTH flavors of compression even out the sound and lend a greater consistency.

Don't mess with or even thing about the settings yet, just AB the tracks against each other and with the compressor bypassed, and try and vibe on how the compression affects the whole feel and visceral impact of the track.

(apologies if the material is sub-par)

Attached Files

(this is folder 01. in the Yep Thread Extra folder)

In the above example I used ReaComp, partly because it's included in reaper, and partly because it is probably the most versatile compressor ever made.

But it also a very difficult one to start out with.

One of the tricky things about compression is that every single setting affects every other setting, and subtle adjustments to any setting can have completely different, even opposite effects depending on how the other settings are adjusted. You can see why this is harder than reverb or distortion, and why two-knob compressors like blockfish or the LA-2A are popular.

I will get in to the settings later and in more detail, but if you want to play around, start by really getting in tune with the vibe and the pulse of the music, and see how compression subtly but significantly affects it.

My example above is not meant to be anything like "ideal" compressor settings, it's just meant to illustrate how compression can almost make it sound like there's a completely different player on bass or whatever. It actually interacts with the music and can actually make the sound MORE dynamic.

More later.

Okay, so I just happened to plug my laptop into some real speakers and wow do I need to learn my own lessons!

The compression in the second track is terrible – the detection filter was set too high for the A note and there are these monster notes every so often... Goes to show why you need decent monitors! The laptop speakers wouldn't reproduce lows accurately, so I couldn't tell what was happening until I plugged the laptop into real speakers three days later. But the example still works for the purpose intended, to show how compression can alter the sonic quality of the music.

In any case, this also illustrates another lesson – don't go using these settings as presets! I will get back to this and talk through some of the settings.

PS quick addition to the great answer from FarBeyondMetal: (SEE BELOW!) palm-muted chugs usually require lower gain (less distortion) than you might think. Past a certain point, more distorted no longer sounds tougher, only fizzier. Also, how you hold the pick makes a difference. The guitar-teacher-hated “pencil” grip/wrist picking combination often sounds considerably chunkier than the more technically correct flat grip/elbow picking.

Keep in mind that almost 100% of all fast-picked metal riffs have the guitars doubled by kick drum, bass, and more tracks of guitars, so it is not necessarily realistic to expect a single track of guitar have the same effect.

> Quote:
>
> Originally Posted by DerMetzgermeister
>
> Great, great thread.
>
> I have one question, please. No need to answer it now, I don't want to derail anything.
>
> How can you get that palm-muted heavy distorted guitar right? I mean that sound that seems like the cabinet is about to explode and you almost feel the air shaking your unmentionables.
>
> Examples: The first chords of Meshuggah's “Soul Burn”, the first chords of Prong's “Snap your fingers”, the final palm-muted riffs of Metallica's “One”.
>
> What are the elements of that sound and how the engineers manage to register them in a recording? It is possible to achieve that with amp sims? I'm ready to be surprised with something totally counter-intuitive

I also recommend you this thread buddy.

http://www.ultimatemetal.com/forum/p...man-sound.html

It is very informative but there are a lot of useless posts to scroll through. I'll give you a hint, the cheese of that thread is the pics of mic placement, especially the “arrow” dual 57 set up with one 57 on axis and the other at 45 degrees off axis. People have experimented with all types of mics and placements in that thread and I found it to be a very good resource, so don't count it out just because there's nothing on the first page.

Edit: Have you checked out Slipperman's guide to distorted guitars ? It's extremely hilarious if nothing else, but I found lots of information in there helpful.

http://www.badmuckingfastard.com/sound/slipperman.html

Edit 2: I guess I'll go over some things that help me since it's 1:30 in the morning and my insomnia is in full force. First thing is that the amp should sound how you want it to sound in the mix before you ever even think about putting a microphone next to it. Next up you should experiment with mic placement. Even though I just referred you to a thread that is primarily about using two microphones, you might want to focus on using one at first, especially since you want that super aggressive palm mute sound. Using two mics is a hole can of wormies that I am just starting to scratch the surface on, but let's just say Andy Sneap, the guy who owns the forums to the threads I've been posting, uses one mic exclusively.

Anyway, to start out I think you should point the mic straight at where the dustcap and the cone meet, about an inch away from the grill of the cabinet to start. Move the mic around and use your ears, but keep in mind the farther away and the more off axis you go, the less in your face those palm mutes are going to be. I'm pretty sure that for the sound you want you are going to end up with the mic pointed straight at somewhere with the mic pretty damn close to, if not touching the grill. Also, I'm about to backtrack to the very beginning, but what kind of guitar are you recording with what type of pickups through what type of amp? All those things matter. String gauge matters, playing technique really matters, shit even the pick matters. If you wanna know my “secret” I use a 1.5mm gator grip, but that's just purely personal taste and is not even a popular thing among metal guitarists.

Try running an overdrive box before your amp (ibanez ts7's are only 40 bux) with the drive on 0 and the tone and volume adjusted to what sounds good to you. It will tighten up and compress your sound a bit, and also boost the mid range some. Here's a tube screamer guide

http://www.ultimatemetal.com/forum/p...rsion-1-a.html

You would be amazed at the metal tones that come from amp simulators now a days. I'm not the greatest at getting a good sound this way but I know that most people getting awesome tones are using Revalver and impulses, but you can get a good tone from any of the amp sim programs if you tweak enough and like yep says, use your ears. And here's a guide to using Impulses, only instead of using Sir like the thread says I would actually recommend voxengo boogex. http://www.ultimatemetal.com/forum/p...pulse-faq.html .

Sorry for all the aimless ranting and outside website posting, and sorry to hijack your question yep.

[begin page 7]

> Quote:
>
> Originally Posted by dero
>
> great thread, thanks to all involved.
>
> Could someone post the audio files as mp3 or .wav? I do my internetting on a very basic pc with no audio software that didn't come preloaded.
>
> Thanks

Just for the record, and for the benefit of any non-Reaper users who might be linking into this thread: Reaper is the most ridiculously easy-to-demo software ever made. Takes about 40 seconds from when you click the "download" link to when you are actually recording with the full-blown unprotected software, on a moderate broadband connection. And I mean that literally. It is nothing like installing Nuendo or Sonar or that kind of stuff, where you have to set aside 2 hours to install, validate, and configure. Any examples are going to get harder to make sense of without some kind of common platform.

Even if you hate Reaper and never plan to use it for anything and have other DAW software that you love and your internet computer is a crappy piece of junk like mine, I heartily encourage you to download the little REAPER exe for the examples. If you have the bandwidth to download wav files, you have more than enough bandwidth to download reaper and my ogg sample project. Reaper is the easiest way to have a common grammar and interactive examples that everyone can use.

> Quote:
>
> Originally Posted by BoxOfSnoo
>
> ...He used the MDA limiter, with limiting cranked way up "to see what's ducking the mix".
>
> I don't quite get this. Could you explain? Is it a viable technique?

I'm just guessing, but I think he meant he was using a limiter with aggressive settings to figuratively "see" what the dynamics or low end were like because he could not trust his ability to "hear" the dynamics or the low end.

There are a few clues that a limiter could give someone in such circumstances. For one thing, limiting artifacts in the higher frequencies (that the speakers CAN reproduce) can reveal what's triggering the limiter in the frequencies that you CAN'T hear. For instance if the cymbals and vocals abruptly suck down every time there's a kick drum hit, then you might have either too much kick drum, or a kick drum that is unbalanced or overly bass-heavy, e.g. if you can hear it clearly well-balanced in the mids but if the low end is obviously causing major ducking, then the lows might be disproportionate.

Similarly he may have been using the limiter's meters and filtering controls to see the "spaces in between" the audible music, to see how the measured signal level differs from what the signal sounds like. Looking at a "limit" indicator or gain reduction meter in conjunction with an ordinary signal level meter can tell you a lot about how the compressor or limiter filters and responds to the input signal. If you already KNOW how the limiter works, then looking at those meters could theoretically tell you something about the program material in terms of how it sounds, especially in terms of how much/what aspects of the sound make it "through" the limiter or compressor and cause more of a jump in output level than they "should."

We're getting way, way ahead of the ground I've covered so far in terms of metering and technical operation, but those are ways that a knowledgeable engineer might try and chase shadows of sounds that he knows he can't actually hear. Either of them could have actually revealed to me that there was a problem with the example file I posted, but I never bothered to check anything like that.

Is it a "viable technique" for getting around the problems of bad monitors? No, not unless you consider eating dead people and tree bark a "viable technique" for camping. People in desperate and demanding circumstances must do what they must do, and some of them make it through in inspirational ways. Are you trying to be an inspirational story, or to make good recordings? (hint: the latter has a much lower rate of tragic failure).

If you need to save money, sell an instrument. Don't eat out for three months. Make your own coffee. Cancel cable. Quit drinking or smoking. But splurge on monitors. Even if they are just the cheapest monitors actually sold as "monitors" they are probably better than anything in a department store, when it comes to monitoring.

How a compressor works (The Gremlin inside)

So how does a compressor actually work? I'm going to start out by talking about a conventional four-control compressor, which is pretty much the norm. The four standard controls are THRESHOLD, RATIO, ATTACK, and RELEASE, or occasionally variants thereof. Makeup gain, included on virtually all compressors, is just a simple gain (volume) control that comes after the compressor and that is completely independent of the action of the compressor. I will also refer to things like "circuits", pretending that we are still in the analog realm, but the principles apply to plugins as well.

There are also simpler two-knob compressors, and more complex ones such as reacomp that actually give you control over the detection circuit, and there are also idiosyncratic things like "time constants" and so on that some compressors offer, but let's set those aside for the moment. If you want a straightforward freeware compressor to play along with then Kjaerhus classic compressor is pretty good.

Inside the compressor is a little gremlin that turns down the volume. That's it. Really.

HOW and WHEN he turns down the volume is determined by the instructions you give him with the compressor controls.

THRESHOLD sets the gremlin's alarm clock. It is what tells him to wake up and start doing what he does, i.e. Turning down the volume. If you set the threshold at -10dB then the gremlin just sleeps his lazy ass off, doing nothing at all until the signal level goes above that threshold. A signal that peaked at anything lower than -10dB will never wake up the gremlin and he'll never do a damn thing. (see why presets could be problematic?) But once the signal goes above the threshold, the gremlin rips off the sheets and springs into explosive action.

RATIO decides HOW MUCH the gremlin turns down the volume, and it acts completely in relation to the threshold. If the ratio is set to 2:1, and the signal goes ABOVE the THRESHOLD, then the gremlin will cut that signal in half. For example, with -10 threshold, a signal that hits -5 (which is 5dB ABOVE -10) will be turned down 2.5dB for an output of – 7.5dB. Negative values can be confusing if you're not used to thinking in such terms so re-read and ask questions if you're stuck. This is important, and it does get instantly easier once you "get" it.

ATTACK is like a snooze button for the Gremlin's alarm clock. It lets the gremlin sleep in for a little while. So if the THRESHOLD is set for -10dB, and the ATTACK is set to, say, 50ms, then once the signal goes above -10dB, the gremlin will let the first 50ms pass right by while he rubs his eyes and makes coffee. An attack of zero means the gremlin will respond instantly, like a hard limiter, and will allow nothing above threshold to get through unprocessed. Any slower attack means the gremlin will allow the initial "punch" to "punch through" and will only later start to act on the body of the signal.

RELEASE is like a mandatory overtime clock for the gremlin. It tells him to keep working even after the signal has dropped below threshold. A release of zero means strict Union rules – once the signal drops below threshold, the whistle blows, and the gremlin drops whatever he's doing and goes back to sleep. But a slower release means the gremlin keeps compressing the signal even after it has dropped below the threshold. This can lead to smoother tails and less "pumping" or "sucking" artifacts that come from unnatural and rapid gain changes.

So, armed with that knowledge, you could, if you want, take a second look at the example project posted above. Or better yet, you could start to mess around with your own settings and material.

Here are some things to think about:

- A compressor with a SLOW attack and a FAST release could give a very punchy, lurchy sound, as the compression lets the initial attack through and then clamps down on the "body" of the note, bringing it down in level, and then lets go as soon as the note starts to decay. This would actually INCREASE the dynamics in the track, and would probably require a limiter on the output after makeup gain was applied.
- A compressor with VERY SLOW release times could overlap the release into the next note, compressing the initial attack even further, leading to a time-dragging feel.
- A compressor with a high threshold and a heavy ratio will flatten out the peaks of the notes, but will leave the body and decay unaffected.
- A compressor with a very low threshold will compress the entire sound, and will make the attack and body blend into the decay, ambience, and noise of the track.

If you "tune" the compressor by setting the threshold low and the ratio high so that it catches every note, you can adjust the attack and decay times so that gain reduction "bounces" along with each note in a way that complements the natural dynamics of the track. Then you can back off the threshold or ratio to get more natural sound.

If you instead "tune" the compressor by setting a slowish attack and realease time, and then tweaking the threshold and ratio to get the right kind of pumping and breathing, you can then adjust the attack and release so that the the impact and decay sound natural and well-balanced.

Practicing both approaches will quickly give you an ear for the subtle ways that compression affects the sound, and you will be able to achieve the best results by tweaking everything in tandem. But remember that certain settings can have opposite effects – with a longer release time, lowering the threshold could cause the release to overlap into the next note, killng your attacks. With a slower attack, increasing the ratio and lowering the threshold for heavier compression could actually produce MORE dynamic swing. And so on.

Every control is interactive, and every control depends on what is going on in the signal. Presets such as "rock bass" or "vocals" are basically completely meaningless. They might as well be labeled "random 1" and "random 2" when it comes to compression. The tempo and source material could make appropriate settings for one song have a completely opposite effect on another song with a different singer. So let's talk about some guidelines for where to set these settings...

THRESHOLD approaches:

- set the threshold just above the "average" signal level if you just want transparent-ish peak compression, like a limiter.
- set the threshold deeper, below the "average" signal level but well above the noise foor if you want to actually modulate the sound or performance dynamics.

(I cannot give numbers, because it depends totally on what your signal is doing. Look at the meters.)

RATIO approaches:

- Any ratio above say 10:1 is basically acting like a limiter – there will be VERY little dynamic variation above the threshold with these settings, EXCEPT as you allow via the "attack" window, or force via the "release." Ask if this is not making sense.
- Ratios of 2:1 or 3:1 will be very gentle compression, basically inaudible as processing effects, just giving a slight evening out of the signal levels.
- Ratios of around 4:1~8:1 will offer medium compression with some pumping
- As said above, ratio is totally dependent on the threshold Attack and release later.

> Quote:
>
> Originally Posted by shemp
>
> ok, two questions for me:
>
> 1. Does a limiter compress? Meaning, I sometimes use the Kjaerhus classic Limiter and I **think** I can hear some compression but there are no threshold and ratio settings on it.
>
> Please explain?
>
> 2. Please explain the 2 knob compressors. Is it more of a pre-set threshold/ratio/attack/release in one knob?
>
> Thanks!!!!!!!!!

1. "Limiter" is a bit of a fuzzy term. A pure, unadulterated brickwall instant limiter would be a clipper. I.e. It would simply clip the top off anything that exceeded the limit, like digital clipping. And this approach can actually be very transparent for short overs.

But most “limiters” on the market are actually very high-or infinite-ratio compressors with a fast or instantaneous attack and carefully-tuned release curves designed to have as little sonic impact as possible without actually squaring off the tops of the wave forms. How the designer approaches the release is what determines the sound and response of the limiter.

Digital look-ahead limiters actually slightly delay the output signal, which allows them to start compressing BEFORE the signal reaches threshold, which in turn allows them to modulate the very top of the waveform in ways that keep a microscopic smidgen of level variation, allowing extremely heavy limiting without the kind of obvious harmonic distortion that would come from a conventional instantaneous attack.

2. Yeah, exactly. For example, in optical compressors, the signal is passed through an LED or lightbulb that varies in brightness according to the signal strength. This in turn fires on a photovoltaic element of some sort (like a solar cell) that modulates the signal (i.e.

Reduces the gain) according the intensity of the light. Besause the light element does not respond instantly and has a certain delay before it achieves full brightness and another delay as it goes dark, there is a sort of built-in attack/release that varies according to the intensity of the light.

By selecting a “just so” combination of light source and photocell, a designer might achieve a continuously-variable response that becomes faster and slower according signal intensity and speed of change that sounds musical and natural at a variety of compression settings and on a variety of material. The designer might not need to add any additional attack and release delays. And a simple control to adjust the relative voltage sent to the light source could control whether it generally responded more quickly or more slowly.

Please note that there are also very fast-response, four-knob optical compressors, and slow-response two-knob VCA compressors. The optical type is just a little easier to visualize the operation of, I think, so that's the example I used.

You could also have 3-knob or 8-knob compressors, depending on how the designer decided to approach it. The famous LA-2A is basically a one-knob compressor plus gain (no wonder people like it!), as is the old Ross guitar compressor. More controls have been added over the years to make compressors more versatile for different kinds of signal and specific technical or creative goals.

> Quote:
>
> Originally Posted by stupeT
>
> Yep,
>
> can you talk about the feedback compressor design and what it does to the sound?
>
> Cheers
>
> stupeT

In most modern technical compressors, the design is feedforward through a sidechain. If you take the opto compressor example above, it would work this way: The signal comes into the compressor, and is split off into two seperate circuits. The main signal is fed right into the gain-modulated compressor circuit for processing, and a seperate “side chain” is fed to the LED or light bulb. This way, the plain unprocessed signal, complete with dynamics intact, is used to TRIGGER the compression that happens in the main compression circuit. That is feed-forward, and when you hear talk of side-chaining, it just means the ability to feed some other signal into the compressor's sidechain, so that for example you could use kick drum hits to trigger compression on a bassline to “lock” the two instruments together.

Feedback designs are actually much simpler. The signal only passes through the compressor once, and the level-detection circuit uses the output of the compressor. This is less precise, but some people like the slower, squishier sound for some kinds of applications. The sonic differences might not be very pronounced until you get into fairly heavy compression settings, but try it both ways if your compressor has a switch.

For technical compression such as targeted control of peaks, feedforward is usually better.

> **Aside:**
>
> The acoustics thread that I referenced at the very beginning of this thread has a lot less hits than this one does. I really meant what I said – studio acoustics is an absolute basic.
>
> Anybody who is following this thread who has not read through the acoustics thread is missing a gigantic part of this stuff.
>
> You can find it here, very top thread:
>
> http://forum.cockos.com/forumdisplay...aysprune=&f=29

yhertogh Post

Yep,

first...you have a way with words man...it's amazing how you can make things so clear with using plain words, rather than the typical behaviour you see where ‚audio experts' use very scientific terms to bury their lack of knowledge under. Thanks again! Awesome work! Quote:

Originally Posted by yep

If you "tune" the compressor by setting the threshold low and the ratio high so that it catches every note, you can adjust the attack and decay times so that gain reduction "bounces" along with each note in a way that complements the natural dynamics of the track. Then you can back off the threshold or ratio to get more natural sound.

This is a very similar phylosophy as the one in Michael Paul Stavrou's book 'Mixing with your mind'. Come to think of it, your and his views on things are very alike i find. He takes this a step further to explain that compressors are like a safe. You crack the dials one by one. He uses the ARRT acronym for this, first you crack attack, then Release, then Ratio, and then Threshold.

First you set ratio to the maximum value, put release to the lowest/fastest setting, and lower the threshold so that the entire signal is compressed. Ignore the horrible pumping you hear, but focus only on the ‚beginning' of the sound i.e. You focus on the attack button only. With this button you can create e.g. the thickness (or thinness) of a sound. If the material is e.g. a snaredrum you can almost ‚tune' the size of the stick the drummer is using. The attack affects the size of the hit.

Once you are happy with that, leave attack alone and adjust release. Release essentially controls the groove, the volume envelope over time. Try to set it as slow as possible while still hearing a nice groove.

Then leave attack and release alone and adjust the ratio (which was at its max). You can think of the ratio as a sort of lens. High ratios the sound will be firm, but small. Lower ratios the sound will be bigger but softer (also less controlled). Lower the ratio until you loose your above created groove, then increase it again to get the groove back.

Then adjust the threshold so that some sound still gets uncompressed so that the compressor comes to rest ‚in special moments' as stav puts it.

I hope this helps some people, and the above is not at all my invention. I just wanted to post this as i believe it is in the same vein as Yep's other comments in this thread AND it surely helped me to finally understand how a compressor works.

Yves

A little more on compressor controls...

I left off describing attack and release controls because I was trying to think of a good, easy way to get started with them, but yhertogh's synopsis does a pretty good job.

(yhertogh post is above)

These things have to be adjusted by ear, but having good meters helps give feedback to what you are doing. The recent REACOMP review at ProRec cited in the main Reaper forum actually gives a great overview of reacomp's controls for experienced users: (This review is in the folder 01. Yep Thread Extras_Started 1-23-09 included with this document)

However, I'm not sure I would recommend REACOMP as a first compressor for a beginner, because the controls are so powerful and so inter-related. The bottom half of the control panel in particular is really advanced stuff, allowing you to design your own detection circuit. And unless you either already understand compression AND frequency in a pretty detailed way, or are extremely patient, it could be hard to make sense of. But I do recommend reading the review. Even if it seems a little overwhelming, there is a huge amount of two-steps-forward-one-step-back to learning audio, and having some exposure to advanced concepts helps as your understanding grows into it.

Usually, compression (and almost all effects) should be adjusted with the whole mix playing, i.e. Not by soloing one instrument at a time. Very often, what sits well and punches through a mix well is very different from what sounds ideal as a solo instrument.

Stages to making a record

That said, there are at least two, and more often three or four distinct stages to making a record. When you're only tracking one instrument at a time, it is obviously impossible to evaluate the sounds you're capturing in context. And for that matter, even during mixdown, it's impossible to compare any single element in the context of the whole, finished mix, because the mix is not finished until you have adjusted all the different elements. I don't want to go too far into mixing approaches yet, because the stuff that we are talking about still has very real implications at the tracking and "pre-mix" stage, even if you track without effects.

For most engineers, there is a stage in-between straight tracking and full-blown mixing where you are doing some basic cleaning up and sound-sculpting just to get the tracks knocked into shape before you settle into the real task of mixing. I'm going to call this "pre-mixing." The specific boundaries between tracking, pre-mixing, and mixing can be a little blurry, but virtually every professional engineer does these as more or less separate stages.

Pre-mixing

Pre-mixing is all the processing that you do to a track before you actually sit down to mix them all together. In the analog days, the division was usually pretty straightforward – anything you did to the signal BEFORE you recorded it to tape was "pre-mixing", and the realities of tape saturation, hiss, limited access to finite numbers of outboard effects, and tape's natural frequency alterations kind of forced you to get clean, clear, punchy, airy, warm tracks of reasonable signal strength if you wanted to have good tracks to mix with.

Analog mixing consoles typically have eq and dynamics controls as well as effects returns for just this purpose (known collectively as "channel strips"). You would do obvious cleanup and intrinsic effects at the tracking stage, and set aside the real work of mixing for later.

In a commercial kitchen, this would be similar to the work done by "prep cooks" – picking out wilted lettuce, sifting flour, making stocks and broths, chopping vegetables, trimming meat, making sauces and marinades, cutting loins into steaks of the right thickness and so on. Nothing immediately edible comes out of it, it's just getting the ingredients into shape so that the line cook can focus on cooking.

In a studio, the idea is to get tracks that not only sound good but that will be easy to mix without getting bogged down in humdrum technicalities. And this process is even more critical to be aware of in the DAW age where it is all too easy to just record everything to an infinite number of tracks with an infinite number of available processors and then have a gigantic mess of ingredients to pick through and manage while you're trying to actually cook.

In the example project that I posted above, (folder 01. in the Yep Thread Extras folder) both versions were over-processed deliberately to illustrate the ways that compression can radically alter the "feel" of a track. You can use a compressor to chop a track into short staccato hits or to flatten it into a gently pulsating pad. You can make it pump and suck in an off-time, funky way or you can lock into an exaggerated syncopation. The compressor's detection circuit combined with how your gremlin handles attack and release can make for some pretty drastic changes, to the point where it sounds like there is a whole different player or instrument.

One of the biggest things that trips up beginners is finding that "sweet spot" of how far to go in the pre-mix versus what decisions to leave for mixing. There is a tendency to either leave every possible decision for mixdown, or to "mix" each instrument one at a time and end up with a collection of tracks that all sound big, hype, thumpy, punchy, and so on, and that are impossible to fit together.

Have you ever tried to make your own sauces or soups without a recipe? If so, you have probably had the experience of making something that tastes absolutely perfect when you dip your spoon into the pot and taste it, only to find that it is way too heavy and over-powering when you actually sit down to eat a whole plate of it. A half-teaspoon on the tip of your tongue is very different from a whole meal of mouthful after mouthful. This is the culinary equivalent level-matched listening. If you make a roux with some cooked fat, flour, sugar, and salt together it might taste fantastic on the tip of your tongue, but try and eat a whole bowl of it and you'll be vomiting in two spoonfuls.

Pre-mixing is the art of making tracks that are clean, consistent, noise-free, well-balanced, and appropriately dynamic, so that they are easy to work with come mixdown. I would encourage beginning mixers to get into the habit of saving pre-mixes as a separate, rendered project. For example, you track all your instruments, save the project as "minimum rage" or whatever, then go through each track and clean up and polish each track with mild eq, compression, gating, and any obvious effects such as intrinsic delays or guitar effects, and save. Then render each track with those effects embedded, and then save that as "minimum rage pre-mix."

Then use that project to do your actual mixing. If you have to go back, so be it. It might take a little trial-and-error, but it much easier and more intuitive to mix a project with cleaned-up, committed sound than it is to try and cook while sorting wilted lettuce and making stocks and so on.

Bringing this all back to compression, it is absolutely standard operating procedure to use more than one instance of compression on every track. And compression does NOT automatically mean killing dynamics – compression can actually make a track MORE dynamic.

Unless you're doing live broadcast work, there is no reason to use compression as an automated volume control to adjust the differences between loud and quiet passages. Fader automation is much easier and much more flexible these days. Use automation to even out the overall performance, and compression to affect the sound and the sense of intensity and performance vibe.

One of the reasons why I'm talking about compression early on, before getting into eq or reverb or even tracking specific instruments is that compression occurs naturally in all sorts of analog processes, and some of the best compression does not even use a compressor. If you listen to some older recordings of rock and roll, there is a great effect where the singer gets louder and more emotional, and the recording saturates and overloads, giving a terrific "effect" of loudness and emotional intensity, without much change in volume. The Temptations' "Ain't to Proud to Beg" is a great example, as are a lot of John Lennon's vocals. There is an explosive, analog "fire" on the intense syllables without actually varying the signal level all that much.

In recent years, there has been a kind of divergence, where cleaner, poppier, more "mainstream" records have avoided this kind of overload sound in favor of "cleaner" look-ahead compression and limiting, and where more "heavy" rock and metal records have tried to get that "overload" sound on every note of every instrument.

I'm not here to tell you what kind of sound you should go for, but there is a lot of potential to use the sonic illusions available to you to really make certain sounds "explode" out of the speakers with saturation and creative/intense compression effects. And having that kind of textural variation makes it possible get recordings that are fairly hot without becoming the constant white-noise earache of modern loudness-race stuff.

Stuff like old Rolling Stones or Velvet Underground has a very "analog" sound that sounds full-bodied and satisfying, even when quiet, and without degenerating into white-noisy fizz and "ringing phone" effects. By contrast, the latest Guns N Roses record sounds somehow too clean and un-ballsy in spite of being a very "hot" record. It somehow never seems to be at the right volume – no matter how you adjust the volume control, it either seems too loud or not loud enough, which is a sad departure from Appetite for Destruction, which is a record that sounds exactly the way its' supposed to (for good or for ill).

There is perhaps no better example of what compression is capable of than the snare on Simon and Garfunkel's studio recording of "The Boxer." That giant explosion that somehow sounds like a gunshot or a bullwhip without overpowering or even sounding artificial against the soft, delicate vocal harmonies is a perfect illustration of how careful dynamics control (plus reverb) can give massive creative power to the studio engineer, and maybe even make a megahit from a single effect.

Compression is a big part of what makes a record sound "right" at a variety of playback volumes. It's not about making things sound louder or quieter so much as making them sound proportionate and "right" in a dynamic sense. It is the closest that a mix engineer gets to actually playing an instrument, because it affects the sound in exactly the same ways that a really good singer or player does – it alters the texture and tone of the sound in real-time, dynamic ways.

[begin page 8]

> Quote:
>
> Originally Posted by FarBeyondMetal
>
> Yep, the bit you have been doing on compression and has been golden and has cleared up almost every confusion I have had with compression. I was just wondering if you explain how the knee effects the sound a little bit.

Great question. "Hard knee" means the compressor reacts instantly and faithfully to the parameters you select. Any "softer" knee means the compressor acts a little more slowly. If you have access to the sonitus effects package, you can actually see a graph of how the compression changes. If you don't, google image search turns up some pics of what various knees "look" like.

But how they look is not nearly as important as how they sound. And there is no substitute for experimentation. The harder the knee setting, the quicker the compressor responds, on both the attack and release curves.

The sound of any compressor or limiter is hugely dependent on a number of factors. The two most important that are likely to be controllable are:

- The detection circuit: does the compressor react instantly to any voltage or sample that goes over, or is the detection "weighted" to detect signals that "sound" louder, or conversely to detect signals that might cause overloads but that might pass the "sounds louder" test? There is no right or wrong, there are just different approaches.
- The response time ("knee") and whether it is related to the above: Some compressors react instantly, for a "hard limiting" sound. Some react more slowly, to try and minimize pumping/sucking artifacts by responding gradually. In some cases, a slower response can actually exaggerate compressor pumping. It depends on the kind of material and how the detection circuit is tuned.

There is no right or wrong, but in general, harder knees give more predictable results for technical compression. e.g. if you want to knock 6db off the peaks, then a hard knee and a neutral detection will allow you to just plug in the right settings. OTOH, a more focused detection circuit and a softer knee might not necessarily limit overs in predictable ways, but it might result in smoother, more natural instrument dynamics.

The difference might be pretty subtle until you get into fairly heavy compression settings. Compression can be hard to "hear" as an effect. A lot of compression is specifically designed to sound transparent. IOW, if you can "hear it" as an effect, you're doing it wrong. This obviously makes it challenging for beginners.

If you can start to“hear” unpleasant compression artifacts, that is exactly the time to start playing with the knee controls, or with different compressors, or with REACOMP's detection circuits.

Hope that helps.

If you can learn

> Quote:
>
> Originally Posted by ringing phone
>
> I don't really understand this..
>
> What everybody else said. It's not a rule, just a workflow suggestion, and Tedwood's approach of just doing it all at once is perfectly legit, especially if you are starting with good tracks.

In my experience, it is very common for the tracks to have certain things clearly "wrong" with them. For instance, the disappearing/reappearing bassline, the vocal that has objectionable essiness or lip-smacking or breathing sounds in places, or where there are wild fluctuations in level from poor mic technique, or the piano where the left hand is too heavy and muddy compared with the right hand melody, the guitar track that has hiss or hum, the hi-hat that has a lot of snare bleed, and so on.

If we start just trying to mix and eq these tracks all at once, it might be hard to get the right tonal balance for the bass while simultaneously trying to manage the disappearing notes, or when we turn up the treble on the vocals, we increase the essiness, breath, and lip-smacking. Or eq'ing the piano becomes challenging because it's hard to balance the lows on the heavy chords, or where reverb is turning all splashy or muddy because those offensive artifacts are still there...

This can lead to situations where we've got crazy-quilt eq with bizzarro cuts and boosts all over the place, and where it's getting really hard to adjust the compressor without over-emphasizing stuff that we don't want, and so on. Of course it's totally **possible** to make all these adjustments in back-and-forth stages, but it can be a lot to keep track of, especially if you're trying to keep up the right-brain inspiration while doing the left-brain creative balancing.

In a sense, this “pre-mixing” stage is making up for shortcomings in the actual tracking. If you started with perfect, perfectly clean,perfectly balanced and noise-, bleed-, and artifact-free tracks, then theoretically there would be nothing to fix. But in practice those expectations are not always possible. So the “pre-mix” stage is just getting the tracks as close as possible to how they would sound if they were theoretically perfect starting tracks.

This is the kind of processing that old-school analog types would do at the channel inserts, before printing to tape. You certainly don't have to do this as a separate step, and if you have infinite processing power and the patience and organizational skills to manage it, you could, in theory, just stack lots of plugins on every track and keep the flexibility by using one stage of input eq to clean up imperfections, a first stage of gentle compression to even out bad performance dynamics, an initial stage of gating to eliminate bleed and noise, and then start to stack on more effects for the actual creative mixing part. Or you might be able to just do the crazy-quilt eq and super-obsessive compression tweaking to treat everything in one pass. Whatever works.

Flexibility is often overrated. Flexibility is a good thing in the service of getting it right every step of the way, but it can also become a backdoor for the kind of counter-productive second-guessing and self-doubt and postponement of commitment that leads to projects where you have fourty takes of every track and stay up all night A/B'ing different speaker cabinet models in Amplitube, burning out your ears, killing your inspiration, and frankly overlooking the fact that the problem with the guitar track is not the speaker cabinet impulse but that the guitar was set to the wrong pickup and that the chords are too big or too small for the effect you're trying to achieve.

Finished is always better than perfect. Always. Perfect but not finished is actually neither.

Getting it right every step of the way before moving onto the next step forces you to make the right sorts of decisions, and to apply the right kind of critical evaluations. The ideal time to do the "pre-mix" is as you are tracking (but only if you're tracking someone else – don't start mixing up your own musical performance with technical stuff unless you're really comfortable doing so). These are difficult and blurry distinctions to draw, and I'm not trying to tell anybody what to do, just offering free advice, worth what you pay for it. You can have your money back if your recordings don't improve.

If you get your tracks perfectly set and finished before going to mix, it will make mixing a lot easier and more intuitive. Just as importantly, it will reveal any serious problems now, and allow you to focus clinically on specific technical challenges so that you can focus on the creative stuff during mixing. It will also reveal whether you need to re-track or punch in anything. Not that we hope to find that, but it's much better to find out now than later.

You know the old saw about "don't plan to fix it in the mix"...? Well, that means having everything "fixed" before you mix. If something sounds muddy or tubby or harsh or noisy or indistinct or uneven, it's only going to get worse when you start mixing. So fix it now.

Do the best you can with mic placement and gain-staging and instrument setup and so on, and there will be very little fixing to do, but if there are still imperfections in the track, then correct them now. And my advice is to simply render them that way. After all, if you could have tracked the “fixed” version, wouldn't you have done so? Well, now's your chance.

PS – this also the time to comp and edit tracks, and get everything settled and ready to mix. Leaving a bunch of non-destructive slip edits all over the place is a great way to create massive headaches down the road. It's just way too easy to accidentally drag an edit point or whatever, and it's way too easy to miss when you do it, so that twenty steps later you realize something is screwed up and you've lost the undo point and don't know how to put it right.

Get the edits right, and there will be no need to second-guess them later. The only reason to keep them is if you haven't actually decided, and if that's the case, you should decide now, before proceeding.

IOW flexibility is good when it is a tool for achieving results, but it is bad when it becomes an excuse for procrastinating. Excessive procrastinating is an indicator that you are either unsure of what to do, or that there is something more fundamentally problematic with the tracks. And neither of those situations is going to get better from adding more complexity to the project further down the road.

> Quote:
>
> Originally Posted by nfpotter
>
> Yep,
>
> I run into the “disappearing/reappearing bass line” fairly often (cheap bass, go figure).
>
> Sometimes I find it easy to solve, and other times not so much so.
>
> Do you have a “standard” technique for dealing with that specific issue?

This is a huge topic, encompassing almost the entire breadth of audio and psychoacoustics. But there are some basic ways to deal with it and I guarantee you're not alone, even among people with expensive basses.

It'll probably be next week before I can get into detail, but for starters, compression and eq (or multiband compression) are the easiest after-the-fact fixes. Listen and think about which notes are disappearing and see if you can zero in on them with eq.

A useful excersize for anyone who plays bass is to sit down and watch the meters while you play some simple lines, and try and get every note to hit the same average level. This is especially valuable for guitar players who may be unaccustomed to the huge dynamic swings with bass.

More later.

> Quote:
>
> Originally Posted by Moose
>
> ...Sometimes you have to pick a path, follow it, and see what's at the end. And realize that listeners won't be as close to the technicalities as the artists and engineers...

Yeah, absolutely. And it's amazing sometimes what you can accomplish just by showing up, going through the motions, and pretending to know what you're doing. Beginners fear that they might be exposed as ignorant, pros know that they are ignorant and proceed from there. And the latter approach usually yields much better results than the former. It's not a matter of "knowing the secrets" so much as a matter of coming to know that there are no secrets: the sound exposes all, and then working from there.

If any human being has ever created anything perfect, it has probably not happened more often than once every hundred years, and then by accident as much as anything else.

Nearly everything worthwhile is imperfect in some respect. If we never did anything that we could not be assured of doing perfectly in advance, we'd never do anything at all. And anything worth doing is almost always more trouble than it's worth. If I never did anything that wasn't more trouble than it's worth, then I'd never do anything at all.

But if we start from the proposition that we are going to expend more energy and time than a thing is worth, and that we are still going to come up short, then we can accomplish some pretty impressive things.

> Quote:
>
> Originally Posted by mamm7215
>
> This thread over at gearslutz is perfect for the bass question...
>
> http://www.gearslutz.com/board/maste...note-bass.html

Okay, gosh, wow. That is a great thread with some serious heavyweights. Bob Dennis and Bob Katz talking shop is like the pope and the president of the USA playing golf together.

That said, I'm going to encourage anyone who does not understand every single letter of what they are talking about to completely disregard that thread. It is mastering engineers talking about how to fix problematic mixes, and the examples they kick around could easily be misconstrued as "recipes", which I am certain is not how they meant them to be taken.

Moreover, if I ultimately have my way in this thread, your mixes will never require these kinds of mastering corrections – the mastering engineer will simply tuck and tail and set the timecode, the way it's meant to be.

On another forum, I once wrote a very long, detailed, multi-page process for home mastering, the long and short of which was that there was really no place for it, but in very detailed ways. It garnered some discussion and debate in other forums. I may at some point post a revised version in this thread or in another in this forum. Or maybe not. But for now, nobody who is learning anything from this thread should even be thinking about mastering. You can send your work out to be "mastered", if you like, or you can simply duplicate it.

I am still working on trying to figure out how to address disappearing bass notes in detail with a minimum of math and acoustical theory, and with a maximum of focused listening, in keeping with the spirit of this thread. I will post more once I figure out how to present it, but I guarantee that it will not amount to recipes of "cut at X and boost at Y frequency."

[begin page 9]

Disappearing bass lines revisited...

The hardest part about giving a clean answer to this problem is that there are so many things that can cause it. And it can't be answered in isolation. You really need to start from the very beginning, with room acousitics and decent monitoring. So if you skipped over the beginnings of this thread, go back and work through one step at a time or you're screwed.

I can't stress this enough as we get into more specific problems and approaches. If you don't have some bare minimum of accurate monitors and a solid grasp of level-matched listening then you're just groping in the dark, and you may as well try to cut your own hair without a mirror. Having said that, here are some of the reasons why bass is so susceptible to bizzare fluctuations in volume:

Human hearing is not linear. We hear different frequency profiles differently at different volumes. This was touched on earlier in this thread, but you can google "fletcher-munson" effects for more details. These effects are most especially prominent in low frequencies.

Basically, the louder something gets (in real-world volume, not signal strength), the more linear our hearing becomes, up to around 83dB SPL or so, and then it becomes less linear once again. Imagine an eq built into your ears that boosts the upper mids and cuts the lows of very quiet sounds but that does not affect louder sounds at all and you'll start to get the idea.

This is what "loudness" switches on older stereos do – they compensate for low-level playback by boosting the lows and sometimes the extreme highs. Modern mp3 players and car stereos often have roughly equivalent processing, and it was very similar to the ever-popular "smiley face" eq curve beloved of teenage car audio. The challenge here for audio engineers is that the overall balance of frequencies changes depending on playback level, which is why level-matched listening is so important.

The thing is, when a bass player is playing live, if she is a good musician, she is just playing the bass the way she wants it to sound, with the intended dynamic swings. And if she's playing fairly loud, as is common, then some notes might very well be deliberately a little louder or softer than others. That's what music is after all.

But when you turn the bass down to mix-friendly listening levels, then a note that is 6dB quieter than average is being pushed EVEN QUIETER by the fletcher-munson eq built into your ear. And notes that are 6dB louder are being pushed even louder. So a bassline that sounded great live, with some notes 6dB louder than average, and some 6dB quieter than average, might sound like it's swinging 12dB up and 12dB down when you play it back at lower listening levels. And this is a very big difference. This is why bass always wins the "most likely to be compressed" award in the audio yearbook. But the problem with relying solely on compression is that, in order to keep those quiet notes from disappearing, you really need to crank down the compressor into the meat of the average signal level, which can alter the sound and kill the dynamics that made the bassline cool in the first place.

The other problem with bass is the very nature of the instrument. The "BASS" part of the bass, the lower-midrange fullness, is more felt than heard. It creates the tonal foundation of the whole band, and has a huge effect on the overall "feel" of a mix, but it is almost "tone"-less in terms of the way we think of instrument sounds. This is not really a problem on its own, but it becomes one when we also want to capture the cool, slinky growl and snap of the strings, or the woody resonance of the instrument, or the burpy funk in the midrange. All of which are increasingly common ways to use the bass guitar in modern recordings, almost like a midrangey, percussive, "third guitar." The problem here is that those midrange and high-frequency elements occur in the more sensitive parts of our hearing, and they are usually SUPPOSED to sound exciting and dynamic and cool, like a guitar. This becomes a SERIOUS problem when we try to compress the BASS part of the sound to even out, as above, because (bear with me), the low-frequency fundamentals are much more powerful than the upper-midrange stringiness. THIS MEANS, that when the bass player plays a really loud note for emphasis, the compressor cranks it down to average level, which causes the semi-audible BASS portion to fit in better, but it causes the VERY audible "third guitar" to suddenly get much QUIETER – the exact opposite of the expressive intensity that the player was trying to achieve. And the compressor actually makes the parts that were supposed to be QUIET sound LOUDEST, because it leaves the upper-midrange performance gestures uncompressed on those notes.

And it becomes like trying to get a grip on liquid – the tighter we try to grab onto the lows, the more that we squeeze out the most clearly-audible highs. And if we let the highs convey the expressive performance the player intended, we have gigantic seasick swings in the low-end, at regular playback levels.

I hope this is making sense...

So one obvious solution is to simply go with an old-school, dull, flat-string, low-passed type of bass sound, and just let the bass be the bass and stop trying to make it sound slinky and snappy and articulate. But that won't win too many friends in 2009.

Another obvious approach is multiband compression. If we simply compress the lows and highs independently, we can create whatever dynamics profile we want for each. The downside is a tendency to end up with an unnatural, worst-of-both-worlds sound. If we flatten out the lows, they become disconnected from the articulation and expressiveness in the highs, and the highs start to sound clackety and fizzy and just "not quite right" without some reinforcement. This is, after all, the bass, and not simply a third guitar. Sometimes it works, but sometimes it just doesn't vibe right – the bass might be clearly audible, but it sounds like the "get up and dance" just got up and went, as though you replaced the bass player with a casio keyboard.

Another approach is to split the bass into two separate tracks, and process each independently and then mix them back together. A very common approach is to record the bass with a Y cable splitting the signal into a DI feed and also a miked bass amp. The engineer can then process the hell out of the DI to get a solid low-end, and use the amp sound to get the instrument "tone", and then mix them to taste. This achieves results similar to multiband compression without having to completely dissect the sound. (watch your phase relationships if you try it!) To be honest, I think 95% of the benefits of this approach can usually be achieved just by cloning a DI track and processing differently.

Yet another approach is to just say the hell with it and go ahead and compress the bass to death, unnatural dynamics be dammned. Especially if you first roll off the lowest frequencies, this can actually be surprisingly effective when combined with a big, powerful kick drum sound. A lot of disco and funk records have little or no bass in the lowest octaves, just a massive thumping kick drum, and then a very glorpy, burpy, pumping bass sound in the midrange. And the dynamics are

are weird and kind of inside-out-sounding, but it works. And the tracks often give the impression of being very bass-heavy.

"Hit bass"

Bona-fide professional studio bass players are among the most sought-after and highly-compensated musicians in the industry. Some play with a pick, some play slap-style, some play with a piece of foam under the strings, some play upright, some are virtuouso arrangement and sight-reading experts, some just play the root notes of the chords, some play extraordinarily slick and sophisticated accompaniments, but one thing that they all have in common is dynamics control, down pat.

They deliver "hit bass." And hit bass is unlike any other instrumental role, because it does not necessarily have anything to do with melodic quality or musical virtuosity in a conventional pop-music sense. It is perhaps most like the criteria used for hiring in the classical world, where tonality, intonation, and sensitivity to the conductor's time and vision are paramount.

"Hit bass" is a matter of being LOCKED IN. It means controlling note dynamics and duration so that the bass "locks" with the drums, and fuses the rest of the band together into a cohesive whole. Session bassists can make or break a song with microscopic performance gestures and nuance. They sound like professional "hit bass" as soon as they plug into the console input, and if you ever get to be in the room with one, it's an eye-opener just how polished, professional, and "finished" it sounds right from the first note.

If you're ever in that situation, and you're anything like me, then your first reaction might be to complement the instrument and ask about it, maybe ask if you can try it out. And then you might go to play the same bassline on it and realize instantly that this person has a skill set that is far different from the conventional definition of "chops." And you might completely change your practice regimen and attitude towards bass forever.

My point here is not to denigrate good bass players who are not session players or "hit bass" machines. Some of my very favorite bass players are not necessarily such. But there are some practical approaches that can get your bass playing a little closer to the solid, locked-in, "professional" sounding bass, and they are not necessarily stuff that is covered in normal practice regimens or lesson books.

"Hit bass" comes from the SOUND, not the notes. All three of the best session bass players I have ever spoken to have independently offered unsolicited variations on this statement: "Notes don't matter."

One said that outright, verbatim – “notes don't matter” (this was no less than Victor Wooten). Another, when I was trying to figure out what notes he was playing in a particularly cool fill, simply said, “Oh, it doesn't matter – I just play whatever my finger is on.” I was floored. I still never really figured out that fill, but I watched him play it through a few times and he was right – he was playing it differently every time, playing notes that didn't necessarily have anything to do with the key or anything, just flipping though this funky fill that SOUNDED THE SAME even though he was just hitting maybe 50% random open, muted, or half-closed strings. The third said, “it doesn't really matter what you play, as long as you eventually land on the right notes nobody's gonna notice the stuff in-between. Just play with the drums.”

And of course, the greatest bass player of all time* was notorious for just playing completely chromatic stuff whenever he felt like it while still somehow managing to sound perfectly on and appropriate, even simple, almost pentatonic.

Of course notes DO matter, especially for those of us without the intuitive mastery of the scales that allows some people to play without thinking about the key or the chords, but the point is telling. And all of these players are perfectly capable of and generally inclined to play along with the root notes of the chords.

But all of them are also thinking like a producer, or an arranger, or a sound designer, almost as much as they are thinking like a musician. Maybe more so, even. They have internalized the critical role that the bass plays in the way that a track feels, and how the low-end communicates differently from melody or chords or harmony. They engineer the track with their fingers, every bit as much as they play a melodic line. Consciously or not, they are creating production value, not just music. Their bass lines breathe and pulse and bring the “get up and dance” in spades, regardless of whether they are playing simple, sustained root notes in a ballad or blippetty blurpetty funky fills and clusters in a funk track or pounding eigth-note pedal tones in a four-on-the-floor rock or dance song.

None of this is to say that you have to have a session player to get good bass tracks. Many of the four-string greats did not necessarily subscribe to this “sound first, notes second” approach. But it is a vastly different approach to performance than most guitar players have, and it is helpful to think about and listen to bass in a unique context, and to adapt one's approach to the totality of the instrument.

Listen to some music that has been primarily recorded with session players and studio cats as opposed to named “band members” – disco, top 40, dance tracks, solo artists, country-western, and so on, and listen carefully to the bass, and to how the sound and dynamics are controlled. It often sounds much different from a lot of “band” bass players. And if you start to listen to bass more closely, you will

* James Jamerson, in case anyone doesn't already know.

start to hear which "band" bass players have "hit bass" and which ones don't. Neither is inherently better or worse, but it's worth thinking about and listening to this element that often gets overlooked.

Especially if you are a guitar player, I wager you will start to hear some basslines that really complement and flatter the guitar, and others that compete with it, and maybe over-step their bounds a little. The bass should not be fighting the guitar, it should be reinforcing it, strengthening it. Ironically it often guitar players on bass who are the worst offenders in this respect.

Bass fills should not usually sound like guitar solos. Bass fills usually do better as focused accompaniment or variations than as singing leads – the guitar is a better instrument for soloing. Bass players cannot get away with the same kind of loose, expressive timing that makes lead instruments sound soulful. When the bass does this, it makes the whole band lurch around like a drunk. Bass should be played with a careful touch, to keep the dynamics consistent and appropriate. Bass notes should start and end at specific points in time, and should not usually just be left ringing out and slurring over the next note.

An explorer is deep in the jungle, being led by a native guide. They are hacking their way through dense tropical growth when suddenly drums start pounding in the distance. The explorer freezes. His guide reassures him: "no worry. Drums good." "The drums are good? No danger?"

"Yes, drums good. Keep going."

The explorer takes a deep breath and they trudge on. As the jungle gets thicker and denser, and dusk starts to fall, the drums continue, pounding louder, ever closer. The explorer asks again, "Are you sure those drums are okay… nothing to be afraid of? It sounds like they're getting louder."

"No. No worry. Drums good."

They continue on.

As night falls and they start to break camp, the drums become even louder, more intense.

The explorer cannot shake a sense that they spell impending doom, but his guide continues to reassure him: "drums good."

Then, just as darkness settles most completely over the jungle, the drums suddenly stop.

The guide's face goes ashen, a look of horror in his eyes! The explorer asks, "What? What's the matter? The drums stopped – is that bad?"

The guide responds, "When drums stop, very bad! Bad thing coming! No good for anybody!"

"What!? What is it? What happens after the drums stop!?!"

The guide responds: “Bass solo.”

You know when a guitar or organ player or singer gets really into it and gets that “bad smell” look on their face and really starts wailing and unleashes a hurricane of musical awesomeness? Bass players shouldn't do that. It's like a big fat guy getting up and trying to do ballet with the dancers.

Bass is a very powerful instrument. The most powerful, literally. It uses more sound energy and physically displaces more air molecules and is louder than any other instrument. Bass has the ability to stomp all over the place and ruin things for everybody.

Playing bass requires a certain workmanlike disposition.

When you play bass, think Barry White, not Robert Plant. Cool and in control. Heavy-lidded, not wild-eyed. Sid Vicious made a great celebrity, but a horrible bass player. When you record and process bass, think clarity and punch. Have the bass player record the part at mix-level, with key processing such as basic compression and eq in the headphone or monitor mix. Ideally, have the bass player practice and rehearse this way.

Make sure the bass player has adequate low-end amplification. A lot of garage-band bass players have never really rehearsed with adequate amplification, and have grown accustomed to pounding the hell out of the strings and cranking up all the knobs on their amplifier. This approach makes for difficult studio recordings.

With specific respect to the problem of disappearing/reappearing bass, this condition is exacerbated by poor fingerpicking technique, where for physical reasons the player's fingers do not have the same “grip” on every string. They may tend to “push” the lower strings towards the body of the bass, and “pop” the top string as their finger “hooks” under it, since their wrist and hand sort of rotates around the strings while the thumb stays anchored. The “D” string often gets the weakest “pluck”, while the E and A strings get pounded and the G string gets popped, slap-style. This is hard to fix.

Bass guitar players should practice with amplification, and they should practice consistency. Playing “acoustic” electric bass breeds bad habits, because the lower strings are usually too low to hear, forcing the player to pound the strings. They're not playing bass, they're playing percussion that gradually morphs into a tonal instrument in the higher registers. This is fine when used as a deliberate effect, but creates serious problems when they want to crank up the bass and sound like thunder but have technique built around playing like a clackety percussion set.

> Quote:
>
> Originally Posted by Marah Mag
>
> Ironically, maybe, a good way to get a feel for this principle is by editing MIDI bass, where you can see the impact of tick-level changes in note onset and duration (and dynamics/velocity, too) in what are otherwise identical performances...

Frankly there is nothing at all wrong with keyboard bass, and I say that as a decades-long bass player who has at times made my living playing the four strings. Obviously a real bass is better if real bass is what you want, but midi can get great results fast, and there is no law that says that bass has to come from strings. It's just the lowest instrument.

And Standing in the Shadows of Motown is a book that has a permanent place right beside my favorite chair. The movie is killer, too.

PS to all of the above...

5-string or detuned bass is a nightmare to record and manage. If you like to use a 5-string live for subsonic effects or slap-style percussion, understand that the chances of it working in the studio are very slim.

Even the low E on a bass guitar is an extremely difficult note to hear, manage, and reproduce in an audio and acoustical sense. Anything lower is apt to come out of the speakers sounding an octave HIGHER, because the fundamental will not be reproduced, only the harmonics, and it wreaks havoc on headroom and signal levels. And nevermind the fact that these notes take something like 50 feet to develop in open air and are an acoustics and standing-wave nightmare.

If the low E on a bass (which is about 40 cycles/second) does not sound low enough, that is almost certainly because your speakers or amplifier are not producing it. Anything lower than that does not even sound like a note, it just feels like rumble, and real-world people are only ever likely to experience it in a THX movie theater, and even then it won't sound like a note.

The ranges of musical instruments have been refined over hundreds of years. Think carefully before going with a 5-string, and make sure that you are actually hearing the fundamentals. Most speaker systems, even higher-end home and car subwoofers, give out at around 50Hz. If the low E doesn't sound low enough, it's probably because you're not actually hearing it. Going lower is just going to pile up more subsonic mush that you can't hear.

PPSS –

With respect to the above, if the disappearing notes are all LOW notes, chances are very good that the problem is simply that your speakers are not reproducing them! A lot of good speakers and even legit studio monitors give out at around 55Hz or so, which is the fundamental of the A string on a bass guitar. And if you have standing wave problems in your monitoring space (and basically every residential space does), then God only knows what kinds of acoustical cancellations are happening. Which is why you really need to begin from the beginning, and get your monitoring and acoustics situation in order.

I'll post more later on dealing with notes that are too low for your speakers to reproduce, because it's not a purely theoretical problem.

> Quote:
>
> Originally Posted by bonefish View Post
>
> fabulous stuff, yep. Thanks for your insights. Would love to hear your thoughts on tracking a band live in the studio...
>
> Yeah, this thread is starting to get ahead of itself talking about effects and mixing approaches.

Phase shift and phase cancellation

Before we talk about multi-mic scenarios such as drum kits and full-band recordings, it's proabably a good idea to talk about phase a little bit. Phase is covered pretty well in standard discussions and books, so I don't want to spend too much time re-inventing the wheel, but it's a pretty important concept, so we should at least cover the basics.

“Phase” as it relates to audio actually refers to “phase shift”, which is the offset between identical or nearly-identical waves. Phase is neither bad nor good, it's a part of all real sound. But its effects become worth paying attention to anytime you have more than one signal path for a single sound.

Anytime the two versions of a sound are not perfectly “in phase”, (to use the colloquial audio expression), the sound will be affected. This is actually exactly how an equalizer works – it slightly delays a copy of the input signal and then combines it with the original.

This causes “phase cancellation” which alters the frequency profile of the sound. Here is a very crude illustration:

If you look at the curves as positive and negative sound pressure, then when both waves are producing positive pressure at the same time, then the intensity is increased. When one wave is producing positive pressure and another is producing equal negative pressure, they cancel out and there is no sound. When one wave is slightly offset, then cancellations and reinforcements vary cyclically and produce frequency-dependent artifacts.

"Phase" is everywhere, and can be caused by reflected soundwaves arriving at the same place at slightly different times, or by different parts of the source being further from your ear than other parts. For instance if you stand in front of a full-stack guitar amplifier, the sound from the top speakers is arriving at your ears before the sound from the bottom speakers. If you sit at a piano bench, then the vibrations from the close side of the soundboard arrive at your ear before the vibrations from the far end of the soundboard. In this sense, phase is no different from "sound." You just move the mic around until it sounds more good. Natural eq (in addition to reverberation and such).

But this is not usually what audio types are talking about when we talk about phase. Where phase becomes a specific issue unto itself is in any situation where there is more than one path for the audio to follow, e.g. if you have two mics both picking up a single source. If the mics are not the exact same distance from the source, then the soundwaves will not arrive at exactly the same time. This might be good or bad.

One very common phase culprit occurs if you record a DI bass track PLUS the miked amp cabinet and then mix the two together. The DI bass arrives at the audio converters almost instantly, but the miked sound has to travel a short distance through open air, delaying it about 1ms per foot. This can result in a situation where each track sounds good on its own, but when you combine them, the sound gets worse – i.e. Too thin, or too boomy, or just weird, or the telltale "whooshing" flanger sound of "phase shift." This is pretty easy to fix by using the JS phase adjust tool in reaper, or any number of other free plugins, or by simply zooming in and dragging the tracks back and forth in small increments until the waveforms line up. The old standby "phase invert" button that exists on practically every mixer and DAW channel simply flips the phase, and may be helpful, but it's a bit anachronistic these days when it's so easy to adjust the phase more precisely.

Other common and easy-to-overlook culprits for audio-induced phase problems include doubled midi notes sent to the same sampler or synth, cloned or bussed tracks that are routed through different processing that does not accurately compensate for processing delay (especially outboard gear), and anywhere else where two versions of the same sound might take different paths to get to the speakers.

This is all very easy to deal with in scenarios such as the DI/mic bass scenario, you just drag the phase until it sounds best. Note that "perfectly in phase" is not always necessarily the best, and it's not always obviously doable – if the miked bass amp has been eq'd or alters the tone somehow, then that means that certain aspects of the phase have already been altered. But whatever. Just make it sound good and you're golden.

Where phase gets a lot more technical and requires closer attention is in situations where you have not just multiple mics but multiple SOURCES. For example a drum kit.

If your snare drum mic is out-of-phase with the overheads, it's not such a big deal UNLESS your snare mic is also picking up a lot of something else, such as the kick drum. Now you can start to get into situations where the kick mic, snare mic, and overheads won't all "line up" together – you get the kick and snare mics perfect, and the overheads are whooshing the snare. You line up the overheads with the snare, and they start whooshing the kick.

Then you line up the snare with the overheads, and the snare and kick are whooshing each other, and you're back where you started.

There are some pretty obvious "mix fixes" here – you could just gate and eq everything to eliminate the offending instruments, but that's not necessarily ideal. Maybe you spent all day getting just the right balance of thump and beater attack on the kick and you don't want to cut all the highs and mids out of the kick mic. Maybe you you want the overheads to have that big, lush, "full kit" room sound. Maybe you worked really hard to find the perfect snare with a great decay and you don't want to just gate it and cut out all the lows.

Maybe you can re-constitute this stuff with reverb, maybe not.

So now you could go back and try to re-position all the mics the get the ideal balance of sound quality and phase integrity, or try using mics with tighter directional response, or whatever. Welcome to the maddening world of multi-mic, multi-source compromise. Where this gets particularly complicated is that the actual sound of the drum kit that you are capturing is not from single mics in isolation.

Even if you don't get obvious "whooshing" artifacts, you still come back to the original principle that phase is just a part of sound. You might eliminate the obvious faults, but still end up diluting and mushing up the wonderfully poppy and resonant snare sound or whatever.

This vague degradation is very similar to extreme eq, and is known as "phase smear." When you have lots of little delays of a sound, it is prone to lose clarity and body. You can simulate this by putting a lot of very sharp eq cuts and boosts on a track – it's not JUST affecting frequency, it's also sort of "smearing" the sound, like an out-of-focus picture.

Instead of hearing one "focused" capture, you're hearing multiple slightly delayed versions.

Avoiding phase problems

There are two basic ways to avoid phase problems in tracking. Number one is to make all mics exactly the same distance from the source. This is obviously impossible with a drum kit, because there's a big cluster of sources. Unless you pull far enough back to capture the whole kit with a single mic or pair (see far-field above), some kit pieces are going to be closer or further than others from each mic.

The other way is to make sure the distances between different mics are big enough so that the sound is significantly different or delayed. The old rule of thumb is 3:1. That is, whatever distance mic A is from the source, mic B should be at least three times that distance. So if the snare mic is 2 inches from the snare, then the OH and kick mics should be at least 6 inches from the snare. This is not very hard to achieve, but what about the toms and cymbals? And you better have every mic in a good shock mount, or you're going to get an instantaneous "DI" track of ever kit piece transmitted through the floor and up the mic stand to wrestle with as well.

A variation of the 3:1 rule can be achieved by simply delaying some of the track, for instance, putting a 10ms delay on the OH mics will effectively push them up 10 feet above the kit, evading the very short delays that cause the most objectionable "whooshing" effects. But we're getting into territory where we are no longer miking the drum kit for the best sound, but instead doing strange things to avoid outright problems.

A lot of times you just get lucky. Set up all the mics and it sounds pretty good. Other times you don't. Some people get super-obsessive about phase, pulling out tape measures and pieces of rope to measure the distance from every kit piece to every mic. Other people just wing it. There is no right or wrong, and there is no one-size-fits-all answer.

This stuff gets exponentially harder to manage if you are trying to record yourself playing drums. In an ideal world, there is a player playing, and an engineer sitting behind glass in a control room listening to the recorded sound directing an assistant through a talkback system who is moving the mics around at the instruction of the engineer, who can clearly hear the recorded sound. Anyone who thinks that all you need is a computer these days should try and record themselves on drums.

There is frankly a lot to be said for sample-replacement when it comes to home drum recording. Even a top-tier solution with multiple mics and complete flexibility such as BFD costs less than you would pay in shock mounts alone to do a full-blown multi-mic drum kit recording, and all the work of mic placement is done for you. Obviously it might not work for Art Blakey, but for a pop or rock backbeat, it's going to be hard to beat the sound quality in a home studio, even assuming you have a good drum room to record in.

Live band recording

In any case, before we get too far into philosophical arguments, this all leads perfectly into the even bigger multi-mic issues of live band recording.

The main argument in favor of live recording is the ability to capture the authentic energy of the real performance. The main argument against it is the massive increase in technical headaches and/or severely limited flexibility compared with one-at-a-time multitracking.

Which considerations are most important is partly a philosophical one, and partly a practical one. The more that the band's live energy and ebb-and-flow are integral to their sound, the more inclined we would be to sacrifice flexibility and technical control to capture that. For example a straight-up jam band or an acoustic jazz combo or Irish Sessiun would almost certainly be worth recording live, in the room.

On the other hand, a young, un-polished garage band with raw material that has not been well-rehearsed or arranged is almost certain to benefit from the increased control and production value that multi-tracking can afford.

A simple hybrid approach can sometimes yield the best of both worlds. Instead of starting with a click track, you could start with a rehearsal "scratch" track of the band playing the song live, and then have the musicians layer their parts on top of the live "scratch." This allows a natural, organic ebb-and-flow to the tempo and dynamics, and gives the musicians something less mechanical to perform to, but it still allows for the technical control of multitracking. However, it does not quite match the full "vibe" of eye contact and a good band who actually interacts in real-time, in response to one another.

I would caution home recordists to be careful of getting too abstract or philosophical with this stuff. There is a tendency to over-rate the importance of almost everything. The easiest litmus test of whether a band should be recorded live or with a multitrack/hybrid approach is to record a rehearsal with an accurate omindirectional mic (the Behringer ECM8000 is a great deal for a reference-quality mic, very handy if all your mics are directional). Listen to the playback and ask yourself honestly whether the biggest shortcomings are related to clarity and overall quality, or the performance.

If there are mistakes and off-pitch notes and inconsistencies of dynamics and instrument balance, then the band would probably benefit from the increased control allowed by one-at-a-time multitracking. If the recording sounds like a poor copy of a great recording and a perfect performance, then this might be a band that has "it" and should be recorded as-is.

it has become increasingly popular in commercial recordings of rock bands to stage elaborate setups that allow for live recording with eye contact and also complete isolation.

Glass walls, big constructions of gobos, iso rooms full of amps fed through to angled, phase-inverted monitor pairs, anything to avoid bleed without using headphones or comprimising "vibe."

The idea is to get the live "vibe" while still keeping the pure isolation and complete control of multitrack. This is a very lavish and expensive way to record, and an approach that you should forget about in a home studio setting. Whether it is a good or bad approach is almost irrelevant until you have a big-budget major label deal, because trying to reproduce it at home is basically impossible unless you have an awful lot of time and money.

The practical reality is that live recording means bleed, and lots of it. There is nothing at all wrong with bleed. You still have to set up the mics so that they sound good, and good sound is good sound, with bleed or without. The challenge is that bleed severely restricts your ability to do punch-ins and overdubs, and it also greatly restricts your ability to sculpt the sound in detail.

Mic setup also becomes more complicated, both for the phase issues noted above, and also because your choice of mic and placement is affected by what you're getting from other instruments, not just the one you're trying to focus on.

A great jazz combo or other dedicated live band basically mixes itself – the musicians change their own dynamic and tonal balances in real-time, with performance gestures. This makes live recording very easy. But a lot of bands that consider themselves to be "high-energy" live bands do NOT, in fact, mix themselves this way.

The biggest issue is vocals. I plan to get into specific approaches to recording vocals later, but for now the most salient point is that often the circumstances under which the vocalist **thinks** she sounds best (e.g. while playing guitar with a live band) are actually just the circumstances under which her mistakes and miscues are most heavily masked and compensated-for.

And this goes for the rest of the musicians, too. It is very easy to think that you're mistakes won't matter or won't be noticed when there are other interesting things happening, and to only focus on the stuff you did well. it's easy to hear the parts you nailed as proof of how good you can be, and to hear the parts you flubbed as "not that important" or "you get the idea" or whatever. Solo tracking removes these blinders, and sometimes puts the musicians in an uncomfortable position. But the musicians who are most inclined to hide mistakes behind the rest of the band are often the ones who benefit most from the scrutiny and studio trickery of solo multitracking.

More on specific techniques and approaches later.

> Marah Mag
>
> Quote:
>
> i would caution home recordists to be careful of getting too abstract or philosophical with this stuff. There is a tendency to over-rate the importance of almost everything.
>
> Hi Yep.
>
> What's especially valuable about these posts of yours is that while they're obviously rooted in experience and a solid grasp of theory, they don't pay homage to tired truisms that get endlessly repeated and that can lead to paralysis and a fixation on engineerial correctness.
>
> Stuff like ‚live' or ‚real' is always better, or nothing beats a band vibing live together.
>
> Procedure matters, but not more than the end result, not procedure for procedure's sake.
>
> Quote:
>
> But whatever. Just make it sound good and you're golden.
>
> Right!
>
> Quote:
>
> If there are mistakes and off-pitch notes and inconsistencies of dynamics and instrument balance, then the band would probably benefit from the increased control allowed by one-at-a-time multitracking. If the recording sounds like a poor copy of a great recording and a perfect performance, then this might be a band that has "it" and should be recorded as-is.
>
> I think this can even be simulated when it's just a solo songwriter and her lonesome computer. The trick, and it's not an easy one, is to work fast, get the basic parts layered.
>
> Maybe a few overdubs, or fast cut and paste pseudo-overdubs. The point is to get something like a basic ‚band' recording, ruff and enthusiastic and exploratory, without worrying about proving your engineering or your playing abilities or what the editors of SOS or anonymous forum members will think of your sounds and your mix... because there are no sounds and there is no mix at that point, there's barely even real performances. Render whatever the fader settings are. Let it sit for some hours or days until you're not as intimately familiar with it as when it's an open session. Then, put on your producer's shoes and see what you think of this demo you've received. What's good or bad about it should be more or less obvious. Don't fall too deeply in love with it.

That's a great assessment, Marah.

I think it's helpful to cover the basic principles, and then to get into ways to break the rules, but you touched on exactly the stuff that solo home producers often miss out on.

Without the ability to jam and rehearse and interactively develop arrangements and so on, there is a lot to be said for just working fast and planning on lots of revisions, as opposed to the morass of trying to create one perfect measure at a time.

A lot of electronic musicians and hip-hop producers have developed ways of working that build on a foundation of loops that have a pre-existing vibe or energy, and then embellishing it and "jamming" with the virtual band. A solo artist could take a similar approach by simply singing the melody along with a basic backing track, like the old-style song demos, and then building up a production around that, using the original scratch demo as a sort of glorified click track.

[begin page 10]

When recording live, there are an almost infinite number of approaches that can work. With an unlimited budget and the right gear in a commercial studio with multiple iso rooms, it is not uncommon to spend weeks just setting up. This is obviously impractical in a home studio/active band setting.

The practical variances are so huge that it is almost impossible to talk about "best practice." If you have a two-day session where stuff has to be broken down afterwards, then obviously setup has to be fast – no spending 10 hours finding the perfect balance of bleed, phase, and sound quality. If the whole band has to fit into an 8x12 room, then there is no way that the bass is not going to end up in every mic. If you're recording in a concrete basement with 7 foot ceilings then acoustics are going to trump every other consideration, and close-miking is practically manditory. If your recording space has to be kept open and practical for other uses, then talking about "ideals" is pointless. If you have only 8 mics and four stands, then what's the point of talking about trying vocal condensers as overheads and matched ribbons as distance mics? Having said all that, there are some basic principles that are worth talking about.

Start with the minimum number of mics and the simplest setup possible, and then add mics that you NEED, instead of starting from the perspective that you have to mic everything. And if circumstances are limiting, and recording live is important, then the fastest and easiest shortcut to good recordings is to work in mono. I'm not kidding. Mono is vastly under-rated, and has produced some of the best-sounding, most immersive and beautiful recordings ever made. And you can always pan stuff later. Unless wide-spread tom rolls and stereo cymbals are really critical to your sound (and I guarantee they're not, because they don't happen live), there is nothing wrong with just recording a drum kit mono.

The more critical it is to capture your "live" sound, the less critical it is to capture a "studio" sound. If your band sounds just right live, and that's what you need to capture, then start with your rehearsal setup and put a mic in front of the band, like an audience.

There's your live sound. If it doesn't sound the way you want it to, then there is a very realistic possibility that your live sound is not actually as perfect as you're thinking it is. But assuming the live sound is what you're after, if you need a little more kick, put a mic in front of the kick drum. And so on. But work fast, and make your decisions practical ones based on what you are hearing, not philosophical ones based on how you think things should be. Don't get caught in the trap of thinking that your live sound SHOULD BE perfect, and therefore trying to force your recording process to somehow fit into an ideal that is based on theory instead of reality.

The ultimate live recordings are orchestral or choral recordings, where a stereo pair is hung in front of a well-practiced ensemble and captures the reality of their sound. The most infuriating and headache-inducing live recordings are million-mic scenarios where you are trying to force a band to sound the way they think they SHOULD sound, instead of the way they DO sound, and trying to make a practical reality fit a philosophical ideal.

Live recording SHOULD be easier, not harder, because you're just capturing a real sound.

You only have two ears, and all you need is two mics (honestly just one, 99% of the time, considering the real ways that people hear live music). Maybe a spot mic here or there to highlight something.

But in practice live recording is often more studio than studio recording. A four-piece rock combo requires more mics and processing than a 120-piece orchestra, because unlike the orchestra, the band expects the recorded sound to be vastly different from the reality of the live sound, but somehow still has it in their head that the live sound is what they are after.

It's like Japanese businessmen who order the most expensive bottle of wine on the menu and then mix it with ice and Sprite because they don't actually like the taste. They have it in their heads that high-class people of refined tastes are SUPPOSED TO have things a certain way, and when they don't actually like it that way, they want it diluted and sweetened and processed so that it tastes like something completely different, but they're proud to consider themselves connoisseurs for drinking it.

Recording a live ensemble is really no different from recording a solo acoustic guitar – you move your head around, see where it sounds good, stick a mic there, check the recorded sound, adjust the position a little, add a second mic if you want to get a little more punch or articulation or whatever, and so on.

Modern drum mic setups evolved from a single mic or pair in front of the drum kit, recording it as the front-row audience would hear it. Clever engineers would stick a supplemental mic in front of the kick and above the snare to up the hip-shaking and hand-clapping, and to simulate the high-volume impact of the back-beat onstage. Gradually, as the kick and snare mics became more central, the mics moved from in front of the kit to above it, to proportionately capture more of the cymbals. Individual mics on the toms allow for dramatic 360-degree drum rolls and eventually you end up with close mics on every kit piece.

None of this is good or bad, but in recent times it has wrapped back around to the point where there is an expectation that every single source will be captured in perfect isolation, with brilliant acoustics, and still will have the same vibe and sonic "glue" of a primitive live recording.

As a onetime professional engineer, those were exactly the projects that I wanted to work on – they took a long time, required professional engineering, and were intrinsically high-budget. But they are like the inverse of the 80/20 rule – 80% of the effort and budget is spent on 20% of the results.

Except the proportion is even higher, more like 98/2. Which is fine if you have the budget and the expectations. There is merit in paying a lot of money to go out for a special meal where every little thing is perfect, where the tables are covered in fresh linen, where each fork is seamlessly removed when you're done with it, where the bread basket is fresh-baked and the butter is fresh-churned and where part of the bill simply goes for sheer real estate because the nearest table is out of earshot, and so on.

But there are a lot of takeout joints that have great food. Wheat flour, fresh tomatoes, basil, garlic and mozzarella can make a pizza that rivals any seven-course dinner at the Ritz. The expensive part of a good meal is the linens and perfect crystal stemware and fresh flowers and the hour-and-a-half spent lingering over a million-dollar view beside plate glass windows and the three waiters per table and the elaborate sides and china coffee cups and all that stuff.

The ingredients of your meal might cost $10, but the experience and peripherals cost $100. And there's nothing wrong with that, if that's what you're after (I mean, there might be something "wrong" with it in a marxist or humanist sense, but it's not like the cost isn't real). And elaborate studio recordings are similarly expensive. It's not just a bedroom computer plus exorbitant markups.

The good news is that you can set up a pizza joint in your spare bedroom that can churn out takeout that rivals the food at the Ritz. The bad news is that the full-blown rock-star lavish studio experience is not fundamentally about the ingredients (although the ingredients are a very important part).

A good engineer can switch seamlessly between between elaborate, big-budget projects and quick-and-dirty small-budget projects, just as a good restaurateur can manage both budget family restaurants and white-tablecloth fine dining. The difference is fresh ingredients vs packaged sauces, the quality of the furnishings and tableware, the cost of real estate, and so on. If you know how to manage a kitchen and waitstaff, you can plug all that stuff into a spreadsheet and it's not all that different.

Better vocal recordings Part 1

Probably the most frustrating and misunderstood part of the recording process is vocals. And it is certainly the most important, and least "fixable" after the fact. It's also the touchiest and most insecurity-revealing aspect of solo home recording. The studio reveals what you actually sound like, instead of what you think you sound like, or what you think you could sound like in a perfect scenario.

Vocal coaching is beyond the scope of this thread and way beyond my skill set, but far and away the biggest problem with most vocal recordings is simply that the singer isn't that good. A good singer has good intonation, a strong voice with full-bodied harmonics (what Pavarotti called "the sun in the voice") and a confident clear delivery.

Intonation – Most amateur singers, in contrast, have iffy intonation, weak-ish voices, and hesitant, uncertain delivery. There are no frets on vocal chords. And please put any thought of auto-tuning bad vocals out of your head for now – that's even worse on a weak singer, it just makes their off notes more precisely off. A singer, just like any other musician, should know what pitch they're trying to hit, and should land ON that pitch.

Singers should practice scales just as instrumentalists do.

A little goes a long way in this regard, especially for singers who have never actually dedicated much effort to it. A week of singing along with recorded scale exercizes in the car on the way to work can work wonders for a singer who has never actually thought about pitch before, and you can bet your bottom dollar that some rudimentary vocal coaching is de rigeur for major-label acts, however punk or indie. Google for singing exercises, or simply find a scale that you can sing both the top and bottom note of, and make a CD of various scale exercises.

Voice – A singer's "voice" is about a million times more important to the quality of a record than the guitar sound or anything else. And voice can absolutely be improved and "learned." Voice is the harmonic and tonal quality of the voice as an instrument. DO NOT YELL OR DO ANYTHING THAT HURTS YOUR THROAT. Seriously – this doesn't sound good and it blows out your vocal chords by causing scarring that renders your voice like a tuneless old smoker's quacky squawk, NOT the full-throated harmonic roar or fire of a metal or soul singer.

And simply shouting at the top of your lungs does NOT improve your voice. It is the first resort of untrained singers who can't figure out how to get the emotional intensity they're looking for. Sing at whatever volume you're comfortable with, but don't do anything that hurts, or that you couldn't do all day. Yelling is like pounding on your piano keys with a hammer – it doesn't sound better, it just ruins the instrument, except there's no way to re-string and re-tune vocal chords.

A quick-and-dirty shortcut to fake "voice" is to sing at whisper-level, and process the vocal through a distortion effect and a chorus or flanger. It's no substitute for good singing, but something to start with while you work on technique.

Delivery – There is a massive catalog of mega-hits that have dumb, clumsy, awkward lyrics and vocal melodies that could have been written by a 12-year-old. If the singer really MEANS what they are singing, then it doesn't matter. It might even be an asset. But if the singer sounds hesitant, or embarrassed, or unsure, it's the kiss of death, no matter how good the material is. Mumbly is the worst sin a singer can commit. The singer has to believe what they're singing.

There is a very tiny handful of artists who have been able to build a career with a vocal delivery based on irony or snide "too smart/cool to be doing this" attitude (see Zappa, Frank). There is a vastly disproportionate number of failed artists who have tried this approach and whose commercial and artistic success does not match their talent level. If you don't really believe in what you're doing, then why should anybody else care about it? Music is not an academic test. There are no points for proving aptitude. If we stop to consider the abject stupidity of such phenomena as Bryan Adam's "Everything I Do", or Black Sabbath's "Iron Man", or the entire genre of disco, it becomes clear that the power of popular music to move people is not based on conceptual excellence or depth, but on some kind of emotional/spiritual/psychic connection that transcends any clinical or academic quality of ideas.

Unless your goal is to create music for college professors, the vocal delivery has to mean something. What it means is almost irrelevant, but it has to be heartfelt and delivered in earnest. Not many 50-year-old men can sing, "For those about to rock – we salute you!" and really mean it, without awkwardness or eye-rolling or winking at the audience. But the ability to sing it and MEAN IT as though your life depends on it transforms an incredibly dumb sentiment into something that inspires millions and that has made countless weekends vastly more enjoyable for innumerable people (not to mention the money).

Don't be too smart or too cool for what you're doing. The kind of cover band who is always winking or smarmy while they show the audience how much better they are then the original band is always vastly less enjoyable than the original material, and they are invariably the first to say that the music business is rigged or all about looks or whatever, because look how they can play anything and still haven't got a hit. It never occurs to them that the reason they haven't got a hit is because they treat music like a commodity, like a roll of toilet paper that they can make cheaper and more efficiently or something.

They're passing all the tests and waiting for someone to give them an A and a million-dollar check instead of doing something meaningful to real people.

Better vocal recordings Part 2

I'm going to discuss vocal recording as though you're an engineer recording someone else. Partly because the following is mostly copied from advice I've given elsewhere in that vein, and partly because this is where the two processes of performing and engineering really start to diverge. So here goes, roughly in order of importance:

1. Psychological preparation

This is the most important part of getting a good vocal recording, hands down. Something about the studio makes many singers tense, pitchy, and forced-sounding. Your primary obligation as a recording engineer is to get the best possible recording, and that starts with the best possible performance. It is your job to make the singer comfortable, relaxed, and inspired. You must be at all times patient, supportive and professional. You are their employee, and should let them take the lead when it comes to the tenor of your relationship. (This does NOT mean that they should take the lead when it comes to the recording process – just that sometimes "English butler" is the best hat to wear).

If the singer wants to be buddies (and they often do), then by all means, oblige. If the singer wants to cuss you out and blame you for their mistakes, put up with it as best you can and be appropriately apologetic and subservient. If the singer looks at you as the boss and wants direction and instruction, then by all means provide it. You get the idea.

Create an inspiring, relaxed environment for vocal takes. Don't leave the singer feeling like they're in the dentist's office or a stranger's living room; make them feel like a rock star.

Keep water or soft drinks handy. If the singer prefers harder stuff, do your best to unobtrusively keep them to a low-level mellow buzz. The best and easiest way to achieve this is by working fast and keeping them busy, which is good practice all around anyway.

If the singer messes up and they know it, just be cool and tell them no sweat, that's what we're here for, 40 takes is typical, they're doing great. If the singer screws up and they DON'T know it, don't tell them they're doing it wrong, just tell them it sounds great, they're doing awesome, and you want to get a couple more takes while they're hot. If they're way off and don't know it, tell them you have an idea and you want to try and run through some possible harmony tracks and ask if they think they could try singing it like "â€¦"(hum the melody). Offer to send a synth part through their headphones with the idea you have in mind, and ask if they would mind singing along to it.

Remember that they're not paying you for your opinions or feedback; they're paying you to make them sound like rock stars. The best way to get them to sound that way is to make them feel that way.

2. Headphone Mix

This is CRUCIAL. A bad headphone mix will make your job and the singer's exponentially harder, and bleed-through is the least of your worries.

Let's start with most overlooked part: Volume and frequency balance. Set the volume of the headphones as low as you can before the singer complains. Turn the lows down, both in the backing parts and on the singer's mic. Human pitch perception at low frequencies is quite poor and gets worse at higher volumes. Bass notes can easily sound a full step flat at high volume, and they are the first thing the singer will hear if the mix is loud. You want the singer's pitch to be glomming onto the midrange, not the bass. If they ask for more low end in the headphones, be aware that more kick will almost always satisfy without screwing up their pitch perception, and that turning up the upper mids of the bass will usually make them happy if they want to hear the bass part louder.

Make sure that they can hear themselves clearly at all times. Compression and presence-range boost on their mic are pretty much required. Pitch and timing are often incidental considerations from the singer's point of view, they want to get nuance and expressiveness and emotion, and if the upper mids are masked in their headphone mix, then they'll start overcompensating. Focus on giving them a crisp, clear, present sound and they'll give you their best performance.

Give them some careful reverb and/or delay or chorus effects. These will have a smoothing and a thickening effect that will make the singer feel less naked and more impressed by their own voice. If you can make it sound like they're singing in the shower you're golden.

3. Mic placement

I assume you're using a directional mic to record vocals. "Generic" starting position is about 8" away from the singer, about forehead level, aimed at their nose (to avoid excessive sibilance or plosives). Use a pop filter, both to control pops and to keep the singer from swallowing the mic.

If you want to get more proximity effect and power and articulation, you can move the mic in closer and aim it more at the mouth. Hard-hitting hip-hop MCs often practically swallow the mic, and you can hear every drop of spit and tooth clicking and it sounds like they're hollering right in your ear.

To get a more spacious, authentic sound, move the mic back a few inches. Forget about Sinatra's mic-cradling live videos and look at the studio photos where he's sitting arm's length from the mic. If the singer is really essy or nasal, try moving the mic further off-axis.

4. Mic Technique

Most singing teachers don't seem to teach this, which is unfortunate, because it's pretty easy and pretty important in this age of amplified and recorded music. It is simply the art of moving further away from the mic when you're loud and moving in closer when you're quiet. If you watch rock stars in concert they do it all the time and it's great showmanship as well as acoustically important.

If your diva has never heard of mic technique, there are two quick-and-easy ways to teach them. Method one is to have them stand sort of sideways to the mic, with their feet shoulder-width apart. Tell them to lean on their back foot when singing, and to lean on their front foot while whispering, and when they're really wailing, to slide their front foot behind the other and lean back on that. This "three position" mic technique is usually really easy for singers to grasp and works quite well.

The other alternative that's even easier and more rock-starish requires your singer to touch the mic stand, which can introduce handling noise, so use a shock mount and approach with caution. Have the singer hold the mic stand just under the shock mount, with their arm bent about 90 degrees. When they whisper, have them pull in close to the mic, and when they wail, have them stretch out their arm all the way. Moving the mic stand is tres rock star, but introduces more potential for handling noise. Getting the singer to move their torso is better in the studio.

One final tip about mic technique is that you have several tools at your disposal to keep the singer placed correctly, with or without their cooperation. One of my favorites is the "dummy mic," which works wonders for singers who can't resist the taste of mics in their mouth, or who don't understand the concept of "off-axis". You simply set up a mic for them to chew on, swallow, spit on, whatever (a shure SM58 is a good pick) and then set up the "real" mic behind it or off-axis or whatever. Whether you tell them that's the real mic or just an extra ambient mic is up to you.

Another useful trick to reinforce mic technique and to guard against straining is to mix in a little bit of a separate bus of the vocals to their headphone mix that is fed through some heavy compression, distortion, or even digital clipping (the "dummy mic" is a good place to get this separate feed from). This serves a similar function to grooved pavement on the side of the highway. It gives the singer an early warning when they're about to go in the red. Sort of a subconscious cue to back in your lane.

5. Studio tricks and mixing techniques

This is not even close to a comprehensive mixing guide to vocals. But I will include a few quick tips that are relevant to think about as you record.

Motown compression (a.k.a. New York compression – don't ask, I don't know). This is a very useful technique for situations where you have a dynamic, expressive vocal track where you need a way to keep the musicality of the performance but also find a way to push the lyrics and the articulation out in front of the mix. You basically clone the vocal track, and apply heavy compression and presence-range eq boost (somewhere between 4-10 kHz) to the clone. Now you can treat the main vocal part like any other instrument, using reverb and dynamics and tonality and whatever, and then just dial up enough of the compressed clone to keep the articulation and clarity. Knowing about this technique can also help keep you from overcompensating as you record.

Doubling the vocal track – having the singer sing along with him/herself can thicken up and even out a thin, uneven, weak, or subpar singing voice. This is easily overused, but on a lot of hard rock records, a combination of low cut and doubled-up tracks is what turns poor singers into powerful rock stars (think Linkin Park). Chorus or delay effects can also be employed with similar results.

The "whisper trick": Having the singer whisper along with the vocal track in a monotone can be a quick and easy way to get a "huge vocal" sound. Again, easily overused, and most effective on weak vocalists in dense mixes.

Autotune and it's offspring: Avoid using it indiscriminately on the "auto" setting. If you have a great performance with one or two off notes, just adjust them manually. If the whole performance sounds off-key, you need to evaluate realistically what the singer is capable of.

> Quote:
>
> Originally Posted by Fritz
>
> Thanks Yep, great stuff. I'm still only on page 5 but in your honor I figured I'd start right at the beginning of my chain and re soldered the connections inside my strat and set it up fresh. Made a huge difference

Thank you for validating all this. Anyone who actually takes a moment away from plugin-shopping to get back to actual sound makes this worthwhile.

> Quote:
>
> Originally Posted by spikemullings
>
> I don't want to put you off your stride yep but if you have chance could you say something about editing vocal performances for breath sounds?...are there any first principles that hobbyists like me should be aware of?

Unless there is some specific reason to do otherwise, get rid of them. Do this in the "pre-mix" stage. Just zap ,em, and don't look back. 99% of the time, the performer would not have "performed" those breath sounds if they could have been avoided, and 99% of the time, trying to mix with them is going to be vastly more difficult. Finding a "place" for those breath sounds that is still clearly audible without being really distracting is a huge job. And if there is no real "place" for them, if they're just going to be subliminal textural elements, then most of the time, they are going to end up as noise, basically, mucking up your definition and clarity. The fact that they occur in the most sensitive range of human hearing doesn't help.

Are you planning to compress and eq these breath sounds so that they are just as prominent as the vocal line? If the answer is no, then why would you want them in the track?

"Son of a Preacher Man" illustrates perhaps the single best principle of getting good vocal tracks: Get Dusty Springfield to sing it for you. The reason why brilliant singers often have more artifacts in their tracks is because they deliver perfect tracks that are simply left intact.

But the smooth, sensitive, natural breathing of a true professional with fluid mic technique is often very different from gasping between notes that is going to turn into a vortex of white noise the second we put a compressor across a modern vocal track. I don't know how much time you have, but if you find yourself trying to de-ess breathing sounds instead of just getting rid of them, ask yourself how important this really is to the performance.

This is similar to finger squeaks on guitar or grunting or performance noises from a pianist. Some artifacts that we tolerate or even embrace from Glenn Gould or Andres Segovia are not things that sound good when your cousin does them at recital.

Obviously it's your call, and if the track sounds better with breathing noises, then clearly they should be left in. But when it doubt, cut them out. Don't invest effort to make them sound good, because if they do not obviously improve the track then they should almost certainly be cut out.

> Quote:
>
> Originally Posted by spikemullings
>
> ...I find myself conflicted between wanting to excise everything unnecessary to the lyrics and melody and at the same time wanting to retain some of the emotional resonance and naturalness that a little breathing noise can give...

This is actually a very important distinction, and I daresay a pretty common dilemma. Here's the thing: what if the "naturalness" that we are trying to preserve is embarrassing and bad?

I'm not saying this is the case with your project or anyone else's, but a lot times there is this sense that there must be some secret out there that lets you turn ugly gulps of air and wheezing into the smooth, sophisticated, conversational delivery of a great crooner or some such. If there were a plugin that did randomized breath sound sound replacement, people would buy it ("breathagog"). To sound natural. And they would use it with three tracks of vocals stacked-up, 12dB of compression, huge eq rips, autotune, and pitch-shifted delay. To sound "natural."

I am not opposed to sounding natural. I'm actually a big advocate. If you can get a vocal track that you can just drop on top of the mix, add some reverb, and call it a day, then you certainly don't need my advice but if you're doing the whole multi-track-and-process thing, especially if you use double-tracked vocals, as is usual these days, then I don't even know how to fit breath sounds into such a thing.

It is important to be in touch with disconnects between philosophy and reality. In time, the most fortunate and gifted among us may come to live in a world where there is no disconnect – where the daily practice of our lives is as we think it ought to be. But a lot of time is wasted when we use approaches based on what we think SHOULD BE the material we're working on, instead of the stuff that is actually in front of us.

If you're standing in the doorway at Burger King, waiting for the Maitre'd to seat your party and bring menus, then you are apt to find the experience more disappointing and frustrating than it has to be. It's important to be realistic in your expectations, and prioritize accordingly.

> Quote:
>
> Originally Posted by dstone55
>
> ...if you want to see what 1 idiot can do with a bunch of instruments, an Mbox, a laptop and Reaper... and the information on this thread... go to the page linked below and listen to the song called "Demo – Baron Haymows Junket" (a work in progress...) Keep in mind, that is a shitty 192 bitrate mp3...
>
> http://www.reverbnation.com/nobodydigs
>
> Diesle (David) from The Magnetrons

Sounds great, David! Love the dynamics – big, modern, but still punchy, spacious, and "real." And the playing is outstanding. The rhythm section is fantastic. Frankly a lot of commercial studios would have murdered this material. Aside from the real horns, I don't think money could buy a much better recording.

Kudos.

(PS I do plan to post more once I get some thoughts in order – requests and questions are always welcome.)

> Quote:
>
> Originally Posted by Chris_P_Critter
>
> ...It's one thing to not be afraid to make mistakes during the learning process, but another to disregard the absolute basics in the hopes of surpassing what many would consider to be too steep of a learning curve...

That, and also that the learning curve is actually a lot less steep than it might look if you get out of the "black art"/magic ears mindset and just focus on the sound and the tools in front of you. To John McCain, people who can send email look like computer geniuses.

There is a lot more to say in this thread, but I started a kind of related spinoff that interested readers might want to contribute to here:

http://forum.cockos.com/showthread.php?t=32580

The focus in that thread is more big-picture production stuff, while this one will continue to focus on nitty-gritty engineering techniques.

More stuff on vocals

Singers (and all musicians) should really invest in some kind of portable recording device.

Singers have comparatively little gear to invest in, so it should not be too much to ask them to pick up a little pocket recorder. Doesn't have to be anything fancy, just a $20 micro-cassette job will do. The digital ones are often just as cheap, and smaller. I keep a little Olympus deal with a built-in USB plug in my pocket. The purpose of this device should be self-evident for anyone interested in audio. It is a super-easy way to record ideas, to test out different rooms, to record something inspiring or cool, and so on.

But for singers it has a special purpose, which is to tell them how they actually sound. The mics on even very cheap devices are actually quite accurate. They are often noisy and have built-in compression, but the former is irrelevant and the latter is actually a plus when it comes to vocal practice tools.

A great many people are quite taken aback by the sound of their own recorded voice.

...

I'm not sure what to say. This is a sensitive topic.

If you think you have a good voice but don't like the way it sounds on playback, chances are 100% that you do not actually realize what your own voice sounds like. You are one of those people who thinks they look terrible in photographs, but who actually looks just like they look in photographs.

The good news is that you are not alone. The bad news is that yes, that is what you actually sound like. The best news of all is that a tiny little bit of dedicated practice can get your voice very close to the sounds you imagine in your head.

The human voice is the most versatile and capable instrument of all. “Range” is not nearly as fixed a factor as people think it is. “voice” and “timbre” are infinitely changeable.

At the risk of sounding sexist, there are an awful lot of singers who approach singing the way an untrained girl approaches firing a gun for the first time. Anyone who has ever witnessed this phenomenon knows exactly what I am talking about. She does not AIM the gun but instead holds it wildly as far away from her body as possible while covering her eyes and ears with her other arm and scrunching up her shoulders, as though the gun is just some kind of dangerous explosion that she needs to be as far away from as possible, but that will somehow hit the target of its own accord. Something like a young little-leaguer who is afraid of the baseball and who shuts his eyes and leans back and swings wildly, as if to chop down a monster with the bat.

Both of these approaches are of course incredibly dangerous, but somewhat natural reactions to unknown and potentially dangerous scenarios. The reflexes take over, and the conflicting impulses to run/fight/hide are all fighting each other. Of course the right way to fire a gun, or to hit a baseball, or to sing a note is to breathe deeply, stay calm, focus on the target, and execute the action. Easier said than done.

Men tend to yell when they are uncertain of the pitch, and women tend to either shriek or mumble. Both are ugly, although certain singers have developed a weird kind of artistry when it comes to tuneless yelling (Keith Morris of Black Flag and the Circle Jerks comes to mind). Men also tend to want to try and extend their range downward into atonality when they can't really sing, and women often try to extend their range upward and cover up the pitch with vague melisma. Both sound silly and amateurish. As well as completely unnecessary.

If you have any musical talent at all, then you have some degree of pitch perception. If you can tune a guitar, then you can hear pitch. And if you can hear pitch, then you can hum along with a steady pitch. Find something humming and hum along with it. A single-coil electric guitar's hum is a great place to start if you have never done this, seriously.

There was a loud-humming electric transformer box in the subway station near where I used to live where the singer and myself would practice humming intervals against the steady note of the transformer while we waited for the train. We would just stand there, mouths closed, humming different intervals against the transformer. It was hard for other people to tell why the harmonics kept changing. Best vocal exercise I ever encountered.

Just find some loud-ish steady tone and start humming until you find the right pitch. It will be obvious, because your chest will start vibrating. Play a long synth note if you have nothing else to sing with. Once you find the unison or octave note, it should be pretty easy to find fifths, fourths and other consonant intervals either above or below the reference pitch. You don't need to know what interval you're singing, the idea is just to get the vibe of what it feels like to sing the "right" notes. You can feel it resonating in your chest and sinuses, and it's obvious when you get it.

This is by no means a comprehensive guide to singing, but a little goes a very long way in this regard. A lightbulb goes off the first time you get that resonance, and from there on, your voice starts to become an instrument that you can control instead of a dangerous weapon that you don't know what to do with.

More on "voice" later.

> Quote:
>
> Originally Posted by stupeT
>
> ...Would you aggree that some reverb (typically much more than in the final mix) in the monitoring is helping vocalists to keep the right pitch?

Also compression and eq. Most singers do better when they hear a "hype" and "big" version of their voice in the headphones. Others prefer to sing with one can off, or just listening to open-air monitors.

But a muffled headphone mix where they are mostly only hearing the dull resonance inside their own skull is usually the worst.

Recording electric guitar

So let's talk a little bit about recording electric guitar. This is a frustrating and sensitive topic for a lot of people. Guitar players often have a significant personal and emotional investment in their "sound." A lot of them can be almost as sensitive to criticism as singers. And they are usually right, although not always right in the right ways,when it comes to studio recording.

When Jim Marshall first began making guitar amps to sell in his drum shop in London, his objective was not to re-invent the sound of modern rock music, it was to make less expensive knockoffs of popular American imports, specifically the Fender Bassman. There was no distortion or "drive" circuit, but players discovered that by turning all the knobs up, they could get the amp to really start breathing fire in ways that put busted-speaker "fuzz" to shame. And popular music would never be the same.

The sound came from a lot of factors, most notably from overloaded preamp and power tubes (especially the then-cheaper EL34s instead of the american-used 6L6), and from excursion of durable but primitive speakers that "fattened and flattened" when pushed to their limits. In the time since, the sound has come to be called distortion or overdrive or high-gain or any number of other things, but it was an unmistakable turning point in music, marked by the most famous customer of Jim Marshall's London shop, Jimi Hendrix.

Since then, there have been countless devices that have aimed to duplicate, refine, or expand upon the "Marshall" sound, and "distortion" has become the trademark sound of electric guitar.

Broadly speaking, the sound of electric guitar amplification diverged into two predominant tracks – cleanish, punchy, bassier "Fender"-type sounds, and saturated, roaring, midrangey "Marshall" sounds. The older "fender" sound is a thunkier, punchier, twangier, and snarlier tone that was actually developed and refined before the solid-body guitar was even invented. Some of the best examples are actually old WWII Gibson Amps. But the Fender name became associated with electric guitars, and there you have it. Primitive, cold-war-era tube amplification (of either Fender or Marshall type) exaggerates the best aspects of electric guitar, which are specifically the unmatched expressiveness and performance nuance of the instrument.

Electric guitar is a crude and primitive-sounding instrument. It does not have anything close to the refinement or richness of a good string instrument, it does not have sparkle or clarity of its acoustic cousin, it does not have the depth or versatility of a piano, and it never quite matches up to horns for boldness and acoustic power. But it does have an unmatched range of sonic texture and expressiveness of performance gesture, second only to the human voice. And distortion puts the performance nuances right out front with the actual notes.

Tube amplification exaggerates the rasp, chirp, growl, thunk, fatness, and slinkiness of pretty much any instrument, but it's an especially perfect match for electric guitar. Solid-body electric guitar is a very bad-sounding instrument without flattery. You can test this by putting a microphone in front of an un-amplified electric guitar. Or just listening to one.

It sounds bad.

Pickups are not microphones. They are very crude magnetic transducers, and they require amplification to make sound, and they generally require amplification artifacts to sound GOOD. We are starting to get to the heart of the reason for all the preceding jibber-jabber.

Electric guitar is an ELECTRICAL system, not an ELECTRONIC system, and definitely not an ACOUSTICAL system. This means that EVERY SINGLE ASPECT of the audio circuit affects sound. Something as simple as minor component variations in a knockoff circuit COMPLETELY ALTERED the course of music history in ways that could not possibly have any parallel in other forms of audio. An electronic synthesizer may sound better or worse or slightly different with one brand of capacitor vs another but it does not effect the absolute sea-change in sound of something like the difference between a Marshall amp and a Fender amp, or a Les Paul vs a Strat.

An electric guitar does not have any sound that is not electrical. Even if we are using electronic digital or analog processors to re-create the sound, they are invariably emulating electrical systems, and NOT intended to deliver "purer" sound.

What this means is that with electric guitar: EVERYTHING matters. The input impedance, the wire gauge of the pickups, the excursion of the speakers, the voltage output discrepancies between pickup positions, the volume setting of the output amp, the speaker impedance, the tone settings before and after the input stage, everything. And there are no right or wrong answers. And the pick gauge and material sure as hell matter.

More to come...

[begin page 11]

I fear that I might start to sound like a broken record as we get more into specific instruments and practices, but the reality is that the same principles apply over and over again. When it comes to recording electric guitar, the most important thing is to make sure that we are actually starting with the player's "sound." One man's trash is another's treasure, and it's not the engineer's job to decide whether the guitar should sound like My Sharona or Cannibal Corpse or Charlie Christian.

For the home recordist, one of the problems with most "how to record guitar" guides is that they presume that the player's sound has already been worked out and well-established. This is usually the case with major-label artists who have already established a following, played hundreds of concerts, and who have had the opportunity to use advance money to shop through dozens of amplifiers.

But what of the Joe Blow who started this whole thread, he of the Squier Strat and the Peavy amp? How is he to know whether he would hear a bigger improvement from Lace Sensor pickups, or from a vintage tube amp, or from a modern modeling half-stack, or from buying a $3,000 Les Paul, or an original Ross compressor pedal, or from a POD vs a V-AMP vs a Johnson J-Station vs actually buying a real tube amp? For that matter, does he even really know for sure what "his sound" would be, even if he could have it for free if he simply named it right now?

He could ask in a web forum, and get a hundred different answers (see the very first post in this thread). And all of them probably have some merit. But one man's treasure is another's trash, and advice from James Hetfield or Stevie Ray Vaughn or might be of limited usefulness to a budding Andy Summers or Chris Isaac. Moreover, when a complete guitar rig could cost anything from $120 to $12,000 or more, it becomes difficult to priotitize. Especially where we're not talking about a Grand Piano or a Renaissance-era violin, but just about the staple of working-class garage-rock.

We start to get into hand-wired, discrete component this-and-that, and all-tube transformerless-output whatever, and hand-wound pickups and fourty-year-old paper capacitors and so on, and it's all basically doing the same thing as a twenty-dollar piece of wood with some thin-gauge wires wrapped around magnets.

I wish I could give a simple answer, and say that all you need is a V-AMP, or even an all-tube Marshall half-stack. The reality is, as I said earlier, that EVERYTHING matters when it comes to guitar sounds. The beautiful and terrible reality is that every single thing changes the sound of electric guitar. And unlike most other instruments, there is nothing close to a consensus. If you ask 100 top concert violinists which violins sound best, 99 of them will say a Stradivarius. If you ask 100 guitar heroes which guitar sounds best, you'll get 85 different answers, and never mind the differences in amps, effects pedals, picks, strings, and so on.

Moreover there is a hugely interactive aspect to good guitar sounds. Someone used to playing a Strat who picks up a Les Paul is apt find it a tone-killing blandness machine that makes every note and chord sound the same, whereas someone used to playing a Les Paul is apt to find a Strat to be an uncontrollable inferno of string noise and fizzy pick attack.

This fact, this reality, that very good guitar players often have wildly divergent opinions on the "right" gear starts to indicate a possibility that is not often considered. Especially when we consider the irrational and otherwise inexplicable reverence for "old" gear. Why would a simple guitar amplifier from 40 years ago sound any better than one made to the same specs today? Hold that thought.

We might reasonably speculate that modern lumber from heavily-irrigated rapid-growth forests might not resonate or sound the same as the dense-grained old-growth wood that was the norm in the mid-1900s, but why should ELECTRICAL circuits sound different? Moreover, why would old guitars sound any different from new guitars made from old wood?

I'm going to suggest a possible theory that, so far as I know, is unique. And that is that a majority of the "sound" achieved by the guitar greats was not from the gear, but from the player. On first pass, from a conventional gear-nerd POV, this might seem like no insight at all. Of Course Jimi Hendrix (or whoever) contributed more than the gear did. So we brush that aside and ask what gear do we need to get the same sound, given that we are playing like Jimi Hendrix?

But consider the possibility that it is not gear that defines the sound, but the player. That Jimi Hendrix would have developed a brilliant sound if all he had was a Johnson J-Station and a Kay guitar. Would we then be obsessing over Kay pickups and J-Stations with original EPROM chips? Would Jim Marshall still be running a drum shop in London selling Fender amp knockoffs?

I'm not saying that gear doesn't matter. I am suggesting the possibility that guitar players develop their sound in conjunction with the gear available to them. Link Wray invented distortion by punching holes in his speakers. Were those "vintage" holes? What if he had put them in the wrong places? Maybe, just maybe, given that the entire stream of guitar signal is a sequence of distortions and nonlinearities, maybe what matters is not so much the minutiae of the gear, but how the player manages and reacts to the distortions in real time. Maybe Jimi Hendrix would have sounded just as good using an amp with KT66 tubes with diode rectification, or a custom-overbuilt American 6L6 tube amp overloaded by British 230V line voltage.

I do not believe for a second that all guitar gear is created equal, but I also think that, given a certain modicum of sonic adequacy, there is a possibility that obsessive pursuit of everything vintage reaches a tipping point where it becomes fetishism.

More to come.

> Quote:
>
> Originally Posted by BoxOfSnoo
>
> Oh I agree with you, but here are possible answers:
>
> 1) Components have changed over the years. Germanium has been replaced by silicon, for the most part. Very different sounds...

I know better than to argue with guitar players over what makes for good tone. But when we start to get into loose tubes and germanium vs silicon and so on, consider: Can you listen to a recording of a guitar and tell right off the bat what material the transistors were made of? Can you tell by listening how loose the tubes were? What do we mean by "very different sounds"?

Because the thing is, you usually CAN tell just by listening whether the player was on the neck or the bridge position pickup, how heavy the pick and string gauges were, what fret position they were playing in, and so on. But how many tone addicts actually START with those things when they're looking for better sound?

An awful lot of them use light strings and heavy picks because they're easiest, they stay on the bridge pickup because it sounds brightest, they hold the pick the way their guitar teacher first showed them to, and then they wonder why their tracks sound fizzy and weak compared to Stevie "013s" Vaughn or James "pencil grip" Hetfield. And out comes the credit card.

Moreover, none of the examples above actually address "vintage" gear. You could wiggle the tubes to loosen them and excurse the speakers by playing some Micheal Jackson records at high volume through the amp. Would that make the amp more desirable? How much of the difference-hunting is really grasping at straws for an explanation of why the old records sound better? We could replace the transistors and components with ones that match the measured output-spec of germanium or whatever. And so on down the list. Has even a single copy of any record ever been sold or not sold because the guitar player had equivalent output-spec transistors made from the wrong material? More to the point, was Jimi Hendrix's amp a bad amp, because it wasn't old at the time? How about his guitar? They sound pretty good to me on those records that were made with then-new equipment.

I realize that the answers are long and complicated and that I'm not framing these questions the right way for vintage gear hounds to answer. And my point is emphatically NOT that a beginner guitar/amp combo from Wal-Mart is just as good as a ‚67 SG and a Marshall Plexi. Every little component does affect sound, and no two guitars or tube amps sound exactly alike. And the sound IS constantly in flux, as tubes, strings, and speakers age and as mechanical components settle and age, and so on.

And my point is not to dissuade anyone from tone-questing. Finding better sound, by any means, is exactly what this thread is all about.

This is kind of a fine distinction that I'm drawing, and I'm purposely over-stating the case a little to counter-balance the widespread implicit assumption that it's “magic gear” instead of magic players who make rock guitar sounds come alive. There is an understandable tendency among musicians to look outward for the problem rather than inward. My intent is not attack anyone's self-esteem, just to point out that actually focusing on good technique and careful listening is always the best practice, on any gear budget.

I can't tell anyone what is a waste of time and money and what isn't. You have to decide that for yourself. But I can tell you that you WILL waste a lot of time and money if you get fixated on the theory of what's SUPPOSED to sound good or bad, instead of focusing on what actually DOES.

> Quote:
>
> Originally Posted by TedR
>
> I have a brief question Yep.
>
> I've noticed that the more I mix the more" Aha" moments I have as I discover new techniques to achieve the sound I am looking for.
>
> But, there is so much to remember that I find it easy to forget these tricks and techniques over the course of time.
>
> I have considered taking exhaustive notes, keeping a recording diary, etc.
>
> What would you suggest ?

Hmm... I might not be the best person to ask, since I hardly ever remember what I did on anything. Actually, I'm not quite sure I even understand the question...

Are you talking about stuff like eq settings or something? You could just save those as a preset. In fact, THOSE are the potentially useful and meaningful "presets", far more so than anything you read in a book or get with the plugin. Something that always drives me nuts is when some famous producer or engineer says something like: "I always boost acoustic guitar about 4dB at around 8k or so, and cut a few dB at 200Hz"(or whatever).

These guys have no idea of the damage they cause with these offhand comments that get passed around as gospel for years afterwards. The problem with the above statement is that it is very often 1. not actually true, 2. only applicable to the one particular guitar/mic/room setup that they use most commonly, and 3. probably almost completely irrelevant to how the guitar actually ends up sounding in the mix. Here's the secret: Dollars-to-doughnuts, this producer has a favorite console or preamp that has a built-in eq with fixed high-and low-frequency knobs for 8k, 200Hz, and probably a sweepable mid. So they plug into the mixing console and as a matter course slightly boost the highs and slightly cut the lows, except when it sounds better not to. And since they are usually recording with the same mic and placement and often in the same room with the same player, chances are pretty good that what worked the first time around will work pretty good next time. So they probably DO almost always use those frequencies in those ways.

But if their console instead had a shelf at 11k and a low knob at 300, then they would probably be using those frequencies instead.

More to the point, if they were using a different mic or different placement then they might take a completely different approach. If they were recording with a Talkamanie Artist series instead of a Gibson super-jumbo, they might do the OPPOSITE.

But the WORST part about it is that this offhand comment almost certainly has NOTHING to do with how they actually mix and process the track. It's just something they do along with mic placement to get the basic sound. They leave out that on their most recent arena-rock hit with the background acoustic strumming track, yeah they bumped the console hi eq up a tick, but then at mixdown they shelved off everything below 2kHz, gated the track slightly, compressed the hell out of it, rolled off most of the highs above 10k, added a delay and exciter, and then sent it through another stage of compression triggered by the vocal bus, in addition to sending it through the drum reverb.

But of course all that was completely different from what they did on the mega-hit before that, when the guitar was a featured solo instrument on an intimate ballad. So when the interviewer asks: "any special tricks or eq you use to get good acoustic guitar tracks?" The famous producer answers truthfully what he does when tracking acoustic guitar, and the entire internet passes around the certainty that the way to mix acoustic guitar is to use the magic frequencies of 200 and 8k.

Not sure if that answered your question, but you SHOULD feel free to make your own "presets" for stuff you record all the time, as long as you don't get locked into thinking of them as closed-ended "recipes."

> Quote:
>
> Originally Posted by BoxOfSnoo
>
> Well strictly looking at the transistors, yes, most of us can, golden ears or not. It shows up all the time when a person hears a killer tone and tries to replicate it. They just can't seem to get the same fuzz tone because their silicon-based fuzzface is too harsh. You can hear this live as well as recorded, so it's not purely a recording phenomenon, either. It's at least a big of a difference as string gauge! Maybe focusing on fuzz is cheating because of the huge significance of the transistor makeup.

I'm going to push this, because I think it's an important distinction, and maybe I phrased the question badly the first time around – are you seriously arguing that you (or anyone) can reliably hear the material used in transistors, to the point where if I posted 20 guitar clips you could tell which had germanium and which had silicon transistors, and which had something else entirely?

Note that I am emphatically NOT talking about taking a fuzz-face and clipping out and replacing the diode (which anyone could hear), I'm talking about the difference between circuits designed from the ground up to sound good using whatever kind of components.

Saying that a fuzzface sounds different with different transistors is a categorically different thing from saying that the presence of a silicon transistor automatically imparts a specific sound to any circuit. If you were to design a circuit using silicon transistors that deliberately introduced reverse leakage comparable to a germanium transistor I bet you'd have a hard time telling the sonic output apart. I mean, you might be able to to tell one from the other in a straight A/B test, but I doubt that you'd be able to do much better than guess which had the germanium transistor.

> Quote:
>
> But you know there's a difference in broken-in equipment. It does sound different. Really, it does. My point was, older well-used equipment is almost always different sounding than fresh off the factory, even if you have the same circuit and same specs. So you can't simply dismiss it saying that it's just in our imagination. There is some science to it, and we haven't really nailed down exactly what it is that makes it sound "vintage".

We haven't nailed down anything. Older equipment sounds different, but so does one piece of newer equipment to the next. Two Stratocasters that came off the same factory line on the same day will sound different.

My argument is emphatically NOT that the difference between one piece of kit and another is illusory. In fact I stipulated pretty early in this thread that EVERYTHING matters. That COULD be read as a reason to pursue every picayune detail down to the Nth technical degree, or it could be reason to just leave it up to the amp and instrument manufactures to figure that stuff and just find stuff that matters to one's own sound. Either approach is entirely valid.

There is a bit of dialog in the film Time Bandits that goes something like this: "So now you're the leader of this group?"

"No, we agreed not to have a leader."

"Right, so shut up and do as I say."

Wherever there is controversy or uncertainty, interested parties will rush in and use the UNCERTAINTY ITSELF as proof that they are right. This can be seen everywhere, in a lot of political debates for example. The line is that if you can't prove X, therefore the truth must be Y. Which is patently false as a logical test.

Please note that I am not accusing Boxofsnoo of anything like this. But when some people are saying the issue is black, and some white, it is very hard to make a sincere case for the answer being "unknown" without being pushed into one camp or the other, or without having people therefore read you as saying it is some shade of gray. Gray is not the same as unknown. And you don't have to espouse one side to doubt the conclusions of the other.

> Quote:
>
> ...So each guy needs to know where to go shopping to get the right tone. Sometimes you need to start in the vintage aisle.

I have no disagreement at all with people whose favorite piece of gear is "vintage". If anyone is certain that loose tubes or germanium diodes or old speakers are the key to great sound, then I have no argument with their personal preferences, but I do expect a technical defense if they expect their assessment to be treated as empirical fact.

> Quote:
>
> Originally Posted by TedR
>
> ...The thing I have noticed is that it is easy to forget things like specific mic placement, preamp settings, Bass tone settings, and such if I have to stop recording for a week or two (because of work and life) and then come back to it...

Oh, yeah. DEFINITELY keep copius notes on that kind of stuff. That's what I was talking about in the beginning of the thread. Low-residue painter's tape from the hardware store.

Stick it on everything. I have it all over the place, with little circles and lines drawn to indicate knob position, and above that I'll have something like "BLP": stands for "Big Les Paul" sound – circles for every knob, BLP V, BLP T, BLP B, BLP G, BLP R – you don't have to know what they mean, because I do. Stuck on the amp, on the preamp, with an indicator of which mic, and so on. Anytime I want to record my own stock "big Les Paul" sound, there it is. I have similar tape markings for SNF (which stands for "snarly Fender" sound), GRB – growly rock bass, BFB – burpy funk bass, HFB – hollowbody fretless bass, and so on.

MMV8 means "medium male vocals 8 inches away" for my-go vocal mic. SFV0 means "soft female vocals close miked" on same.

On my mic stands, I scratch lines into the metal to indicate common positions. That makes it quick and easy to set up a boom stand and lock it into place, knowing that it will line up at for instance the right height to get the top speaker of a slant cab. I actually keep a steak knife on the recording desk for this purpose that also helps to motivate singers. I have tape on the angle-adjustment to mark the angle. So if I have to set up in front of the same amp for the "big les paul" sound, I know to set the boom arm to the third scratch, the height adjustment to the lowest, and the angle to the BLP3L mark, and I'm going to be awfully close to where I was last time as long as I remember to aim the mic across the speaker cone almost touching the grill. And that much I remember just from taking the time to jot down those few marks.

In REAPER's "project settings>notes" I ALWAYS write the key and/or rough chord progression and performance notes (something like "verse ADAG chorus DEDCBA bass 1415 except turnaround fill"). It's not a proper lead sheet but it is enough that if I re-open in three years I don't have to go hunting for chords. It only has to be enough to remind me, it doesn't have to be a diary.

In the same field, I always include production notes. Who what where when how. E.g. BLP, RAT, GRB, Jfdr, bsn1, MMV 8, SVF12x2, DimHO78.

To me, the above reads Guitar 1 Big Les Paul; Guitar 2 Raunchy Archtop; Growly Rock Bass; Jeff's Drum kit; Birch Snare 1; Male medium vocals 8"; Soft Female Vocals 12" double-tracked; Dimension Pro set to Hammond Organ 78 preset.

I know that "big les paul" means the little 5-watt tube amp with the gain, eq, and reverb settings marked in tape, and the mic stand notched. I know that "growly rock bass" means the heavy-body maple-neck bass through a sansamp (settings marked in tape). I know that my notebook has the standard setup for "Jeff's Drum Kit" if I don't remember it. I have little pieces of tape tape all over the oriental rug to mark where different drums and mic stands go. I know that Female Vocals 12" means that they were tracked in the corner to the right of the couch, and so on. I could probably do punch-ins on a 3-year-old project using these notes.

You don't have to use my system. In fact you shouldn't. The most important thing is to make it easy, otherwise you won't do it. Just make it easy to take notes, and to keep them in a place where you will find them later. YOUR NOTES DON'T HAVE TO BE PERFECT. They don't even have to be very good. They just need to jog your memory, they don't have to be a historical tome documenting your exploits for future generations. Approximate knob settings and positions are fine, since it's always going to be a little different anyway.

This is why you need that organization and PAD OF PAPER that I mentioned earlier. That way, when you think of something, you can WRITE IT DOWN. Forgot how you set up that awesome sound last weekend. WRITE DOWN ON YOUR PAD OF PAPER: "Figure out way to remember awesome sounds." Then get on with recording. Then, your PAD OF PAPER will remind you to buy painter's tape, sharpies, and to make notes next time.

The PAD OF PAPER is your producer, telling you in the cold light of sober reflection what was good and bad about each recording session, keeping track of the details that need to be worked on for next time, noting which things you were and weren't happy with at different times and providing a kind of emotional ballast against the temptation to reinvent everything every time you get frustrated or annoyed.

Monitors and a pad of paper. It's all you need. (and maybe some painter's tape).

> Quote:
>
> Originally Posted by BoxOfSnoo
>
> Well I'm not gonna fight you, yep. You clearly have more experience than I do. But I can hear it, so I think most anyone can...

Well, whatever my level of experience (and it's not all that), my first piece of advice would be this: Never trust anyone just because they have more experience than you. Experts are the easiest people to fool. This is not personal opinion, it it proven fact in a whole lot of scientific peer-reviewed studies. My favorite example is the wine experts who failed to detect that they were drinking white wine with red dye, and who also failed to detect when they were drinking the same wine from different bottles:

http://scienceblogs.com/cortex/2007/...ource=rss_feed

There is also the great monster cable vs wire coat hangers listening test:

http://gizmodo.com/363154/audiophile...-a-coat-hanger

My whole point is not to brow-beat anyone with ad-hominem arguments of "I know vintage gear and you don't", it's the opposite. Look at the emperor and see what his clothes look like to you. "I have more experience" or "I know X and Y better so therefore I must be right about Z" are stupid arguments, and patently false logical tests. And they are exactly the kind of arguments that most gear debates fall back on. Which is exactly wrong, whether they come from me or anyone else.

I really hope I haven't done anything to set myself up as an "authority" here, because I'm not. If my opinions disagree with your own ears, you should say so, and you should let other people know where I'm wrong (because I probably am, in any number of opinions). I am myself disagreeing with a lot of people with much better credentials than myself, in this thread. That's the whole point, and I actually wish more people would disagree, as long as it is meaningful, substantive divergence of opinion.

But as far as I can see in books and on the web and whatever, most of the discussion is frankly a bunch of know-nothings debating what they've read about other people's opinions. (I'm not accusing you or anyone else in particular of this.) I personally have never done a study of trying to construct the same signal output from silicon-based vs germanium-based transistors. So I have no idea whether it is possible. I know there are a boatload of people online who think that germanium sounds better, but there are far bigger boatloads of people online who are wrong about all kinds of things.

And most of the "germanium-is-better" folks seem to ultimately fall back on the "well, everyone knows its better" or "Jimi Hendrix used germanium" arguments, which are worse than useless.

Until Jimi Hendrix came along, everyone knew that Fender amps were the best. Everyone knows what everyone else knows as long as everyone knows the same thing, even if it's not true.

I'm not trying to pick on you. I've been the main person posting in this thread so far, but that doesn't make me or my insights more important or more useful than anyone else's. If I harp on divergent opinions it's not because I'm trying to bully dissenters but because proof of concept is helpful to me, personally, and I hope to others. I so far do not even disagree with you.

> Quote:
>
> Originally Posted by thinking allowed
>
> In my town, 9 bands out of 10 think they're good enough to record live. Do you have any tips on the ‚rattling snare' that happens during live recording in a small room?.....I swear that better drummers (with more expensive kits) don't have this problem as much. Am I crazy?"

Reply from Smurf

FINALLY, a question I know something about!

The best trick I have used, being a drummer, to get rid of that rattle is to de-tune 1 lug that is right beside the snare bed. It usually only takes 1 turn, but by slacking the head tension at the point where the strainer comes off of the snare bed, it releases a lot of the tension across the head that is directly under the snares, without affecting the tone of the snares like taping, or putting small pieces of napkin or cloth between the snares and the head.

Also, the better kits have better tolerances on the bearing edges of the shell, and the angle & depth of the snare bed itself, thus making the better drum set easier to tune.

Hope this helps!

Quote:

Originally Posted by thinking allowed

...Do you have any tips on the ‚rattling snare' that happens during live recording in a small room? I figured out that low frequencies from the drums, guitar, bass all cause the snare's spring to rattle like crazy on certain notes. I swear that better drummers (with more expensive kits) don't have this problem as much. Am I crazy?

Good advice from Smurf about tuning. Also, decouple everything from the floor, especially bass amps. Sometimes the worst resonances come through the floor instead of through the air.

If the problem cannot be controlled, a neat alternative to sample-replacement is as follows: Record the drum performance with the snare wires slack, to prevent buzzing. Then, take a guitar amp and lay it flat on its back. Send a gated snare track out to the guitar amp input so that the guitar amp is playing a short "pop" on each snare hit and nothing else. Now take the snare drum and place it face-down on top of the amp speaker. Turn up the volume and it's like a ghost is playing the snare. Set up a mic and you record your drummer's actual snare sound.

Reply from Smurf

I have done this also, and you can create some great sounds by changing the EQ on the amp (all high, all mid, all lows).

[begin page 12]

> Quote:
>
> Originally Posted by nfpotter
>
> Invitation:
>
> I beg to differ on a point. Computer audio is not RAM-intensive. It is CPU intensive, much as graphics applications are. The only time you need big RAM is if you're loading big sample sets, etc.
>
> That said, of course more memory CANNOT be bad by definition...

? I don't actually disagree, but I'm not sure what you're differing with?

> Quote:
>
> Yep on mastering (cakewalk forum):
>
> http://forum.cakewalk.com/tm.asp?m=475013
>
> (Yep starts at the 3rd message, I think.)

Wow, blast from the past by an intrepid googler! I might actually update some parts of that advice later, since there are some things I kind of over-stated. But that sure saved me a lot of typing!

> Quote:
>
> Originally Posted by sly
>
> ...You strongly stress the importance of comparing things at the same listening level. You write...I am a bit confused by this, also in accordance to question 1.). I thought that VU-meter/RMS-meter (are they the same?) was supposed to represent the nature of human hearing?...

Yes, VU and RMS meters are generally supposed to reflect something like the "average loudness" as perceived by human beings, and are much better than peak meters for that purpose. Although one cool thing about being a human being is that you don't actually need a meter to know how loud something sounds to a human being. YOU are the calibration reference. That's how they came up with the VU standard – by asking people when something sounds louder.

> Quote:
>
> ...What I am really aiming at is this: Isn't there any way of automating this way of listening with a "steady gain plugin" to put after e.g. an equalizer. Something that would hold the volume on the same level even though you boosted something 10 dB. If not, what is an recommended procedure in for instance an equalizing situation? Simply constantly riding the speakers volume control?

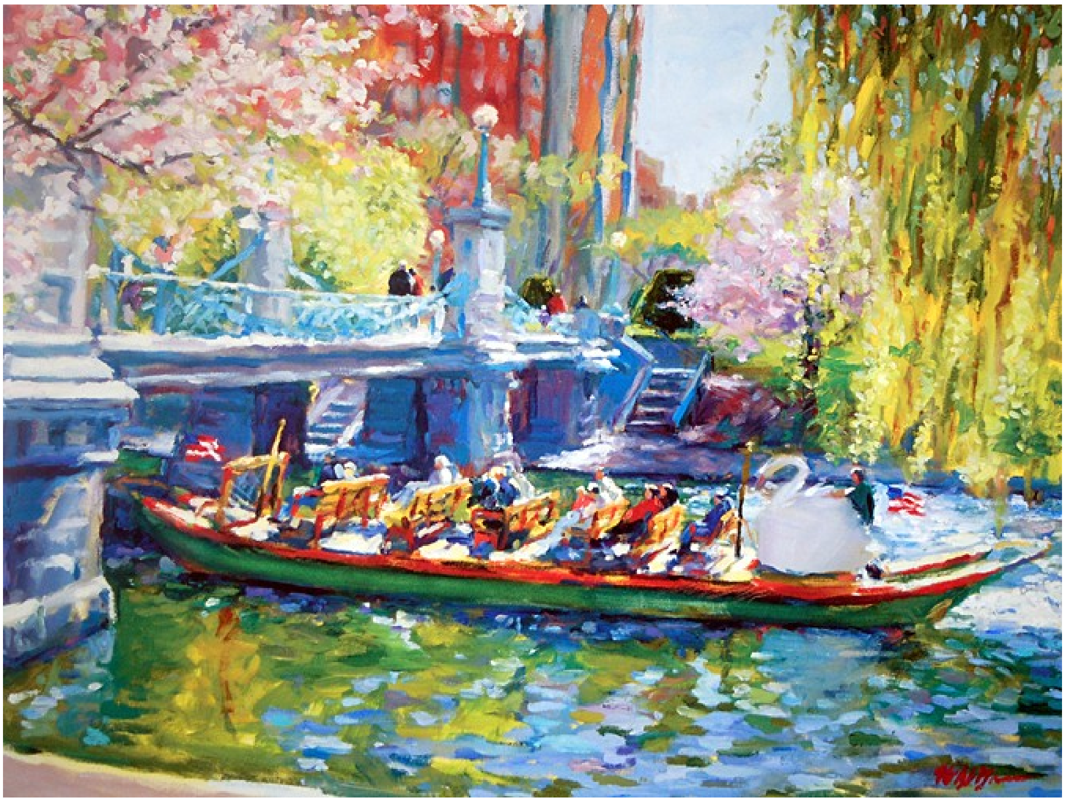
Ahh, well, if you're asking what I think you're asking, like an equalizer that automatically adjusts output gain to compensate for the increased/decreased signal level caused by the eq, I don't know of any such thing. But you don't have to ride the speaker level, just turn down the track gain. And don't go too overboard with the level-matching, just be aware of it at all times, and use it to double-check yourself. It will start to become second nature.

Normalizing has nothing to do with any of this, since that is a process that is entirely about peak level.

And I'm afraid I have nothing to say about "Mixing with your mind." But I do have some things to say about mixing, and particularly about mixing with your ears, but it might be a few days before I get to them.

Mixing

Take a look at this picture:



Whatever you think of the painting or the quality of the art, notice the colors. All of the colors are vivid and bright, but there is still a lot of contrast – it's not a "flat" or "primary" looking picture. Even though most of the colors ARE fairly primary. There is depth in the shadows but not murk. The water gives a sense of light reflected off of the swampy pond, but the painting itself doesn't LOOK swampy, as it would if the water were painted army green with some reflected highlights. We can tell the bridge is made of concrete or stone even though it's painted blue. We do not perceive the shadows as glowing purple and turquoise. The bright red and orange rectangle in the background still somehow looks like an old brick or brownstone building. There is a sense of dazzling late-spring brightness and sharp contrast even though almost none of the colors used are very dark.

Mixing a record, especially a modern-sounding one, is a game of contrast and difference. In grade school we learned that mixing different parts of the color spectrum (such as red and yellow) makes new colors (such as orange). We also learned that mixing ALL the parts of the color spectrum makes a sort of swampy murk.

When you take a bunch of musical instrument sounds, all of which have some sound across the entire spectrum, and you mix them all together, the result is often a similar kind of swampy murk. Flashes of pure white in a painting can almost hurt your eyes with their brightness, but a blank sheet of pure white paper just looks like a sheet of paper. In the same painting, a couple of dark patches can convey a sense of underwater depth, but a black piece of paper just looks flat. Flatter than even white paper does.

Sky above an open ocean is a staggeringly awesome thing to look at, but makes for the most boring photograph in the world. There is no sense of scale. You have to get some close-up grains of sand along the edge of the beach, or some big ripples in front of a lonely buoy or something. Everything big is the same as everything small.

Mixing is a game of contrast. If you want something to sound loud and explosive, it needs to be contrasted with a bed of quieter stuff, or even better, with a preceding silence. There is an effect used in some movie soundtracks where they pull all the sound down to silence immediately before a big explosion or something, and the effect is deafening and dramatic.

The ear is attracted to motion and difference. Steady-state sounds (such as strumming chords on a distorted guitar) fade into background noise. People are immediately drawn to whatever sound is in motion. This creates some potential for problems if there is a too-busy arrangement. That should have been dealt with at the arrangement and tracking stage, but the mix engineer has a great tool at her disposal: just cut out the stuff that distracts from the vocal. There is nothing at all wrong with a verse that just consists of bass, drums, and vocal. Or even less. First rule of mixing: Just because it was recorded doesn't mean it has to be in the mix.

In the above painting, every part of every inch is EQ'd into a specific and focused frequency (to mix metaphors). As a result, none of it is muddy or washed-out. It's all vivid, dynamic, and what we might call "punchy". Every part is focused.

If you have a guitar track playing open chords in the low registers, and a bass part mostly playing root notes, then there is only one octave of content (about equivalent to an EQ with a Q setting of 1.2) where the bass is playing content that is not masked by the guitar. And because harmonics extend the range of an instrument upwards, the guitar is masked through its entire range. If you also have a piano playing a left-hand figure in the bass range and right-hand chords in the midrange, then everything is masking everything. And we haven't even tried to fit the vocal in yet. Plus, if we figure that we have push the overall level down about 3~6dB every time we add a new instrument, then these instruments are all getting quieter and quieter. An instrument part that sounds perfect is going to sound a lot different when it's 12dB quieter and masked by four other instruments.

So how do we fit all these clowns into a phone booth so that you can still see them all, and nut just have a big mush of random body parts showing from the outside? The old answer was with clever arrangements – instruments all had specific roles and assigned ranges, sparser parts to accent and highlight the vocal melody, and the band was structured around the singer's range. The modern answer is with a lot of cosmetic amputations and plastic surgery.

More later.

Once you are done with the high-pass filter, try using a shelving filter to extend the low cut a little further. Just keep going up until it sounds bad, with maybe a 6 or 12dB cut. Start aggressive. Then try the same on the high end, cutting out all the hiss and air from tracks that don't need it.

> Quote:
>
> Originally Posted by dudie
>
> Which 5 watt tube amp (if you don't mind me asking)?

I don't mind your asking, but I'm sorry to report that it is basically homemade from a custom combo chassis with a spring reverb from a mid-80s Peavy combo hacked in. The speaker is a celestion green, FWIW. But that doesn't mean that it's any better than anything you could buy off-the-shelf, or even from a plugin. It's just the amp I happened to have hacked together over the years.

If anything, it's meant to sound like a mid-80s Marshall half-stack, except without the phase discrepancies from multiple speakers. To each her own.

> Quote:
>
> Originally Posted by Moose
>
> ...To a guitarist, be it because he's looking for mojo, emulating a hero, or heard a recoded tone that he just fell in love with, crap like the brand of opamp in the overdrive or clipping diode in his dirt box could be the difference between tonal night and day.
>
> Now, stop thinking like a guitarist. Start thinking like a recordist...His sound has to inspire him to play his absolute best...

This was a great post all around that hits some perfect notes about the differences between creating music and recording music. The better you can learn to separate the “behind the glass” process, the more productive you will be. If you're recording yourself, then you have to learn to trust yourself on both sides of the glass.

Unless you have an infinite time horizon for completion, at some point you have to close the door on each step, cross it off the list, and call it “done.” And there is no sense at all to trying to second-guess future steps until you get there.

There are a lot of musicians for whom home recording has become just a sort of ongoing process, like gardening or something, where after work they like to sit in front of the computer and mess around with sounds and knobs and so on, without any particular objective or clear to-do list. Which is perfectly fine. It's probably a much better way to pass the time and to keep your brain engaged than watching television. But I daresay it is unlikely to yield a finished record at any predictable future date.

Businesses have brainstorming and R&D budgets and time planned for open-ended sandbox creativity, and then they have a separate production process. Probably every album ever released has something wrong with it that the artist wanted more time for – a bad verse, a brilliant synth intro that never got finished, a solo that has some mistakes or that had to be copied over with outtakes because something was out-of-tune, a mix that wasn't quite there, a bassline they wish they had re-written to go better with the strings, whatever. But hopefully the good outweighs the bad and people ultimately get their $18 worth on release day.

Back to mixing, fitting clowns into a phone booth, and creating contrast...

- If you have a really cool part that is getting obscured in the mix, instead of going through convulsions trying to make it audible, feature it in an early and/or late breakdown. e.g. if you have a really cool left-hand piano figure (top of the list for coolest and most likely to get obscured instruments), start off the first verse with just the left hand of the piano and vocals, or maybe do the same right after the bridge or something. Then the audience will still hear it even once it's buried. Plus, the effect of starting with something cool (piano bassline) and then overwhelming it with something even cooler (whole band) creates a tidal wave of awesomeness.
- If you have a guitar riff or some such that is obscuring or fighting the vocal, pull back or cut out the guitar part when the vocal is singing (duh). You can use the vocal to duck the guitar with a compressor side-chain (maybe even more effective if you use bandwidth-limited compression to only duck the upper presence range), or you can just turn down or mute the guitar part during singing.
- Before auto-tuning anything that sounds pitchy, try cutting the lower mids with deep, broad eq. Pitch perception is a weird thing, and for a whole lot of reasons, "perfectly in tune" does not always sound perfectly in tune. Pitch correction does not always solve the most frustrating kinds of pitch problems, especially with harmonically complex stuff like the human voice.
- Get in the habit of "marking out" important eq ranges early in the mixing process. You don't necessarily need to DO anything with them, just kind of drag around shelving and bandpass filters across individual instruments with the whole mix playing. See where things start to jump out or fade back.
- Following the above, know that the best clarity and impact will come from focus and minimalism in key areas. If the hi-hats and guitars both have the

same "sizzle" range, then you probably need to decide which of them to pull back at that frequency. Otherwise your beautiful "sizzle" turns to fizzy hash. You might decide to have the guitar riff sizzle the first time through, to showcase it, and then to pull back the highs for the rest of the song and let the hi-hats take over for a more overall "hi-fi" sound. Or maybe you punch the guitars back up for the pre-chorus or something. Whatever. Same goes for all the different frequency ranges. The more separation and focus you can get the better your overall clarity and headroom will be. Five instruments all playing on the same frequency is like scribbling over the same spot with every crayon – not a vibrant explosion of color, just swampy murk.

- Speaking of guitar riffs and other kinds of overwhelming musical awesomeness... less is more when comes to really heavy-handed and iconic musical figures. By way of example, I direct your attention to "Smoke on the water" by Deep Purple. The guitar riff to end all guitar riffs is really nothing more than an intro and a measure used to transition out of the chorus. If you want to make your audience go crazy (in good ways), punctuate your musical ideas with really cool fills and transitional devices. If you instead want drive your audience insane in all the wrong ways, loop the riff from Smoke On The Water over and over again for five minutes. I don't think there are more than a dozen repetitions of the signature riff in the album version of that song. You might be afraid your audience won't notice a figure that is only played twelve times throughout the song. Trust me, nobody who's heard it has failed to notice the smoke on the water riff.
- With the above in mind, it may be necessary to make some very painful cuts in the mix.

I will bet dollars to doughnuts that the first version of "Smoke on the Water" had a LOT more repetitions of that riff until a producer or someone with a sense of proportion came in and pointed out that too much gravy spoils the meat. And this is not just for guitar riffs.

We've all heard neo-soul singers go so over the top with miasma that there is no longer any sensible melody. The lead is basically accompanying a virtual melody that nobody else knows, and there is nothing to remember, nothing to sing along with, and no longer any real "song" there. Horn parts that were written before the lead vocal was recorded are another common culprit. And slap-style bass lines can often make mixing a waking nightmare.

Musicians who have played a song hundreds and thousands of times often lose sight of what it sounds like to someone who only ever hears it for three minutes at a time a few times a year. There is a tendency to err on the side of "too much" rather than too little. Like a cook making a sauce all day, every time they taste it, their tongue gets a little number to the flavor. So they keep adding more salt and spices until they finally deliver a dish to the table that is overpowering and unbalanced. This is a good reason to keep tracking and mixing separate, and to mix with fresh ears in short bursts.

A lot of modern rock and hip-hop music has a very monotonic quality throughout. The vocal line may not have much melodic variation, the guitars basically play the same open chords throughout, the bass is playing pedal tones, the backing beds stay in the same key, and so on. This is not necessarily a bad thing, in spite of the protestations of music nerds who don't "get" modern pop. The creative essence of a lot of popular (i.e. Non-classical) music is not melody and harmony and all that, but the expressiveness and performance gestures and stuff that an audiologist might call "formant." And this stuff might vary quite a bit in very compelling and yes, artistic, ways.

But what to do when your expressive and compelling performance gestures and aching or slamming vocals are getting weighed down by a dull, droning lower midrange that just sounds like a soup of gray monotony? The answer to this one is so easy and obvious that most people never even think of it as such, even if they do it without knowing why: CUT THE FUNDAMENTALS. Just cut out the actual "note" and leave the "formant", harmonics, and performance gestures and hear your tracks spring to vibrant life. A lot of people are afraid to cut lows for fear that they will lose depth and weight, but you don't need to paint swampy green to show a pond.

Probably every third record on the radio features a strumming acoustic guitar that has been high-passed right up into the upper midrange, and instead of sounding like a dull throbbing it sounds sparkly, lively, and dynamic. The ear is drawn to motion and tunes out monotony. So to make things "pop", turn down the steady-state portions. This works ESPECIALLY well with "power" instruments and vocals that have a lot of lower-midrange chest, and even with bass guitar. Skeptics: try it and thank me later.

Dealing with high frequencies

If you have been trying to get clarity, air and spaciousness by boosting highs on everything, you're doing it wrong. Every time you add another track, you add more hiss, fizz, and "veil" to the sound. And boosting the highs just adds more veil and makes the high end more obscure and muffled. The key to top-end clarity is CUTTING the highs, just as the key to low-end punch and depth is cleaning up all the mud and murk. Not many instruments need ANY content above 11k. And few even need much above 5k. A lot of them don't need much above 1 or 2k. Once you have opened your eyes to the world of cutting lows, the next step is to get just as ruthless about cutting highs on every track until it sounds bad, and then see who is the "last man standing." The thing about deciding which highs to leave intact is that it depends a LOT on the specifics of the sounds involved. In my home studio projects, it's frequently the drums, solely because I usually trigger through BFD which has much cleaner highs than what I get recording in the spare bedroom. If I'm recording a real drum kit, then it's probably the opposite – my home drum recordings might have more hiss and background noise than for instance the vocals or acoustic guitar or something else that was recorded close, with fewer mics. In a "real" studio with better isolation and better signal paths, it could be anything. It also depends a lot on the actual "content" up in the highs. A whispery alto female is a much likelier candidate for having meaningful musical information above 3k than a falsetto metal screamer singing in the same range.

This brings us to a really important point, which is the definition of the "highs." In my opinion, the useful "highs" are roughly the two-and-a-half octaves from about 2k to 12kHz. Remember, high C is something like 8k, and that sounds REALLY high when you hear it. (An alto sax tops out at like 800Hz, about three octaves lower). Anything much higher than 12k is more perceived than heard, and usually not in good ways. This range should be thought of as psycho-acoustical "super-highs" that devour headroom and contribute little to most music except as an occasional "special effect." This is really deceptive to novices who look at an equalizer and see almost a full octave that is practically inaudible at the top end. It's one thing to push up the top end of the "smiley curve" on a muscle car stereo, but anybody with good monitors who fancies themselves a mix engineer and who shelves up the highs on more than one instrument should probably be beaten with a ball-peen hammer.

Hiss is really an ugly enemy of audio, especially when it happens in ranges that are too high to clearly hear but that are low enough to still veil and mask the sound. In courtrooms, when the lawyers approach the bench to talk to the judge, the jury is often treated to a spray of white noise from overhead speakers to mask sound. Mobsters talking "business" might turn on a faucet or a loud air-conditioner to mask what they are saying.

This is what hiss does – it emphatically does NOT make anything sound "airy" or "clear", it instead makes things sound veiled, muffled, and obscure, even if it's too high to actually "hear." If you go to an old person who can't hear anything above 13k, and play some white noise filtered to 14k and above, they won't be able to understand anything people are saying. Similar technology underlies real-world "cones of silence" used by spy agencies and the like. The old RIAA AES mechanical rule for mastering had everything cut above 12kHz, and a lot of great-sounding records were made that way right up until the mid-1990s.

The point of all the above is to reinforce the reality that boosting high-shelf filters on everything does not INCREASE “clarity” and “air” but instead muffles and masks it. Especially with content that contains an acoustically significant amount of hiss, noise, or fizz. That means most home recordings and ANY electric guitar, bass, or unbalanced instrument such as analog synths and keyboards. The fact is, you could do a lot worse than to simply cut everything above 12kHz or even lower, across the board.

You might be thinking “won't that sound muffled?” Isn't the “air” part of the sound? Well the good news is that sound won't reproduce in a vacuum, so your audience will by definition be listening in a space that contains air. And real-world speakers and reflections produce harmonics. So no, you don't need to bring your own air to the party. But if you're skeptical, instead of trying to think it through, just try it and see. If you actually LISTEN to your favorite records and the instrumentation, you might be surprised at how much the individual instrument sounds are the exact midrange opposite of the intuitive beginner “smiley-curve” eq.

> Quote:
>
> Originally Posted by PAPT
>
> I have a problem understanding what goes on in the highs.
>
> On my own mixes, if I try boosting at 12k or higher I notice that there is actually not much of anything there but hiss.
>
> If I look on a spectrum analyzer it shows a fairly steep roll-off above 12k.
>
> However, when I put commercial music through the spectrum analyzer there is content up to around 20k.
>
> Admittedly, the high end on much commercial music just sounds like noise added as an overlay to the actual music.
>
> Maybe that is what it really is?

Content below around 45Hz or above about 12kHz falls under the "special effect" category. If you have a record where a spectral analyzer shows content outside those ranges, just drag a copy into REAPER and filter it for those frequencies to see what they sound like. Alternately, try cutting those frequencies and see whether/if/how bad the sound suffers (my guess is very little if not zero on most material, and it may even improve).

My point was not that the ultra-high end is bad, just that it's not ipso-facto GOOD, especially if it mostly or entirely consists of hiss and electrical noise, which is likely if there is not any significant musical content up there. Don't try to think it through, just try it out. It takes more time to type this stuff than it takes to drag around a low-pass filter in Reaper.

> Quote:
>
> Originally Posted by EVAD
>
> ...Do you have any suggestions about getting a good vocal mix as well? As you mentioned, vocals have complex mixes of tones and such and each voice is unique in its own way.
>
> Thanks in advance

If you have (as is common) a good instrumental mix and are having a hard time "fitting in" the vocals without either burying them or overwhelming everything else, a generic starting point is to aggressively cut the lower mids of the vocals. I say that without knowing anything at all about the song, mix, performance or recording, so it's got maybe a 55% chance of being applicable to your specific record. But it's a bigger chance than any of the other thousand possibilities.

Vocals are no different from any other instrument, except that clarity and articulation are paramount, and that there is a massive potential embarrassment factor if they don't sound great. The bass player might frown a little if her part is not heard clearly and the way she envisioned it, but she's never going to hate the record the way that a singer will if the vocals are not the way they imagine themselves sounding.

I posted some stuff earlier in this thread about vocals specifically. Happy to talk more about specific questions.

I think this might be a good time to re-visit some of the basics in light of some of the ground covered since the start of the thread. Earlier I mentioned that there is a big “2 steps forward 1 step back” aspect to learning audio engineering, and that everything affects everything else. As you move from recording to mixing, it is common to change the approaches you might use while recording.

Beginners often start out trying to get each track to sound as big and hype and powerful as possible, so that each track sounds as close to a satisfying, full-spectrum solo recording as possible. And then they find that the process of mixing is largely a process of removing a lot of that size and hype and power from individual tracks. Which is fine, there's nothing at all wrong with that. Sometimes it's actually a better approach.

But as you start to make more successful mixes, and as the whole recording process begins to inform your listening to other commercial recordings, it's not unusual to find yourself tracking in different, more focused ways. Maybe you're selecting different mics, and backing them off the source a little, knowing that capturing gobs of massive proximity effect is only going to mean applying gobs of massive low-cut further down the line. Maybe you find yourself more confident about applying some basic eq and compression right at the tracking stage, now that you have a better sense of what the track is going to need to settle into the mix. You might find yourself re-visiting some of your old standby synthesizer patches or guitar sounds after having had a hard time wrestling them into a mix. You might start to re-think the kinds of drum sounds you're trying to capture as you get a better sense of what is likely to complement particular kinds of songs, mixes, and tempos.

Everyone is different, and everyone takes a different approach. And this is yet another reason why "presets" and "recipes" are of limited usefulness. If you and I record with the exact same bass guitar, but mine is set with heavier strings and with the pickups half a centimeter closer to the strings, then I might be tracking a much hotter, flatter, mushier sound that needs a whole different approach than yours, which might be thinner, clearer, and have jumpier dynamics. If we both use the same mic to record the same singer, but you start with the mic 12" away, off-axis, and I start with the mic 3" away, on-axis, then there are very likely going to be massive differences in frequency profile. It will sound like we recorded with totally different equipment. And which is "better" is a totally subjective call.

[begin page 13]

Sample rate

> Quote:
>
> Originally Posted by DerMetzgermeister
>
> Hi yep.
>
> I have a question about sample rate. Like everyone else I have read tons of threads and articles about the subject, yet it's not 100% clear to me.
>
> I'm quite sure that my recording gear is not capable of capturing sounds above the range of a 44.1 Khz recording.
>
> So, there is no valid reason then to record at a higher sample rate? And, if there is an advantage to recording at 88.2 Khz or higher, what it is? Thanks in advance.
>
> Edit: Sorry if the theme have been addressed previously in this thread.

This is a big, sprawling, and somewhat controversial topic that might be better-suited for another thread, but here's a short version: First of all, there is not necessarily any need to "understand" all the technical aspects of digital audio in order to make good recordings. In fact, a little bit of knowledge can be a dangerous thing, an awful lot of people would be making better-sounding recordings if they stopped to trying to "understand" or "figure out" the technical stuff and simply trusted the equipment designers to do their job.

Most serious people generally agree that human beings cannot hear above 20kHz or so. There are some who think that people may be able to subtly "perceive" higher frequencies, but it's certainly not critical for good-sounding recordings. And a 44.1k sample rate WILL accurately reproduce content up to 20k, so it seems like that should be the end of the story. But it's not, necessarily. Because the sounds you are capturing actually have an infinite harmonic sequence that goes above 20k, which needs to be cut off at 22.5k before conversion to digital. The frequency at exactly half the the sample rate is the highest frequency that can be perfectly reproduced by the system, and is called the "Nyquist frequency."

And that filter that cuts off the super-highs, by definition, has to be an analog filter, and it has to be a fairly steep hard cutoff filter. Which is the most artifact-inducing kind of filter, and causes "ripple" effects both above and below the cutoff frequency. EQ is basically just a delay or a series of delays, and very high-frequency eq is basically adding excruciatingly short delays to cancel out certain frequencies. So a poorly-designed or poorly-made filter (or possibly even a good one) will cause artifacts in the audible frequency range, especially if the filter's cutoff frequency is close to to the range of normal human hearing.

How big a deal are these artifacts, potentially? I'll get to that in a sec. For now, we'll just say that a "perfect" AD converter with "perfect" filters WOULD BE perfectly adequate to capture audio for human consumption at 44.1, setting aside exotic theories of supernatural hearing. (and "setting aside" exotic theories can be a more dangerous business than it sounds, when it comes to subjective sensory experiences).

HOWEVER, the "capture" is not usually the only thing that happens to digital audio. It is also likely that you'll be using plugins and processing within the digital realm, and this brings up another set of potential concerns. Processors such as compressors and distortion and analog-emulating tubeifiers and tape sims and so on are all likely to add harmonic distortion, either subtle or pronounced. And harmonic distortion extends the frequency content. What happens when the compressor creates harmonics higher than the Nyquist frequency? Aliasing distortion, that's what.

And aliasing is a pretty ugly and unpleasantly "digital" distortion. When you get frequencies embedded in a digital system that are above the Nyquist frequency, they come out through the playback converters as sort of randomized, fluctuating "subharmonics" of the too-high frequencies, modulating the audible frequency range in unnatural ways. These kinds of digital nasties are responsible for a lot of the "digital synths/guitar effects sound like crap" opinion out there.

Now, clever programmers can and should and usually do come up with ways to handle these problems, which are not necessarily terribly difficult or exotic (internal anti-aliasing filters or oversampling are a pretty good start). But those in turn add another layer of complexity to the audio processing, and a lot of obsessive purist types (such as mastering engineers) prefer to stick with the simplest processors and just start out with high-sample-rate recordings so that any aliasing artifacts are all pushed up into the inaudible range (there is no need to worry about giving the mastering engineer a low-sample-rate recording, they'll upsample it if they want). To each her own.

You, the home recordist/musician, should probably NOT try to "figure out" this stuff, nor to "think through" what kind of approach to aliasing is best. YOU should probably just leave that to plugin designers, and then use the effects that sound good, and don't use the ones that don't. Or, if you have the space and processing power, you could just record at higher sample rates. But high-sample rate mixing eats up a LOT of processing power. Every plugin uses 4x as much CPU on 192k audio as it does on 48k audio, so the tradeoff is not insignificant. And get ready for your head to explode, because higher sample rates can actually sound WORSE than lower sample rates in some cases.

Jitter

The single biggest problem with AD conversion is jitter. "Jitter" is what happens when the samples are taken at non-perfect intervals. The playback converter is and must be counting on perfectly-spaced samples, and it will reconstruct a waveform based on the expectation of same. If you sample a pure sine wave at non-perfect intervals, the playback converter is going to "re-space" those samples perfectly (or as perfectly as it can), and the sine wave will come out all crooked and stretched and squashed in weird ways, which sounds like lots of ugly and random harmonic distortion. "Digititis" in short.

The thing of it is, the higher the sample rate you record at, and the faster those samples have to be taken, the more potential instability there is. So especially with cheaper converters, higher sample rates can actually come out with worse jitter than lower sample-rate recordings. There is no free lunch. And forget what you may have heard, you CANNOT EVER improve an AD converter with a fancy external word clock. Always record with your audio interface set to internal clock. If you don't know what this means, ignore it, it almost certainly doesn't apply to you.

Remember all the way back in the beginning of this thread when I said that CONFIDENCE in you gear was more important than having GREAT gear? And that test, of recording a great-sounding CD or record through your soundcard's inputs to see if it still sounded great? And how that tells you the results that you can and should expect of yourself and your rig, sound-quality-wise?

Last but not least is the consideration of target medium. Sample rate conversion is very, very easy to do perfectly at exact multiples of the target sample rate. If you want to convert 88.2kHz to 44.1k, all you need to do is to throw away every other sample. But if you want to convert 96k to 44.1, then the SRC has to sort of interpolate or "figure out" where the sample points would have been on the original analog recording. And this creates the potential for aliasing (in fact it actually causes at least some aliasing in all real-world sample-rate converters, AFAIK, but the best ones are extremely good at it.) So to keep things simple for best audio integrity, if you're recording for CD, you should probably stick to 44.1, 88.2, or 176.4.

I recommend against trying to "think through" this stuff. There is way too much to know and way too much to keep track of and way too much room for "paralysis by analysis", as my father would put it. More importantly, it is way too easy to lose focus on LISTENING when you start THINKING. There is not and doesn't HAVE to be a "perfect" sample rate, and even if there was, you wouldn't have to use it to get great-sounding recordings.

Record at a sample rate that makes sense practically and sonically for what you're doing. Try the "record a CD" test at different sample rates if you like, and see if you can hear a difference.

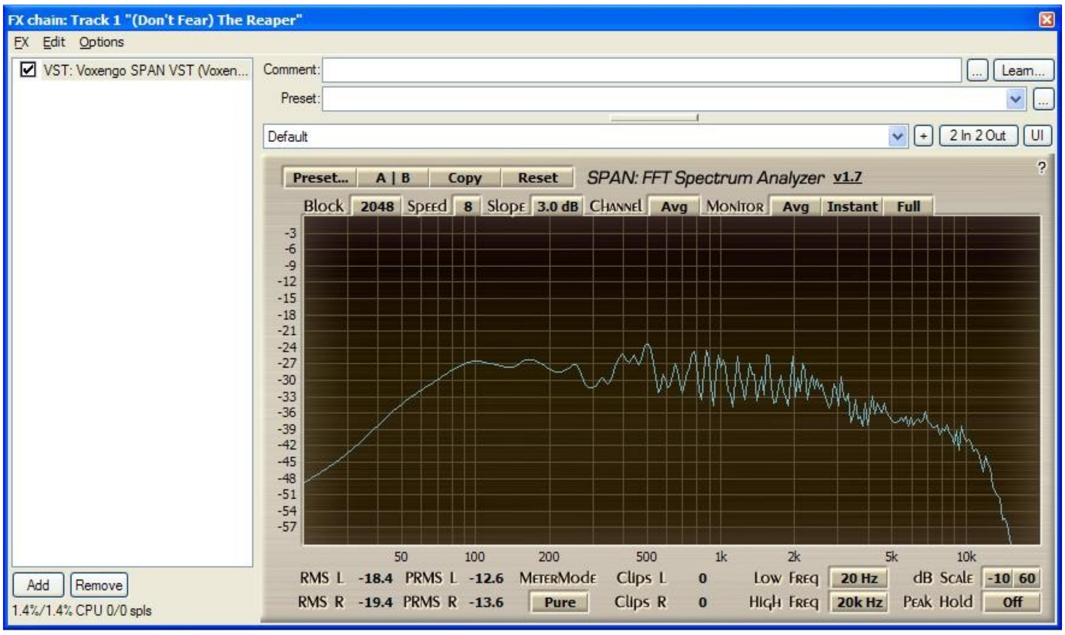
PS – None of the above has anything at all to do with bit depth, which is much simpler: always record at 24 bit. The CPU hit is negligible on a modern DAW and the benefits are much more clear-cut than with sample rate.

Ultra-high frequencies

> Quote:
>
> Originally Posted by GrantsV
>
> ...Above 12k contains vital frequencies that contribute to the "sheen and polish" of a track.
>
> Every commercial track has strong clear highs and upper harmonic content....

As I was reading this, the radio happened to be playing "Don't Fear the Reaper" by Blue Oyster Cult. Say what you will about the song, but in my opinion it's a very good recording, with rich, airy highs and a great sense of spaciousness, as well as gobs of "sheen and polish."

So I went and dug up the CD, and ripped it as a WAV file, and loaded it into Voxengo SPAN for a frequency analysis. Let's take a look:



What do you know? I see a sharp cutoff starting at about... 12k! And essentially zero content above 15k! (incidentally there is also a pretty pronounced low-end rolloff starting at about 100 cycles, which is pretty high.)

Now, maybe there are some people on this message board who have higher standards and would not deign to release a recording that sounds like "Don't Fear the Reaper", but my personal opinion is that this an excellent example of a great studio track. Again, not to say that extreme highs are BAD, nor even that they're not GOOD, just that they are not mandatory to make a recording sound clear, airy, polished, professional, and hi-fi. I think this song is all of the above.

I don't want to argue the point, and I certainly don't want to "bully" the thread, so this will be my last post on the topic unless a new question comes up. The fact is that "Don't Fear the Reaper" DOES have a pretty pronounced high-end rolloff at 12k. And that's not because Columbia Records did not have access to high-quality gear, and it's certainly not because they were going for a "lo-fi" sound. That there is some tape hiss at -80 or so might or might not be significant, and if anyone thinks that the production on the track sounds "lo fi" or "old fashioned", well... to each his own. I picked the track because it struck me as a production with lots of polish and sizzle and rich airy highs. Maybe some prefer something else...

But you'll find the same results with, for instance, Miles Davis' "Kind of Blue", the Beatles' entire catalog, and basically anything released before the mid-90s, and a great deal of the stuff released since. The old RIAA mechanical rule stipulates a

30-degree shelf cut from 12k up for vinyl release (and a similar cut at 47k), and that was just applied as a matter of course before the record even went to duplication. Which is why I knew I could pick any song released pre-CD to illustrate the point (yes, I already had the answers to the test and therefore cheated).

A great many big-name producers and engineers still do the same as a matter of course, and there are still records released on vinyl, although that might sound too “lo fi” for the iPod age... to each their own. Don't take my word for it, try it and see if it sounds better. Moreover, and with specificity to GrantV's points, is it even possible to make a rock record with a noise floor below -85? I mean, is a Les Paul plugged into an overdriven Marshall amp even capable of such a low noise floor? (mine certainly isn't, even with a noise gate pedal, but maybe that's just me) How about a Hammond B3 plugged into a real Leslie speaker cabinet? Unless I'm missing something, these are not instruments that can HELP but produce hiss and fizz above 15k, regardless of how you record them.

And even if you have excruciatingly quiet instruments, you're certainly not going to get such a low noise floor using gear such as, for example, an AKG C12, or a U47 plugged into something like a Neve or Telefunken preamp... are those “low quality”? And if you start resorting to gates and expanders to clean up the recording, then what's going to be left above 12k on those kinds of instruments anyway? Just pumping hiss, basically. Moreover, if we assume listeners are playing back at an average 83dB SPL, then what are they going to hear at -85? If they play back louder, then is their short-term hearing really going to be sensitive enough to pick up that stuff, even if they are listening on ADAM S7s?

I know that there are a lot of modern, all-digital records that have significant content above 12k. Some of that is because the engineers don't know what they're doing, and some of it is because of the genuinely extended frequency range available with digital. But there are also a lot of big, modern-sounding, very airy commercial releases that still follow the old RIAA curve. Even in spite of Great River preamps and Lavry converters and access to noise-free virtual instruments and all the rest of it.

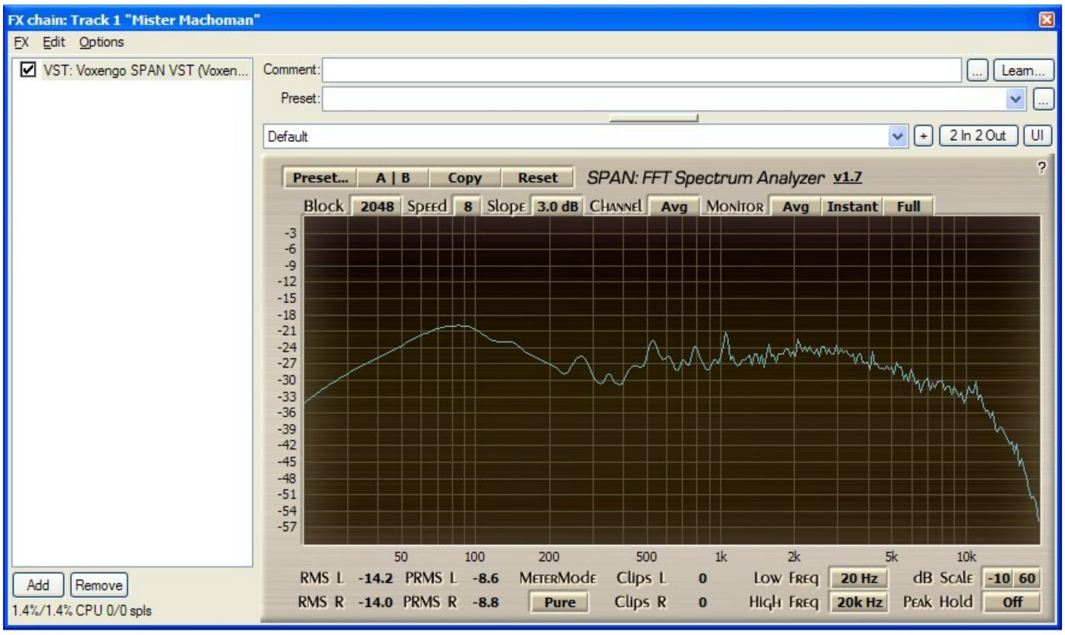
I do absolutely encourage anyone and everyone to think and especially to LISTEN critically, and second-guess everything, including myself. I am very happy that GrantV brought up what a lot of people think about the ultra-highs, and I don't actually even disagree with anything he's saying, only with some of the implicit conclusions that people tend to draw, e.g. that if you have good enough gear, the extreme highs always contain important content, or conversely, that if rolling off the extreme highs improves the sound, then something is “wrong” with your equipment.

There is a lot of internet feuding and such about what "should be" important, or what "ought to" sound better in a theoretical or academic sense, and some of it has some merit. But in the studio, trying to make a killer recording that has clarity, impact, drama, and sonic excitement, a lot of what "should" be the "correct" approach falls by the wayside. Especially when we are dealing with overloaded amateur arrangements such as three guitars all playing in the bottom octave behind a baritone singer.

Some of the great consoles feature high eq that is famous for the ability to crank the knob and create screaming hype and punch in the highs (the old Neve modules with the 12k high shelf spring to mind). And this was commonly done during tracking, especially to boost the highs before they got buried under tape hiss, and then it was just as commonly backed way off at mixdown, like a form of DIY noise reduction, and then rolled off at 30 degrees come mastering. But the "character" was still there, and a lot of modern engineers with only a halfway understanding of audio read and hear about those legendary studio practices and want the noisy, hissy console modules so they too can get those "magic highs", not understanding that the process was inseparably related to the sound of 15ips tape, and that just because legendary producer X did one thing at tracking, it doesn't mean she didn't do an opposite thing later at mixdown.

In fact, from the sound of "Don't Fear the Reaper", I will bet long odds that at one stage, it was a very treble-heavy project, either from high boost or low cut. But there has obviously been a steep, across-the-board high-cut applied at some stage.

Last but not least, here are the two of the hypest, most modern-sounding album tracks I could think of off the top of my head (among records I own – something by Christina Aguilera might have been a better pick). The first is Mr Machoman by the Lords of Acid, and the second is New Kicks by Le Tigre. Both feature extremely hot, hype production, with a big mix of samples, real, and electronic instruments. I swear they are the first two songs that came to mind. Either one would soncially drop right into a Hollywood Sci-fi/Action flick. I defy anyone to call either of them "old-fashioned" sounding. And they BOTH show the distinctive, pronounced 30-degree shelf rolloff at about 12k and 50 cycles (although neither drops off as sharply as Don't Fear the Reaper). Here they are:



PS – the real point: notice how every one of these graphs is the exact opposite of the instinctive "smiley" EQ curve that people tend to reach for as a first resort. If you've been cranking the lows and highs, and producing weak, mushy, fizzy, quiet-sounding records compared to these, then it's time to re-think.

> Quote:
>
> Originally Posted by mamm7215
>
> What I find interesting about these two graphs is the pronounced cut at 300-350hz, presumably to tame the low-mids. You see the same but with a narrower Q in Don't Fear the Reaper. A trend?

YES.

See earlier, RE: cutting the fundamental.

Mud range.

> Quote:
>
> Originally Posted by mamm7215
>
> What I find interesting about these two graphs is the pronounced cut at 300-350hz, presumably to tame the low-mids. You see the same but with a narrower Q in Don't Fear the Reaper. A trend?

Yes. That struck me as well, and sent me actually look at some things through Voxengo. It's like a minor smiley in the middle of the broader inverted smiley.

I would think this is one way they're able to get the apparent loudness and fullness up, by skewing the frequencies to the right away from the mid fundamentals, with all of it supported by a carefully shaped low end.

FWIW, some samples.

[IMAGES SEE WEB SITE]

Reverb, delay, and general "thickening" effects

> Quote:
>
> Originally Posted by TedR
>
> Hi Yep
>
> Thank you for continuing to contribute awesomeness and field questions.
>
> (It just so happens that i have one, hehe)
>
> When mixing the simple scenarios I am showing below, what can I do to make them sound a little "larger than life" or "sonically bigger" and more impressive without adding additional instrumentation.
>
> Scenario 1:
>
> [2 Tracks]
>
> Track 1= Acoustic guitar (rhythm)
>
> Track 2= Lead Vox
>
> Scenario 2:
>
> [3 Tracks]
>
> Track 1= Acoustic guitar (rhythm)
>
> Track 2= Acoustic guitar (Lead)
>
> Track 3= Lead Vox
>
> I guess what I am looking for is a way to maximize the impact of a very simple recording. If you (or anyone else) has any insight into this I would surely appreciate it.
>
> Thanks

Ah, well, that leads directly to something I haven't much talked about yet, which is reverb, delay, and general "thickening" effects. Most people who get into recording start from some degree of exposure to either synthesizers, guitar sounds, or both. And these are both areas in which presets and recipes are extremely useful. So beginning recordists are often frustrated by compression and EQ in a mixing sense, where presets are close to useless.

But reverb is a an oasis of useful presets, as long as you're using good reverbs, and as long as you know how to use them. For the near term, I'm going to be using the word "reverb" as a catch-all shorthand to indicate all sorts of ambiance and delay effects, unless I'm specifically contrasting reverb per se with something else, such as short delays, chorus, etc. "reverb" is not something we ever consciously "hear" in the real world, unless you're in a parking garage or something, but it is all around us. As I touched on earlier in this thread, if someone were to lead you blindfolded through your house (or probably even through a stranger's house), you'd be able to tell whether you were in the living room, or the kitchen, or the bathroom, or a bedroom, just from the quality of the silence.

In studio recordings, reverb might be used to simulate this kind of real-world sense of subliminal "space", or it might be used as a dramatic "effect" to increase the size and scale of a thing. Neither approach is right or wrong, but it's a significant distinction. And making everything sound like giant 80's "drums of God" is not necessarily an improvement.

If you can get the reverb to decay along with the tempo and natural decay of the instrument, and if you can get the reverb tonality and frequency shape to be less present and forward than the instrument, you can often dial in some massive reverb and still have it fall sort of "behind" the dry track, so that the dry track doesn't get washed out or lose its immediacy, but simply has an increased sense of size and spaciousness.

OTOH, if the reverb decay extends past the natural decay of the instrument, or if the reverb has its own frequency profile that is distinctly audible as a separate sound from the instrument, then you'll end up with that 80s drum sound where the it's like the audience is listening to a drum kit and also to a room or a reverb box.

Neither approach is right or wrong.

If you ever play video games, you might be familiar with the phenomenon where, just before the last level, after you've had to fight hordes of bad guys with underpowered weapons and limited ammo, you get the "big gun" that relieves you of such concerns. I recommend viewing reverb the same way. It is something to use AFTER you have got everything else knocked into place, never a substitute or easy "wash" to drown out the problems in your mix (this is an all-too-easy temptation).

So the first stages are to get the best balance of thump and body and air and presence and clarity and so on that you can get, using mic placement, mic selection, signal path, eq and dynamics, and to get the cleanest, most dramatic and flattering representation you can, dry, and THEN to bring in the big gun of reverb.

Like makeup, reverb can smooth over a lot of flaws and cover up a lot of imperfections, but it's no substitute for a beautiful face. And in the world of music, where everyone can date the prettiest girl for the same price as every other, being heavily made-up is not going to win any dates. Everyone can buy the best CD ever made (whatever that may be), so being halfway there and covering up with makeup still lands you with nobody asking you to the prom.

So... before we go any further, get right with the concepts and practices of contrast based on frequency and dynamics.

PS – as an aside, nicholas' Reamix book deals specifically with a lot of acoustic-only, bassless, drumless mixing. It's also a great primer on using some of Reaper's advanced features. Some of the approaches are a little different than what I've been talking about in this thread, but more points of view are a good thing.

> Quote:
>
> Originally Posted by Marah Mag
>
> ...it's like a minor smiley in the middle of the broader inverted smiley.
>
> I would think this is one way they're able to get the apparent loudness and fullness up, by skewing the frequencies to the right away from the mid fundamentals, with all of it supported by a carefully shaped low end...

Marah Mag FTW.

Using reverb

“Reverb” specifically is the dense accumulation of reflections that occurs in a space, densely enough to cancel out much of the frequency content of the actual instrument in favor of the frequency of the space. Similar to the way that different guitars or different pianos have their own “sound” that is often more pronounced than the sound of the pure string(s).

But there are a lot of other ways to create “reverb” as a sense of sonic size and space. Short delays are a very obvious example. Delays that are not quite dense enough to disappear into “reverb” per se can still achieve a very similar psychoacoustical effect, often with less loss of clarity and without the specific “localizing” effect of true reverb. We get a similar “embiggening” effect, but without placing the instrument in a specific “place”, and without sublimating the instrument “sound” into a wash of room sound. This is especially well-suited to modern-sounding “artificial” records that tend to favor a dryer sound that adapts to the playback environment, as distinguished from more naturalistic recordings that try to contain their own sense of space.

Tied directly to this approach of “delay as reverb” is the increasingly common practice of double-tracking (or triple-or quadruple-or quintuple-tracking and so on), as is the practice of “layering”, for example tracking a synth part to mimic the vocal or guitar, or vice-versa. Tracking a thing more than once thickens up the tonal qualities and creates depth and texture similar to reverb. A similar effect can be achieved just by slightly modulating or detuning delays (as with a chorus effect). You could also experiment with panning different short delays and eq'ing them differently.

Of you could just load up a reverb box and flip through presets, and then tweak the best ones to taste. Which brings me to a very important point...

A lot of reverb plugins suck.

No, it's not just you. Reverb is the redheaded stepchild of plugins. It gets no love and gets treated like an afterthought, or thought of as something that can be killed by just cramming enough reflections and CPU horsepower in there. They sound splashy and trashy and completely unlike any kind of real-world space that you would ever find yourself in. They might have infinite controls and massive feature lists, but a lot of them still sound like a metal room.

Creating a good reverb takes love and dedication, of the sort that is usually reserved for vintage compressor emulators. The best across-the-board "free" solution AFAIK is to use an impulse reverb like SIR or REAverb and then dig through bajillions of impulses looking for a good one. I have some reverbs that I like but I have not found anything close to a short list of "best" reverb plugins. My own personal go-to favorite is the Sonitus reverb, but even that is like maybe 30% of the time, and it's not free. Suggestions welcome. More later.

> Quote:
>
> Originally Posted by spikemullings
>
> Pipelineaudio's impulses from the resources page sound good to me and there is a great variety of them:
>
> http://stash.reaper.fm/tag/Reverb-Impulses
>
> (I should add that I don't have any expensive hardware or software verbs to compare with. Oh except where I hear them on CDs I guess)

There are LOTS of great impulses for impulse reverbs, and my point was not AT ALL to say that you can't get good reverb "in the box." Only that there are a lot of bad reverb plugins out there, much more so than bad Eqs for instance.

> Quote:
>
> Originally Posted by Marah Mag
>
> Hi yep. I'm hoping you might expand on that, esp the part in bold.
>
> I'm not sure just what I'm looking for you to add to that... but there's something there...
>
> maybe the mechanism or process of the adaption to the pb environment, if that makes any sense... that I wouldn't mind having a better sense of.
>
> Thanks!

Here's the thing... the listening space where you listen to music has its own acoustical properties, even if you listen on headphones. If you're in a bar, or in your living room, or in a shopping mall, or in a car, then a theoretically perfect "dry" reproduction of the band, based on close-miked sources mixed and panned, would play back through the speakers and it would literally sound like the band was playing IN THAT SPACE. There is not necessarily any need to embed ambient information in the tracks, because the listener is going to hear the space that they're in.

One could make an argument that this is actually a more authentic and ideally "pure" approach to take, to simply capture and reproduce the content and leave the environment up to the listener. Another argument might be made that the best, purest, and most authentic way to create a recording is to either capture or create (through mixing and processing) the ideal "third row center" listener experience, and then count on the listener to make sure that they are listening in a good space and on a good system. This is something like the approach taken in movie mixing, where the presumption is that the listener will be hearing calibrated, tuned speaker systems playing back at a reference level of 83dB SPL in a room with high ceilings and soft, dark walls, and so on. In this case, we want to make sure that **everything** the listener is supposed to hear is embedded in the track. And we expect them NOT to hear anything else.

You could make an argument either way. That said, I've never been much of a fan of arguments, personally, and I don't really care too much about the "right" way of doing things, assuming there is such a thing.

It's safe to say that prior to the rise of the cassette tape, records were mostly made to sound "best" in ideal circumstances. I.e. It was the engineer's job to capture that perfect "third row center" experience, and the listener's job to have a good playback system. In more recent times it has become increasingly common to make records that are made with more concessions towards real-world listening, if not making records made for the outright lowest-common-denominator. See loudness race, etc.

Part of this has been a trend towards hyper, hotter, and dryer-sounding records. With this has come an increasing tendency to use delays instead of reverb, and to use ALL ambient effects (delays, reverb, or whatever) less as a way to create a naturalistic sense of spaciousness, and more as a sweetening, embiggening effect. More gated verbs, more ambiance that is tuned to fit "behind" the dry track (e.g. decays timed to end with the note), less sustain, longer predelays, more artificially hard-panned stereo effects, and so on.

At the same time, there is also a sort of resurgence of interest in old school, naturalistic, vintage, and even outright "lo-fi" recording sounds. And then there are also approaches that specifically try to emulate artifacts from days gone by: hard-gated 80's-style drum reverbs, vintage slapback echo and tape delays, and so on. And there are entirely new possibilities opened up by modern technology. What is good and what is bad is entirely a subjective call, and is totally up to you.

Analog summing and EQ

Got this in a PM and I figured I'd answer here, since for every person who asks, there are usually a dozen others wondering:

> Quote:
>
> Dear Yep,
>
> Thanks for producing such an amazing thread over at Reaper forums. An oasis in the desert of internet information. I particularly enjoy the logical and objective structures.
>
> I didn't want to post this question on "the thread" as it might mess up your order structure. I have limited experience OTB, instead I use Lavry conversion good monitoring etc. and all DAW. I was wondering your views on EQ and mix bus.
>
> In your experience have you seen any benefit to mixing on an actual mixing desk such as Soundcraft, Allen Heath etc. in terms of sound quality over using DAW summing, faders and EQ. My curiosity arose since DAW reverbs as so terrible compared to Impulses, I am wondering if EQ plugins suffer similar quality issues compared to real analog mixers?

Lynn Fuston did a big and systematic project on summing busses a while back, to try and figure out whether analog vs digital summing really made a difference. You can google "awesome DAWsum" to check his tests for yourself. The short answer was no. People could generally not tell the difference between different digital summing busses, and while some people could pick out the difference between analog and digital summing, there was no clear consensus that one or the other was "better", and differences were generally pretty small, subtle, and hard to detect, even among a room full of audio engineers listening on forensic reproduction equipment.

That said, "summing" is only one tiny part of record-making, and blind listening tests might not be 100% indicative of the way that real people work in the real world of music-making. Does having the real-world tactile control of genuine zero-latency, true analog knobs and faders help a recordist to make better decisions, compared with mouse-based or stepped digital controls and computer latency? I don't know how anyone could possibly test such a thing in a scientifically conclusive manner.

And as for whether "analog" eq sounds better than digital eq... each example is its own thing. I don't know what the best-sounding eq in the world is, and I certainly don't know whether, for example, the top third are mostly digital or mostly analog. I do know that there are an awful lot of really bad analog Eqs out there on the cheap end of the spectrum, and an awful lot of really good digital ones on the free end of the spectrum. On the expensive end of the spectrum, one should expect great sound as a matter of course. And most of the name processors deliver, whether analog or digital.

Don't know if that helps, but that's what I got.

> PS – anybody should feel free to post anything they like. This is not is not my forum and was never meant to be a "yep tells people how to record in sequential order" thread!

[begin page 14]

> Quote:
>
> Originally Posted by stupeT
>
> For my understanding I will summarize what I learned for bass and treble for the poor man's one room studio (and please correct me if I'm wrong):
>
> Roll of bass and treble on most tracks to get rid of rumble and hiss as much as possible for that particular track. Cut more than the unexperienced would think. That process is called cleaning and should be done first in mixing.
>
> Keep the bass on just the necessary tracks, typically BassDrum and Bass. Shape these tracks differently with EQ to seperate them spatially and achieve a clean, not boomy fundament.
>
> Keep the air (above 10 or 12k) on tracks only which can contribute hiss-free and non harsh treble. That could be hi-quality samples of Cymbals or HiHat, Triangle, Chimes and so on. If there is NO such airy-track it is better to accept that fact. Excellent mixes do exist without too much going on beyond 12 kHz.
>
> Could this be a way to go – or would it sound amateurish? After cleaning try to "generate" some air – if necessary – from hiss-free and non-harsh treble (7 to 10 kHz) by means of an exciter (the kind which actually adds freqs one octave above).

Please don't use these or any other settings as "recipes". The point is not to tell anyone where to boost or cut, just point out some counter-intuitive things to look for that a lot of beginners miss. ALL YOU NEED IS EARS. Really.

Go Red Sox.

> Quote:
>
> Originally Posted by TedR
>
> Actually I think yep is referring to a little known practice in many of the older major record label studios where the head producer wears red socks. In the event that their is a serious and heated disagreement between the artists and the producers as to the final artistic or sonic outcome of a particular song, the head producer has the option of displaying his socks, thereby exerting his control and dominance over the situation. Once the socks are displayed the artists have no other recourse but to accept the final decision of the head producer or risk having the entire project scrapped.
>
> It is rare that a head producer would resort to this drastic an intervention since it is considered by most to be heavy handed. In most cases a more diplomatic and tactful approach is used to resolve any issues that may arise during the course of production.
>
> Correct Yep ?

Nail meet head. That's it exactly, and my sign-off had nothing to do with opening day of the 2009 baseball season.

> Quote:
>
> Originally Posted by HOFX
>
> Yep,
>
> You wrote here (from memory) that the fx chain on most tracks contain 2+ stages of compression.
>
> Could you give one or two examples of this, with a brief description of what each compressor is trying to achieve?
>
> Thanks! Also, this thread has been most helpful!

This a good and timely question, and invites re-visiting some of the stuff from earlier in the thread in terms of overall process. Without checking back, I'm not 100% sure whether "most" fx chains would have this, but it certainly is very common to have more than one stage of compression (and more than one stage of eq and maybe other effects, as well).

Wringing out your signal chain

Before we proceed with compression (I promise I will get to it), much earlier in this thread I went through some very detailed and picayune advice on wringing out your signal chain, gain-staging, etc. I suspect that probably 90% of the people who have got this far never bothered with that stuff, but I started with it for the benefit of the 10% who are really interested in a systemic approach to getting the best sound quality.

The subsequent pages have been a lot more casual and theoretical about doing stuff like using filters and effects and mics and whatever, because they are starting from the presumption that you already know and trust your gear. Which as I said right at the beginning, is more important than having great gear. And as a shitload of threads on this site and on other sites prove, there are an awful lot of people who know every preset, recipe, and “magic setting”, and who own good equipment, and who also have know idea what half the controls, plugs, and switches are for. And when their expensive magic box doesn't sound as good as they expect, instead of reading the manual and actually wrapping their head around the basic technical operation, they start a new thread asking whether they need a new plugin or a different brand of magic box.

When you flick the hi-pass or lo-cut switch on your mic or preamp, that's EQ. I don't know whether it's good or bad EQ. When you push the “limit” button on your preamp, that's compression. I don't know whether it's good or bad. If you paid $200 for a little mixer or audio interface that has 8 channels of preamps and AD conversion and compression and EQ circuits on every channel, there is a significant likelihood that some or all of those features will either sound bad or rapidly become noisy or degraded with age or will not have consistent controls or God-knows-what.

Everybody has had headphones or guitar cables that get crackly or dull-sounding or cut in and out due to partial shorts or whatever. You can buy very good-sounding equipment very cheap these days but the quality control, durability, and mechanical and electrical integrity is probably not going to be military-spec, last-forever, heirloom-type.

You may have had the experience of trying to use a screwdriver where the bit was kind of worn out, dull, or notched. And you may have found that using this half-dead screwdriver just strips and kills the screws, and then you have to go back and drill out the screws to extract them. You may have had similar experiences using a perfectly good screwdriver that was the wrong size for the screw you're trying to drive. A bad tool, or a good tool used badly is worse than nothing at all. It's better to hold off on a project than to actively ruin it.

If you have decent monitors and more or less functional hearing, then all I or anyone else can do is to help you think through the stuff you're hearing. There are no rules. If it sounds good, it's good. And all you need is ears. Not magic ears, not golden ears, just ears. If you can hear the difference between good recordings and bad ones, if you can hear good sounds, then you can tell far better than I can what's working or what is not.

Let's walk through an entire process for, say, a clean-ish electric guitar. We could do this for any instrument, but let's say you have a very punchy, spanky Fender guitar sound on the bridge pickup with a close-miked amp. Let's say that the waveform has a sharp, near-instantaneous pick attack ("transient") that is some 10 or 12 dB louder than the average RMS "body" of the sound. And let's say that the "body" itself consists of a fairly short sort of stringy "thunk" that quickly decays into a low-level "note", which in turn decays more slowly into noise. The above is all pretty common. So far, we're just talking about the signal coming right out of the mic cable into the mixing console or preamp or whatever (or, for that matter, the signal coming out of your POD or amp plugin).

There are some frequency issues that might crop up immediately, even with a good basic sound. Some kind of DC-offset-type thing is pretty common with guitar rigs, especially in close-miking. If you look at the captured waveform, this is when it looks like there is sort of more "stuff" either above or below the zero-line, i.e. The waveform is not symmetrical.

So a simple high-pass filter to cut the extreme lows will help "re-center" the soundwave and open up some headroom as well as cleaning up rumble and mic thumps. So that might be step one, either at the mic's hi-pass switch, or at the console's input, or even in your recording software, depending on how you roll. This will also improve your downstream compression and other effects by eliminating non-musical and irrelevant content that the compressor would otherwise try to track and respond to.

In a similar vein, there is probably a fair amount of hiss and not much worth keeping above a certain frequency, so a high-cut or low-pass filter maybe somewhere between 7 and 15k might be in order. There might also be some obvious lower-midrange murk or soupiness to scoop out in the 150-400hz range, and/or some obvious low-end thumpiness that is going to conflict with the kick drum. Whatever.

Here we come to some divergent schools of thought when it comes to processing at the tracking stage, especially if we're talking about irreversible processing before the signal hits the recording medium:

1. **Moderate**: get rid of stuff that you obviously don't want in the track. Clean it up, erring on the side of caution. **Pros**: makes life easier further downstream, forces you to make sure you have a sound you can work with now, so you don't uncover a horrorshow at mixing, opens up headroom and flexibility for later processing. **Cons**: you might overdo it and wish you could take it back later.

2. **Purist/conservative**: Leave everything for mix-down. No reason to commit before you have to. **Pros**: you'll never regret a decision if you don't make it in the first place. **Cons**: leaves a lot of basic clean-up work for mixing, which is a pretty

big and complicated project to begin with. And commitment doesn't necessarily get any easier when you have 40 tracks to deal with instead of just one.

3. **Liberal/aggressive**: Keep only what you're sure you want, and get the track as close to perfect as you can as early as you can. **Pros**: maximum headroom/resolution, makes mixing a much easier and more creative process assuming you get it right, may help fire inspiration and creativity throughout the process by sounding more like a "record" from the get-go, forces you to critically evaluate the tracks and arrangement as you track, avoiding potential wish-I-coulda/woulda/shoulda at mixdown time when it's too late to re-track (e.g. this approach heads off "we'll-fix-it-in-the-mix-itis" by forcing you to confront the real evolving sound of the record as you go). **Cons**: big risk of getting it wrong if you don't know what you're doing. Beginners especially have a big tendency to misunderstimate what kinds of sounds will work in the mix as opposed to sounding good solo.

These decisions are more critical when recording to tape, since the natural effects of hiss, tape compression, and the uneven frequency response of tape mean that certain kinds of tracking mistakes will end up as semi-permanently embedded artifacts in the recording.

DAW makes it pretty easy to undo if you do everything digital, but negates some of the headroom/resolution benefits of cleaning up the signal in analog before "printing to tape" so to speak.

However, even if you have only a soundcard and a computer and do everything 100% in the box, I recommend at least considering experimentation with a moderate or aggressive approach, where you set up one set of effects while "tracking", then render all the processed tracks, delete the effects chains, and save as a new project. A lot of the benefits of these kinds of approaches are in freeing up psychic and creative energy downstream, and also in the fact that it is often easier to make specific, targeted corrections one-at-a-time as opposed to, for example, trying to recreate a wholesale dynamic profile with a single compressor. Not to mention the fact that upstream processing certainly has an effect on downstream processing.

So, specifically to the topic of compression, it is extremely common at this stage (especially when tracking to tape) to either engage a limiter switch or some mild compression or both to do any or all of the following:

- Knock a few dB off some of those 15dB transients with some limiting or fast, high-threshold compression (note that we are not even close to approaching modern "loudness race" debates at this stage, just getting the guitar into audible range if we have an average -14dBFS mix).
- Even out some of the differences between chords and single-note passages with some very light (maybe 2:1) compression at a low threshold.

- "Punch up" some of the thunky "body" that lives below the transient attack but above the "note decay" with medium-attack, gradual-release, medium-threshold compression that will exaggerate rather than compress the average dynamics of the track

Any of the above might just be regarded as part of the "tracking" process, just getting the pure instrument sound that you want to work with.

When it comes to mixdown, let's say you've got a good bass/drums/vocal mix going that's averaging around -15dB, and you need to bring in the guitar. And maybe this "spanky" guitar sound is having a hard time fitting in the mix without overwhelming everything else. Here you might want to pull up the average level and the decay a little bit, to get the guitar to "sing" and fill out a little more, so you might put some short-attack, medium-release, medium threshold heavy compression on the guitar track, to get the sustain to "swell" a little bit compared with the attack. And maybe that gives a little pumping/sucking effect where the initial "spank" rapidly drops down and then the sustain swells back up as the compressor releases. Maybe that's good, maybe that's bad. Maybe you need another stage of compression to further control or smooth out the front end of the pick attack, or to clamp down on the very tail to prevent pumping hiss.

And then maybe as the mix finally starts to come together, you through a little bit of compression over the whole rhythm section of drums, bass, and guitar to "glue" the whole thing together, and sort of tighten up the sound and timing a little bit, and to get the overall dynamics to "seat" a little better.

Sometimes a pair of jeans fits better after you wear them and wash them and wear them wash them again a few times. A sculptor usually works by first cutting out the rough shape, and then by blocking out the important features, and then finally by cutting the important details and then finishing the smooth curves. Painters very often paint many layers of color on top of each other. Writers almost invariably rewrite and revise. Whether these steps are theoretically "necessary" in a technical sense is almost irrelevant. The practical reality is that small, obvious, gradual improvements often lead to a better overall result than, for instance, starting with a block of raw marble and trying to perfectly carve one finger and then the next, and so on.

> Quote:
>
> Originally Posted by heater
>
> Well, i just wanted to thank yep and the rest of you guys for this incredibly informative thread. I've been playing guitar for 25 years, but have yet to record out of fear of digital recording and a general lack of any idea as to where to start. I've had a digital workstation for a few years now, but haven't been able to figure out just what the heck to do with it.
>
> Yeah, i'm pretty much an idiot. But this thread has gotten me to climb in from the ledge and close the window. So again, thanks so much for the info.
>
> I don't mean to threadjack, so feel free to ignore this yep, but if you or any of you other guys could give me a little advice, i'd be most grateful. I fingerpick on an old gibson, don't use picks, pretty much hillbilly and mississippi blues type stuff. I have a fairly cheap tube preamp, an cheap condenser, and an sm57. Oh, and one of those sure elvis mics. I realize my gear is a bit lacking, but i'm not really interested in creating a sound to compete with anything modern (those 20 foot tall acoustic guitars you spoke of and so on). Honestly, if i could capture a sound in the spirit of those first two Dylan albums, from songs like "Dont think twice" and "Boots of Spanish Leather", i'd be ecstatic. Just one track of nice, natural, warm, fingerpicked guitar.
>
> I live in an old farmhouse, with the main room being about 35 by 15 with 9 foot ceilings. If i understood the earlier posts, this would be a good room to record the guitar with the mic pulled back, to bring in the natural room sound (i could have completely misinterpreted that.) So, should i use the condenser? And would there be any need to bring in one of the other mics for close miking? What would such a step seek to achieve? And are there any other glaring concerns or processes i should consider? Again, just looking for a simple old folk sound, nothing fancy. Sorry if these are vague questions – just looking for some general tips. You guys have been a great help to this old newb, so thanks again. And props to yep for using "embiggen" in a sentence. Most cromulent.

Well, without knowing anything in particular about your sound, I will say that the particular combination of "fingerpicking", "old gibson", "old farmhouse", and "playing for more than 25 years" sounds like the audio engineering of a romance novel encounter. I mean, that's about on par with "concert pianist", "Mason & Hamlin baby grand" and "18th-century cathedral."

My first reaction is just to limit background noise, walk around to find the spot where it sounds best, and then set up the best mic I can get my hands on and count my lucky stars. Not much for me to do but to push the record button, keep quiet, and then send the file to duplication.

Really well-practiced solo musicians and ensembles with good instruments tend to engineer and mix themselves. The original "audio engineers" were people who wore lab coats and kept spare boxes of capacitors and tape rollers and and who aligned tape heads and adjusted capstans and that kind of stuff. Totally different from the modern role.

Really skilled musicians (including just about anybody who has been playing for 25+ years) tend to control their own sound. One really obvious example of musician "self engineering" is the simple fact that musicians tend to play single-note leads in the peak fletcher-munson frequency range (i.e. The upper midrange). This makes the "lead" sound louder proportionate to lower-range full chords, for instance.

Far subtler variations are the ways in which experienced musicians will, for example, play higher notes or lower notes with a different "touch", to get appropriate and balanced variations in dynamic and frequency profile. But even more important is just having mature, sensitive, balanced arrangements, even on a single instrument.

A lot of the stuff that is applicable to making a young garage band sound polished and professional is kind of beside the point with an artist who already sounds polished and professional.

What marks a great young pop/rock/R&B/hip-hop/electronic act is very often having fresh, imaginative ideas and an energetic, charismatic, and sort of "cup overruneth" creativity. Often, by definition, their inspiration outpaces their technical competency and musical maturity. And this is where a skilled engineer or producer can really help to turn an overloaded imagination and hyperactive creative vision into a polished, professional recording. That's where the 20-foot-tall acoustic guitars and so on come into play.

But at the risk of over-generalizing, as musicians get more experience, and as dedication and practice start to outpace fevered and unrealized imaginative vision, the performances start to become more sophisticated and complete in their own right. And the production and engineering roles start to diminish into a purer, more technical, and less-intrusive utility. So, with all that in mind, this is not at all a threadjack, but an absolutely fantastic way to bring in the topic of more naturalistic forms of recording.

(Cheap) microphones revisited

Specifically to your question, "cheap condenser" is a pretty broad term. An SM57 would not be my first choice for recording solo acoustic guitar, but it is certainly an adequate mic. Whether your specific "cheap condenser" is better or worse is impossible for me to answer.

For a whole lot of reasons, I'd prefer to avoid talking about specific gear in this thread, especially along the lines of "best mic under $XXX for acoustic guitar." Partly because there are about a bazillion other threads on exactly that topic, and partly because I have certainly never systemically tried every cheap condenser under $XXX on acoustic guitar and I'm pretty sure nobody else has, either, and even if they had, their results on, say, a strummed Talkamanie Artist Series would almost certainly be different from your results for fingerpicking an old Gibson.

There a lot of really good and also pretty bad Chinese-capsule condenser mics on the market in the sub-$400 range. Unfortunately, there are excruciatingly few people in the world who have extensive direct experience in this market. Professional engineers who might have the budget to buy all 300 of these mics and try them all out on a wide variety of real-world recording projects usually don't bother, because they can instead buy a handful of proven performers for the price, and frankly because their livelihood depends on getting professional results every day, not to mention on having big-name gear. The cost of the mics is hardly relevant to a big-name commercial studio (they could get them for free, anyway).

Their real cost to test them out is not the cost of the mics but the cost of studio time that they're not billing for (or worse, the cost of making real artists sit around playing the same stuff over and over) while they're conducting "shootouts" to see which $100 mic comes closest to sounding like a U47 or whatever, when they already have a U47 sitting there in the mic cabinet. I mean, if you were suddenly given a six-or seven-figure business loan to launch your music career, would you go out and buy hundreds of cheap instruments to try and find the one that sounds "almost as good", or would you simply get the real deal? Even if you also had free access to a truckload of the cheap knockoffs, would you even bother?

Moreover, there is a serious and fundamental problem with trying to conduct a clinical "shootout." Real mics are used in the real world in different ways. A good mic with a flat frequency response and a broad pickup pattern is going to sound completely different from an equally "good" mic with a tight, focused pickup pattern and a "big"-sounding proximity effect, depending on how you use them. Do you put them both in an iso booth or an anechoic chamber 4" from the source pointed dead-on? If so, then you've completely negated the very significant differences in pickup pattern, and the "proximity effect" mic is going to sound a lot more bass-heavy than it would if it were a little further away... if you put them both in a "live" room 3 feet from the source, then the sound of room is certainly going to compete with the sound of the mic when it comes to the broad-pickup one (for good or for ill, depending on the room), and the "big proximity" mic might get cast to the wayside only because it was being used in completely the wrong context.

So anyway, having now ranted and raved against "recipes" presets and "best X for $XXX" kinds of stuff for umpteen pages, I'm about to take a step backwards and offer some cheap gear and "purist" mic recipes, with significant caveats. Anyone is of course welcome to join in, but if you find this thread helpful, I encourage you to keep it so by focusing only on stuff borne out by competitive real-world experience. I.e. If you have only ever tried one condenser mic in a particular price bracket, please abstain.

Keeping in mind all the caveats in the above post (and throughout this thread in general), here are the mics that I have used in (I think) the sub-$100 category that are really head-and-shoulders better than other super-low-priced examples, and specifically how they compare to an SM57 (which is sort of the de facto sub-$100 studio mic). Bear in mind that I have not even come close to trying everything, and that most of these examples have been around for a few years, and there may be better alternatives. And none of them are necessarily "best value" mics, just standouts in the "super-cheap" category. I.e. Sub-$100 mics that I would personally be okay with using on a paid project Vocal mic: MXL V67G – extremely "big" sounding LD condenser with massive "movie announcer" proximity effect. Very forgiving near-range placement/pickup that picks up minimal room sound while still sounding consistent when recording a moving head. Very hype and "big" sounding, with a forward, slightly crunchy top end that smooths and flatters dull, weak singers but that might turn a little brittle on airy females or whispery males. Like a slightly overdone impression of classic tube mics. Most "expensive" sounding mic I know of under $100.

Close instrument mic: MCA SP-1 – at $40 apiece, this is an unbeatable deal. Like a condenser version of an SM57 –forward, present, but slightly faster and more sensitive, and with more depth and low-end clarity. Very focused polarity makes it tough for vocals and might be a little fizzy/"too sensitive" for close-miked heavily-overdriven electric guitars. Awesome on drums, not least because you don't have to worry about the drummer killing an expensive mic.

Far instrument/all-purpose wide-pickup mic: I have to split this between two picks. Probably the most useful to readers of this thread will be the line of MXL 603s/604s, which are wide-pickup, extended-range small diaphragm condensers. Very accurate, very "airy." Almost too airy, in fact. The sensitive high end has a tendency towards brittleness compared the German pencil mics it's modeled on, but at 1/15th the price, who's complaining? Good for Ohs and room mics, and not bad for acoustic guitars, piano, or clean guitar amps where slightly hyped clarity and realness are desired. Second pick is the Behringer ECM8000. This is an absolutely fool-the-ear accurate omnidirectional mic that looks and sounds like a knockoff of Earthworks reference mics that cost about 100x the price. If you throw this mic in the room while people are talking and then play back the recording through decent monitors, people will start responding to the recorded conversation. It's that accurate. For good or for ill. If you have a great room and $100, then a pair of these might be all you need. If you have a bad room, then these mics will pick up all it's badnesss with nary a trace of flattery or forgiveness. Note that both of the above mics have high-ish self noise and will produce more hiss than their more expensive inspirations (or an SM57, for that matter).

-Last but not least, all the above reviews were written in reference to the venerable SM57.

Originally made under contract to the US military, the SM57 is designed to deliver accuracy, clarity, rejection of feedback/background noise, and ease-of-use/placement. It delivers all in spades, and is arguably the most useful mic ever made. It is unarguably the most widely-used professional mic ever made. It is a dry, direct, very forward-sounding mic with a slight "flattening and fattening" effect, almost like a high-grade telephone. If you look at those frequency charts above of produced records, they are pretty close to the frequency response of an SM57. It has a very forward presence range but would never be described as an "airy" mic, and lacks the fast response, depth, richness, and detail of more sensitive condenser mics. It is no surprise that its most popular studio applications are electric guitars and snare. Its sister mic, the SM58, is basically the same mic with a wider pickup pattern and a built-in windscreen, and has become the de-facto live mic for rock vocals.

So having said all that, companies like Behringer and MXL also make a lot of crappy mics.

Endorsement of one model is not endorsement of a brand, and "cheap condenser-itis" is not a good sound, compared with the all-57 ADAT sound of budget studios from 15 years ago. If anyone has used these mics and prefers other in the same price range, then you may well be better-informed than I am. I hope anyone offering suggestions will be clear and honest in their experience of other mics. There is nothing more useless than 100 people all recommending the only condenser mic they've ever used.

> Quote:
>
> Originally Posted by Lokasenna
>
> A year or two ago, some guy in a music snob chatroom was telling me that neither his studio nor any pro studio he'd ever been in had a single SM57, nor would any of them ever want such a shitty mic. I almost died laughing.
>
> When most studios have at least one, if not three/four/ten, I think it's a pretty safe bet.

As much as is pains me and runs against my nature to take sides with audio forum wankers, he might not have been that far off if it was recent and he's only been in studios for a few years. Recording studios are dying out left and right and pro audio engineers are lucky to make minimum wage, in the US (even ones with gold-record resumes). On the one hand, the commercial record industry is rapidly eroding due to the vicious cycle of piracy and the accompanying consolidation and conservatism of major labels and distributors. On the other hand, cheap digital recording has empowered musicians who, a decade ago, might have saved up $2,000 to record an album to instead spend the money on mics and a better soundcard.

As a result, recording studios are under increasing pressure to find some way to set themselves apart, and a lot of them are in a desperate scramble to do so with gear lists that do not include anything that a home recordist could afford. The irony is that, as home recording has exploded, and with it an increasingly affluent hobbyist contingent, the prices and demand for "vintage" and "boutique" stuff has shot up. The manufacturers want to get their stuff into "pro" studios, so they can use the name to sell to amateurs, and the remaining "pro" studios are eager to showcase a gear list of all-boutique stuff, so they are happy to trade endorsements for free or discounted gear.

As a result, you get studios with a name engineer (who might be having a hard time affording his daily Ramen, but who once set up mics for a Judas Priest record) replacing all of his SM57s with free or discounted boutique mics from some guy's garage, who in turn wants to use the endorsement "Joe Blow (judas priest, et al) says ‚Great mic! I've scrapped all my SM57s and replaced them with XXX! I wish I had them when I recorded JUDAS PRIEST!'"

Now, never mind that the only gold record the above engineer was ever affiliated with was a Judas Priest record recorded with 65% SM57s. If you call him today his first line over the phone is going to be, "lemme guess, you're trying to make a record with an SM57 – we throw those things in the trash around here. All our mics are handmade from solid blocks of aluminum by this expert guy in his garage who does magnetic imaging tests and only makes 83 mics a year and they cost $1,000 apiece, retail. No wonder you can't get good results."

Bash him all you want, but you're not 60 years old with a Billboard resume and nothing to show for it but bad tattoos.

> Quote:
>
> Originally Posted by Rec
>
> Yep, great threat. Could you elaborate on this? Lets say we have a mean four on the floor Rock groove @ 120bpm with drums playing a straight 4/4 beat. Kick on 1 & 3 snare on 2 & 4 and a Fender Jazz Bass pumping 8th notes. In very general terms (since I know this will vary a lot) could you describe how you would EQ the two scenarios you mentioned.
>
> Scenario 1: fat from bass, punch from kick
>
> scenario 2: punch from bass, fat from kick
>
> and how they sound different, maybe with some examples of songs you know.

Well, it's not just eq, and sometimes "fat" and "punch" mean different things to different people. And if the kick only hits on the 1 and the 3 but the bass plays every 8th note, then that's going to dictate a lot in terms of the approaches that will work. Same if you have a DW kick drum and a flatwound Gibson bass, for instance.

But let's say there was a really "deep", "punchy" electric bass sound that could be gated and compressed to get a sharp, somewhat percussive dynamic profile and that didn't have a lot of lower-midrange harmonics "fattening up" the sound. This might complement well with a "big" "vintage" drum sound where the kick drum sounds an actual note that sustains for a quarter-note or more, and a snare sounds that might sustain for a half-note. This could be accomplished with reverb decays and/or compression to extend the sustain, assuming there is something there to work with in the first place. That would be sort of opposite of how most modern mixes are done.

Again, it's not about "settings", and you can't just take an eq and turn one bass or drum or mic position into another. It's about everything from the song to the arrangement to the instrument selection to the playing style to the setup to the initial gain-staging and so on. That shouldn't stop your from trying, though. In the long run, the way to get good is to push the limits and figure out where the boundaries are, and you'll start to develop an intuitive sense of which kinds of bass sounds will work well with which kinds of drum sounds and song arrangements and mix approaches, for instance.

Specifically, the problem with trying to say "how I would EQ" the instrument that is supposed to sound "fat" is hard, because usually the "fat" instrument is the one that already SOUNDS "fat." So I might actually be **cutting** the "fat" frequencies on that one.

It's not like sound design on a synthesizer, where you just create the sound out of whole cloth. You're taking pieces of an imperfectly-made jigsaw puzzle and trying to trim them to fit each other as best you can.

> Quote:
>
> Originally Posted by TedR
>
> In your opinion, would you say the Marshall MXL 604 compares favorably to the Shure SM 81 on acoustic guitar ?
>
> I haven't heard either but I was considering purchasing the 604 as a less expensive alternative to the SM 81 and would appreciate your opinion.
>
> Oh! Do you have any idea where to find an MCA SP-1 ? I've read they are discontinued ?"

There are literally *thousands* of threads on the web comparing various cheap mics to more expensive mics, and to other cheap mics, and medium-priced mics, and so on. You can't swing a dead cat on a stick without hitting a "best mic under $X" thread these days.

I'm sorry to say I have nothing to add to that line of discussion, and no advice on where to buy. The mics I mentioned are all really good mics that also happen to be very cheap. Whether they are better/worse than anything else or how close they are is not something I even really know how to talk about intelligently.

> Quote:
>
> Originally Posted by soul&folk
>
> yep, I found out about this thread from a link in the Tapeop forum. I've read the first few pages in one sitting and intend to catch up and follow along from this point forward.
>
> You seem to have a lot of firsthand knowledge and not just book-smarts. Do we get to find out who you are (there is no information in your forum profile and all we learn from this thread is that you're in Boston), or will you remain as "Batman", hiding under your yep mask?"

I'm nobody important! (At least not in the audio world...). But I am somebody that has a day job and other pursuits that require me, sadly at times, to maintain an illusion of respectability, and I'd rather not have my name or personal details show up in Google searches with pages and pages of me talking about how to record guitar in your bedroom. (Believe it or not, in some fields, that kind of stuff can make it hard to be taken seriously. Now, if I were a golf forum regular, on the other hand...)

> Quote:
>
> Originally Posted by VortexOfShit
>
> ...I've been following the advice here. In summary, I'm mostly deeply cutting fundamentals, bass and high cuts on almost all tracks. Then I'm being very selective on a few tracks for each song that will hold the key positions – bass (usually bass and kick, nicely separated); mid-range (mostly vocals); mid-highs (mostly guitars – a lot of cutting required here); and highs (mostly snare and cymbals). In those few cases, I'm letting those instruments form the foundation – everything else has to get out of the way and make room...

I'm glad to hear that this has been helpful. I also want to respond specifically to the stuff that you cite as being "the advice in this thread" RE: cleaning up the tracks with EQ and such. My guess is that those kinds of mixing strategies will be the first take-away that most people get from a thread like this, as much as I've tried to stay away from "presets" and "recipes." After all, a simple eq rip to roll off the lows and highs works wonders on most home recordings, and is probably the fastest path to improvement.

That said, the REASON why these kinds of things offer such a blanket improvement is because most home studio tracks were recorded wrong to begin with (not to mention that many amateur arrangements are flawed from the outset, although these days the line between "mix" and "arrangement" is often a very blurry one).

Going back to the post by heater, who said he wasn't interested in "20 foot tall acoustic guitars" but just a clean, accurate recording of his sound, how many home recordists (or professional musicians, for that matter) really subscribe to that notion? If I were producing your record and offered you the choice, would you really say "oh, no, I Don't want that big modern sound, just an accurate recording of how I sound live..."? When you look for reference material on the radio or among your record collection, do you try to duplicate all the thousands of subdued, well-placed, well-seated acoustic guitar tracks that back up the melody and reinforce the bass and drums, or do you always go for the one "sound of God" 20-foot-tall version?

There is nothing wrong with the latter, except when you also do the same with the drums, and the bass, and the vocals, and the synth, and the electric guitars, and the piano, and so on. When everything is huge, it's like trying to take a picture of the sky over the ocean – it just looks flat, and boring, with no sense of scale. And that's a best-case scenario. Worst-case, everything is fighting and burying everything else and it all sounds small and annoying under a big crush of conflicting mud and fizz.

What happens is that everyone is chasing that one awesome bass sound that totally sold the song, and also that one awesome guitar sound that totally stole the show, and also that one screaming synth lead that blows everyone away, and that massive vocal that seems to come from all around and above, and that huge drum kit the size of mount everest, and so on, all at the same time. Which is really hard to pull off.

The thing is, if we really stop and listen to those reference tracks, the REASON those drums sound as big as mount Everest is because they DWARF the guitar sound and everything else. If the guitar is trying to be bigger than the drums and the drums are trying to be bigger than the guitar and everybody is trying to be bigger than everybody else, then you just end up with a shirtless David Hasselhoff lying drunkenly on the floor of a hotel room trying to shovel a disintegrated Wendy's cheeseburger into his mouth, moaning "This is a MESS..."

This instinctive draw towards everything bigger, hyper, and hotter affect mic choice, mic placement, gain staging, processing, everything. In a solo shootout or A/B test, we always reach for the mic, preamp, eq curve, etc that has more, bigger, hyper. And ears calibrated to modern loudness-race records don't help matters. So we track every single source with the hypest, hottest, loudest, biggest-sounding signal chain we can, and end up with David Hasselhoff. Then some jamoke like me comes along and suggests a few eq rips to undo all the bigness that we worked so hard for, and all the tracks sit better and sound cleaner and flow and breathe better.

But hopefully these exercises start to lead to better, more tasteful, more judicious tracking in the first place, and to a better understanding that what sounds best solo does not always work in a mix. Moreover, something that was tracked right in the first place often sounds a lot better than using eq to undo something that was tracked with the mic shoved right in the source and the preamp gain cranked and so on.

A lot of that "air" and "warmth" and "smoothness" and "punch" that we attribute to vintage gear actually comes from vintage **PRACTICES** and *TECHNIQUES*. When we start with good arrangements, good performances, good instruments, good setup, a good environment that is free of rattles, squeaks, electrical noise and undue resonance, then we can record a naturalistic-sounding representation of the instrument with authenticity as the main goal. And when the musicians have a live, rehearsed, polished performance that has natural dynamics and instrument balance, and that doesn't count on the studio to "make it sound right", then authenticity becomes a perfectly acceptable goal.

And when we're starting from the proposition of trying first CREATE, then CAPTURE or "record" a finished soundscape, then maybe we don't need all the whispered vocals and triple-tracked guitars and soaring strings and growly bass-leads and clicky whirring synth sweeps to make the music sound good. There is nothing wrong with that kind of "manufactured" soundscape, but if your recordings don't sound good WITHOUT all the icing and sprinkles, then maybe it's time to get back to basics. And if you need a lot of hot tips n' tricks and cool eq curves and vintage compressitubifiers to make the tracks sound "warm" and "punchy" and "musical", then maybe something is missing at the source.

[begin page 15]

> Quote:
>
> Originally Posted by nerdfactormax View Post
>
> As much as it feels like asking a doctor how best to use cocaine; What general rules and philosophies (as opposed to specific plugins and presets) would you suggest for mastering (and indeed mixing, if it applies) for the times when you want to compete in the loudness race?

Shelve down all the lows and lower mids on the master out a few dB more than you otherwise would (let the listener adjust the tone controls). This will free up a lot of headroom to turn up the presence range. Clip or limit all transients shorter than a few ms.

Pan strategically, and dynamically as necessary, to maximize use of both channels. Send the bass into a fast compressor with the drums triggering the side-chain to clamp down on big bass waves during drum transients. Similarly, use vocals into the side-chain of a slower compressor to duck guitars/synths slightly during singing, which will allow you to crank the guitar/synth a few more dB.

But honestly, the real-world cases where there is ANY reason to try and compete with the loudness race are so few and far between as to be practically nonexistent. Every commercial broadcast and most modern public sound systems already use broadcast processing to level-match everything, so people are going to hear your songs at the same average level as everything else, meaning that really loud mixes/masters are only going to end up QUIETER and more degraded in these scenarios. The radio and TV stations have a very strong interest in maintaining consistent playback levels, and they will not allow your record to play back louder or quieter than any other. And CD listeners at home and in the car can and will simply adjust the volume to taste, defeating any attempt you make to try and sound "louder." And even most contemporary mp3 players and computer media players have some kind of level-matching built-in these days, to protect you during "shuffle" play, assuming the listener is not capable of adjusting volume to taste. Which frankly most people do constantly (adjust volume, that is).

This is the real tragedy of the loudness race – it's really and truly pointless, if not outright self-defeating. It is based entirely on ignorance and the misunderstanding of in-studio A/B tests that **seem** to be neutral and empirical but that are actually completely misleading.

Going all the way back to the very earliest posts in this thread about level-matched listening, and "louder always sounding better, even when it's worse", the loudness race WOULD BE defensible IF you could actually use signal level to reach through the speakers and turn up the volume at playback. But you can't, no matter how much it seems otherwise. The listener has the volume knob. All you can do is to either make clean, dynamic, exciting, high-quality recordings that the listener will want to turn up and enjoy, or grating, shredded, flat, unnatural recordings that the listener will keep turning down due to ear fatigue.

Edit: It should really be called the "FLATNESS race", not the "loudness" race.

> Quote:
>
> Originally Posted by VortexOfShit
>
> ...I think next time around, if I know the song will have drums/bass, I'll do some very quick guitar/piano/vocal roughs that I know I'll throw away. Then I'll lay down the bass/drums foundation. After that I'll track the guitars/vocals/piano to be used in the mix, monitoring the drums/bass during setup. I think this will help me to make better decisions on micing technique, placement, etc.

Yes, this is exactly right. One of the few actual "rules" of recording is that starting with the drums+bass (or drums then bass) produces the best recordings. Incidentally, the same is true with mixing. It is usually always best practice to start from a good drum and bass mix, and then build the rest of the mix around it (I usually like to bring in the vocals first). We could talk about and debate the reasons why all day and all night, but the reality is that it just tends to work better.

Also, a "scratch track" of an actual performance of the song with vocals is always better than a click (even if the scratch was cut to a click). So the best process is **usually** scratch track>drums, bass> rest of the rhythm instruments> leads and vocals> sweeteners and backing parts.

It is very hard to play good accompaniment without hearing some version of the vocal cues. And getting better at mixing leads to getting better at tracking, which leads to getting better at performing, which leads to getting better at arranging, which eventually makes you a full-menu producer. I.e. Someone who just has an intuitive feel for what makes a good record, even of a bad band.

> Quote:
>
> Originally Posted by TedR
>
> Is there a magic secret that will allow me to add verb to an acoustic guitar recording without destroying the clarity ?

Hurm. Go back and check page 13 of this thread if you have not already done so. Lokasenna gave some useful suggestions but it's very hard to generalize about reverb.

There is no "magic secret" but if you're struggling there is probably some help. There are, broadly speaking, two kinds of reverb, dictated not by the brand of processor nor the specific settings but entirely by how they are used to affect the sound. The first and most obvious kind is the audible "effect" reverb such as included in synths and guitar amplifiers. This is the kind most familiar to musicians. The kind more widely used by mix engineers and producers is a much more subliminal kind of "sense of space" reverb. Where you are not trying to make sound bigger or watery or far away or spacey, but just trying to create a sense of place and depth.

The latter kind of reverb is all about the details, and they vary a lot from one source and one mix to the next. The trick is that it usually should not "sound" like reverb. In fact it usually shouldn't "sound like" anything at all. It should just be a sense of spaciousness that you only notice when it's gone.

Finding the right decay, so that the reverb tails out as a proportinal extension of the notes, and finding the right balance of high-and low-filters so that the reverb sounds like a natural resonance instead of an "effect", and finding the right reflection density so that the reverb naturally complements the sound of the instrument and sounds appropriate to the sonic "space" that it should exist in (e.g. a living room vs. A concert hall), and finding the right predelay so that the reverb has an appropriate sense of size and separation relative to the virtual "space" you're trying to create is all a matter of trial and error. As is the all-important task of simply picking the right "flavor" or reverb to begin with.

This last might actually be the hardest for home studios. As I said earlier, a lot of reverb plugins suck. A lot of them sound metallic or ringy or splashy or essy and harsh or dull and mucky or just generally fake, and so on. The reverb algorithm itself will determine the sonic quality of the "room" that your sound is placed in. The rest is just fiddling with where the listener is seated in relation to the performer and so on, and is frankly easier to do than it is to describe, because really who cares whether you put them in row 3 or row 25 if they're sitting in symphony hall? Just mess around a little, keeping in mind that reverb should be at a subliminal level, and flow with the vibe. Setting up the reverb should be fun, and kind of a right-brain creative thing compared to, say, adjusting the compressor or tuning out esses with an eq and so on.

Not quite sure if that answers your question, but don't hesitate to ask...

> Quote:
>
> Originally Posted by Rec
>
> GENERALLY, Should all drums trigger the side-chain or just kick & snare, or maybe something else?

Just kick, or kick+snare are the most common. But fortunately Reaper's routing makes it ridiculously easy to try any combination you can think of. This is the kind of processing where it is not so much an effort to "sound good" as to find out how much gain reduction you can get before it starts to sound bad. It's also completely pointless, as I said above.

> Quote:
>
> Originally Posted by Marah Mag
>
> Completely pointless, OK. But it was also (I thought) one of the more interesting ideas in this idea-rich thread. The principles at work in the procedure are worth being aware of, since they would apply and be useful in contexts other than the loudness race. You can see just how and why it would work. It's quite clever, really. I plan on giving it a go just to see what kind of damage I can do.

There is certainly nothing wrong with maximizing signal level, as long as it is non-destructive to the sound quality. If nothing else, it gives more resolution.

Similarly, side-chain ducking can absolutely be used as an aesthetic effect. A lot of dance music ducks the bass with the kick to create a tighter, more unified low end, and to lock the bass and kick together. And using a smidgen of "vocals ducking guitars" can help the impression of "loud" guitars that overwhelm the vocals in level while still allowing the vocal to be heard.

What is pointless is trying to make a mix or master hotter for the sake of being hotter. It basically never makes any real-world listener's experience of the music "louder", just "flatter." (Please note that there ARE certain occasions where a "flatter" mix might be desirable for some reason or another, but in those cases a "flat" mix normalized to -12dB is just as good as, and usually better than, a "flat" mix normalized to -.03dB).

> Quote:
>
> There was also something poetic, almost haiku-like, in how the recipe read. Check it out! (This is verbatim... with a little editing, it would totally sing!)

I'm a poet and I didn't even know it!

> Quote:
>
> Originally Posted by nerdfactormax
>
> I had a hunch that the Loudness Race strategies would have wider applications, but my specific scenario at the moment is this:
>
> Trying my hand at mastering (for the first time) on a friends band. The tracking/mixing was (it seems) done in a bedroom studio and suffers from uneven bass guitar (1 or 2 notes dominate and the rest disappear) and massive transients on the kick and snare eating up all the headroom.
>
> The songs are destined for the myspace arena, so trying to retain quality is partly an academic exercise. I tried to get the average loudness somewhat close to a reference cd the friend gave me, but struggled to deal with the kick and snare transients.
>
> Main Questions:
>
> When dealing with transients and trying to gain headroom, is a compressor with pre-comp useful or should I turn to soft/hard clipping/limiting? AND
>
> should low frequency transients (kick) be dealt with in a different way? I tried twiddling with some precomp and quick release, but gained hardly any headroom before noticable pumping.

Mastering

Mastering is a completely, completely different animal, and I'm not sure how much I have to contribute to a general discussion of mastering other than to say don't master your own mixes. If they don't sound right, fix the mix. There is nothing that a home recordist can do in the way of self-mastering that they can't do better by re-mixing.

The point of "mastering" (in the sense that most recording forum-goers think of it) is to fix the stuff that's wrong with the mix. Which, if you can hear it in your own mix, should be fixed in the mix. That way the mastering engineer (even if it's you) doesn't have to worry about anything other than duplication. What nerdfactormax is doing is not actually "mastering." It is "making a two-track mix sound better." Which **is** something that mastering engineers frequently do, because they often get flawed mixes that they can improve, but that process is a sort of "pre-mastering" (in fact, pre-mastering is exactly what it is called).

From that point of view, audio is audio and there is nothing different about processing a two-track recording of a whole band than anything else in this thread. You can do whatever you want to make it sound better. Frankly running the whole thing through a distortion pedal should not be ruled out.

To the specific point of evening out the dynamics, you could obviously a compressor or a limiter, or use Smurf's method of manually drawing in envelope changes, or simply ride the faders. The obvious challenge with any of these is that whatever method you use to drop the level of the kick drum is also necessarily going to drop the level of every other instrument. Whether that pumping is good or bad, and which method will be the least offensive on any particular source material is an open question, and the "best" answer depends on your skill, your gear, and how much time you want to spend.

Using a multiband compressor can mitigate some of the pumping artifacts. For example, if you took two tracks of the exact same source material, and use a high-pass filter to cut off all the lows at say 500Hz on one track, and used a lo-pass filter to cut off all the highs ABOVE 500Hz on the other, and then sum the two tracks together, they'll basically be the same as the original source material. Except now you can apply a compressor to ONLY the material below 500Hz, and then mix the compressed lows back into the unprocessed mids and highs. This might allow you to sculpt the dynamics of the kick and low bass without causing the sensitive vocals to suck and pump in and out on the kick hits.

The above is the oldest and original form of multiband compression. But lots of modern multiband compression (a.k.a "dynamic eq") plugins automate the whole splitting and summing business, making it very easy to have lots of bands, maybe including one "tuned" to the spikiest frequency of the snare drum or some such. So there's one approach.

Parallel processing is another potentially useful trick in situations where it's hard to get the just the right processing. You basically clone the track, then apply the hardest, flattest compression (or whatever) to one track, to try and completely flatten out the level, and then mix it back in with the unprocessed track, allowing you to more easily "tune" the critical balance. This is sometimes called "Motown" or "New York" compression (don't ask).

You could try multiband processing on the parallel, and you might get a very transparent form of compression. You could take it a step further and try filtering the "unprocessed" track with eq or some such to try and zero in on say the vocals and strings or some such to try and really separate the two streams into "needs compression" and "needs to sound uncompressed." You might even end up with one compressed track of just the lows, one uncompressed track of the whole mix, one focused, uncompressed track of the "vocal focus" eq, and one compressed track of "snare focus", then mixing those four stems together to effectively try and "remix" the song as though you were mixing a live multitrack with a lot of bleed.

You might also find that some reverb or delay applied after the compression (of whatever sort) might help to "smooth over" the pumping. You might also try some saturation or mild distortion effects (maybe in parallel or multiband) to substitute a little added "crunch" in place of compressor pumping. This might happen either in place of or in conjunction with compression, parallel, multiband, whatever.

A mastering engineer might also use creative phase cancellation, gating and eq'ing a clone to isolate just the kick, snare, and bass, and then inverting the phase and mixing with the original stem to try and reduce those specific elements.

Last but certainly not least, don't overlook plain-jane eq. If you can hi-pass at 50Hz, shelve down everything below ~12k by 4db, and scoop out some of the "mud" frequencies at ~250Hz, then you might be able to turn up the track by 4dB or more before clipping.

The slightly lighter bass might be less offensive than compression artifacts, and will almost certainly make the band happy in a straight A/B test if your ethics don't prevent you from "cheating" by using the loudness effect to deceive them in that way. You could also of course combine this version as yet another stem with some of the other processes above.

Good mastering engineers will combine any and all of these techniques, and others, to "remix" material as necessary in the premastering stage, which gives them a reputation for being magic-workers, which in turn leads to the misperception that "mastering" is the key to great sound. But as you can see from all the above, it would be much, much easier for a home recordist to simply go nac and re-work the mix if the kick and snare are too loud.

These kinds of techniques push the technical and aesthetic limits of audio processing, and working all these stems and crossover filters for multiband and so on tends to introduce progressively more and more phase smear and other processing artifacts. For this reason, professional mastering engineers tend to be pretty obsessive about using specialized, high-quality equipment and minimalist processing whenever possible. With 50 stems and unlimited processors, you could practically remix the whole song this way, but the audio degradation would be worse than the improved mix. Moreover it would be exponentially more difficult and time-consuming than simply doing a remix of the original source tracks.

"High-pass" means the filter "passes" everything above the cutoff frequency and blocks everything below the cutoff frequency. So the filter shape should sharply curve down to an infinite gain reduction, depending on the steepness of the filter (Q setting). "Low-pass" is the same thing, except in reverse – the highs are blocked, and lows are "passed" through the filter.

A "**shelving**" filter is just like a "step" or "shelf" that evenly raises or lowers everything past the filter frequency. It's like a tone control on a stereo.

As an aside, using terms like "high pass" to refer to an eq curve that cuts all the lows might seem a little confusing – why not just say "low cut"? The thing is that "low cut" could refer to a shelving filter or even a notch filter. "High pass" clearly describes a hard cutoff filter that only passes frequencies above the cutoff.

Hope that makes sense.

Personal comment on specific recording advice

I apologize in advance for this somewhat personal post – people looking for recording/mixing/audio advice can skip it:

Apparently in response to my (hopefully helpful) comments in this thread, I've gotten a few Pms similar to the one quoted below (personal info removed):

> Quote:
>
> yep, late last year I tracked 12 songs for some friends band...
>
> So i have the songs mixed and i'm just doing final tweeks before I send them of for mastering, but as this is my 1st recording project (except for doing a few of my own songs) I,m not really sure where to say thats it there ready. So i'm wondering if you'd mind having a listen to 1 of the tracks and giving me your opinion on if it sounds right or if it needs more work in some areas ?...

While I am certainly flattered to be asked, I have a couple of generic thoughts to offer:

- I like forums because they allow everyone (myself included) to learn and benefit from the specific challenges and wisdom of individual experiences. I am not necessarily a better judge of your recording than anybody else, and other people might offer better advice than me. Just as importantly, other people in a similar position to you might benefit from the public discussion of your specific concerns and challenges.
- I post on public forums in the hope learning and of sharing my own insights. Private consultations are another matter. My time is limited, as is everyone's, and it is worth something. If I offered you a gig at a public venue in front of a public crowd, you might take it for free, just for the pure sake of sharing your music. But if I asked you to come over to my place and play a private concert, that might be a different scenario.

- I hope it will not be seen as a solicitation but just as a piece of info when I say that I do remixes at nominal rates for full-album projects that I like, especially if they have consistent instrumentation. Hourly and project rates are variable, but I'll mix a typical "cheapie" self-tracked garage band album mix for ~$500 depending on the project, and a full-blown (remix) production with additional layering and instrumentation for double or triple that. I always provide all stems, the complete project files, and detailed notes with every project. And it won't need "mastering." (these days you may well get much bigger names than me for less). It does not take me any less work or time to load a project into the studio, figure out the changes, and type up the notes. So I'm not really able to give a "discount" to just provide the advice and settings and not do the work.
- That said, if you have the courage to post your files and accept public criticism, and if you ask good, informed questions and have put some effort into trying to do it the right way to begin with, and if I happen to stumble across it and have time to respond, I'm happy to offer general opinions on recordings if I think I have something to contribute. Public discussions of specific examples help everyone, and free advice is worth what you pay for it.
- Last but not least, if you specifically want my general opinion of something, just send me a link (or better yet, post it). I dislike promising in advance to give input on something, especially without knowing whether it already applies all the free advice that is already available.

Sorry for the aside. I post it here only because I've had a number of similar requests, and that tells me there might be other people wondering the same things.

> Quote:
>
> Originally Posted by Evan
>
> In the subject of high-passing everything... I am worried because I have read about distortions and phase shifts especially with utility track Eqs (as opposed to high quality and CPU demanding latency inducing linear phase -whatever they're called-EQ).
>
> So even though I have gotten into the habit of highpassing most tracks into a mix I am working on, I fear I may be making the way for a sterile end-result.
>
> I am an amateur and I cannot make proper judgments on this. What I did, on this rock song I am working on, is reserve the low-end for the bass and kick, and high-pass everything else from 150Hz-200Hz down (either the ReaEQ gentle high-pass or a JS 12dB highpass plugin).
>
> Maybe high-pass filtering is better suited for tracks that have **problems** (noise, rumble etc) on the low end? Maybe shelving filters are smoother and more gentle for reducing lower frequencies?
>
> Thanks

That's a big question, and a big topic. YES, eq does cause phase smear. So does tilting your head to one side or another while listening, or moving the mic off-axis, or listening with speaker at an angle, reflections in the room, everything else.

"Phase" is just an element of sound. "Phase problems" are no different from "bad sound" in that sense. So if you are doing something that makes bad sound, stop.

Please note that nowhere in this thread or anywhere else have I said that you should high-pass everything. Just that those filters are often overlooked by beginners, and something to experiment with. "Rules and recipes" are exactly what I have hoped to avoid.

But the most important part of your post is this:

> Quote:
>
> I am an amateur and I cannot make proper judgments on this."

If you can't hear the difference between good sound and bad sound then neither I nor anyone else can help you to make good recordings. However I very much doubt that is true. Can you tell the difference between a good-sounding recording and a bad one? If so, then you CAN make the necessary judgments. Once you start to learn in a practical sense WHY certain things sound bad or good, and how to bridge the divide, it will become a lot easier and progress faster, but the ability to hear "sounds good" and "doesn't sound good" is all you need.

Go all the way back and re-read the first page of this thread. The very first and most important step is to trust what you hear. Trying to make good recordings without that trust and confidence is madness. And ALL YOU NEED IS EARS. Seriously, go back and start at the beginning. That's where the most important stuff is.

As an aside, if you want to hear what severe phase-smear sounds like, take a track, put a bunch of steep, narrow eq boosts on it, then drop in a second eq plugin and put a bunch of exactly opposite eq CUTS, to undo all the boosts you just did. Now toggle the fx on and off.

The eq'd version will sound sort of veiled and grainy and generally “lower quality” than the unprocessed version. The steeper and narrower your eq, the more phase distortions you'll get. So if you do the same test with just a couple of very shallow, low-Q-setting boosts and cuts, the effect might be almost indistinguishable from the unprocessed version.

And all of this kind of stuff is the reason to do everything as well as you can, every step of the way. If you find yourself shelving all your tracks, then maybe it's time to look at the mic and placement and recording techniques you're using.

> Quote:
>
> Originally Posted by stupeT
>
> Ehrmm... maybe ReaEQ is implemented phase-neutral (linear phase)?”

No, I suspect I made a mistake by suggesting a test I've never actually tried, and that nerdfactormax's first guess was correct – that the phase distortions from one digital eq neutralize exact opposite phase cancellations from a previous instance of opposite eq. My bad! I'll try to think of a better test and update later.

[begin page 16]

> Quote:
>
> Originally Posted by stupeT
>
> Just a guess: Use one EQ to cut and the same settings (freq and Q) on a different EQ (with different alogorithm) to boost.

No, digital eqs are basically all the same, except for deliberate "vintage" distortions and/or stuff done around the nyquist frequency. It was my fault for suggesting a bad test. Even analog eqs would basically cancel out if you used identical but opposite eq curves. The phase distortions would be the same, just inverted.

I'll still try to think of a better test, but boost-then-cut is intrinsically self-defeating, distortions and all.

> Quote:
>
> Originally Posted by nerdfactormax
>
> Of the Eq's I tried, about half passed the null test. If you are really accurate you can almost get ReaFir to null.
>
> My reason for challenging the original hypothesis (apart from being fairly certain that the maths of it is correct) is that I often feel like phase is the boogey-man: no-one can hear it or see it, but still fear its presence.
>
> All you need is ears
>
> If you are using ReaEq or something else that shows you the phase effects and you are worried that your narrow boost or cut is ruining the phase...
>
> If it sounds good, it sounds good
>
> Don't be worried that someone playing back your song in their ipod is somehow going to pick up on this phase problem and think ill of you. Unless you have bad monitoring/room/hearing
>
> For the mathematically inclined, here's another thought to challenge preconceived ideas
> http://forum.cockos.com/showthread.php?t=30576

Phase is one of those over-rated things that people misuse in message-board arguments to "prove" that one thing sounds better than another. It's not entirely correct to say that "nobody can hear it", because it is real, and is absolutely perceptible as degradation by human ears. I've been trying to think of a good "hard" subjective test, but it's hard to separate phase from other distortions such as delay and eq (which is really just a form of delay, speaking in a physics-of-sound sense).

Sometimes in movies or TV shows, or even radio broadcasts, you can hear speech that has been heavily eq'd, and it's hard to make out what the people are saying even though the "clarity" frequencies are hyped to the max. There's a sort of fake, boxy, veiled, graininess, sometimes combined with exaggerated essiness or stretched-out plosives. The dialogue has been made to sound dramatic, big, or "punchy" rather than clear. A lot of the distortion comes from stuff like compression and embiggening delay effects, but a lot of it also comes from sharp, extreme eq filters. This is sometimes most prominent in "live", on-scene, non-voiceover dialogue, where the sound engineer has isolated and exaggerated the speech frequencies in relation to the background sounds. It's like you can never get the tone and volume settings right to understand the dialogue.

All that said, the main point about "bad phase" being no different from "bad sound" is 100% correct. And there is altogether way too much worrying about invisible gremlins ruining one's audio, usually on the part of people who have other obvious and glaring uncorrected flaws.

Phase is just a part of sound. Equipment manufacturers absolutely need to think about it, but most home recordists should probably not, except in the sense of when something starts to sound bad, back up and try a different approach.

I'm still trying to think of a good, isolated way to illustrate what phase distortion sounds like, without including obviously whooshy "flanger" and "phaser" effects, and without muddying the waters by forcing un-compensated delays or eq into the mix.

The single most obvious and easy-to-explain example of where phase distortion occurs is in full-range speakers. If you imagine a single speaker that is producing both a 50Hz and a 10,000Hz tone, then the driver is moving back forth, creating pressure changes forwards and backwards 50 times per second for the 50Hz tone, and it is also producing the 10k tone. In effect, it is like the "10k" speaker is moving back and forth, closer to and further from your ears 50 times per second. Which is obviously going to screw up the pure arrival of sound-pressure changes arriving at your eardrum every 10-thousandth of a second, because those pressure changes are also being modulated on a much slower curve that is shifting their arrival at your ears. IF you picked up a speaker and pushed it back and forth really fast, you'd hear an obvious "wowing" or "whooshing" phase effect, but that's not really the same sound-wise, although the principle is the same.

As I said, I'll try to think of a good AB test, but in the meantime, just think of phase a part of sound, and focus on getting good sound.

> Quote:
>
> Originally Posted by Marah Mag
>
> I was just about to say more or less the same thing. If it's true that (as per yep above) that "tilting your head to one side or another while listening, or moving the mic off-axis, or listening with speaker at an angle, reflections in the room" causes phase smear... and it is... and that phase is "just an element of sound"... and it is... then why be concerned about it?
>
> If you have a sound, and it sounds good and is appropriate and works in your mix design, then that's what it is. Why does it matter if it's "phase smeared?" And what is it phase smeared relative to? Another, theoretically more accurate or perfect version of itself? If that non-phase smeared sound is better for your purposes, then shouldn't that sound be used instead?
>
> What am I missing?
>
> A more general observation/question.... If all aspects of the listening environment will affect how things sound, including the tilt of your head, your angle towards the speakers, the number of people in the room, whether your hair is falling in front of your ears and deflecting sound or pushed behind your ears thus changing the angle of your outer ear...
>
> then doesn't it follow that, even though there might generally be a consensus on what sounds better or worse... by these standards no two people can possibly hear the same mix the same way... and that there's actually a fairly wide margin within which you can design a mix that doesn't require a listener to be restrained in position just so between the ideal monitors in the ideal room, unable to move without wrecking the aesthetic? Rather than being haunted by all these subtleties, shouldn't they be liberating? Again, what am I missing?

You're not missing anything! Knowing "when to stop worrying" is one of those general maturity/wisdom questions that is hard to boil down into a precise algorithm or set of "rules." What is beautiful? What is witty? What is sexy? What is obscene? What is gross? What is exciting? What is boring?

We can chop these concepts up into a million pieces and write endless pages of analysis and "rules" and never actually nail it to the wall. But that doesn't mean that the terms are meaningless. Autumn in New England is always beautiful. Pretty much anyone can spot the wittiest person in the room, even if we don't know how to duplicate it with a computer algorithm. Same with boring people.

Sound quality is actually pretty easy, as these things go. And it's pretty easy to get a consensus opinion. Ask your family and friends which of your songs are the best, and some of them will say it's all genius, others will find fault with anything. But ask them which recordings sound the most "professional" or "high quality" and I bet you get near-unanimous consensus. Either that or no strong opinions.

Man, I haven't been getting much sleep lately, but I can't believe I didn't think of this instantly – try the above boost-then-cut test with regular eq first, then a "linear phase" eq for the complimentary cuts. I haven't tried it, but it should work.

Note that "linear phase" eq's are still not truly "phase free" – nothing is in audio, and then introduce their own problems of predelay, but it should give a sense of what phase sounds like...

> Quote:
>
> Originally Posted by flocentblack
>
> Hi everyone, been lurking for a while (following this thread), felt I should chime in here...
>
> You will not hear phase shift, unless the phase shifted signal is mixed with the original signal.
>
> The proof is in this article:
>
> http://www.ethanwiner.com/phase.html
>
> I used to get hung up on the idea of eq's causing phase shift (which it turns out is not audible. The real question is "does this sound better with EQ or not".
>
> Everything we hear in the world we live in undergoes some phase shift, yet we cant detect it, unless we are hearing the same sound from two different sources, one of which has a shift in phase.

The whole thing with eq is that phase shift IS mixed with the original. That's what eq IS. And it's exactly the same thing as what happens when you tilt your head or move a speaker and the freq response changes.

Phase is an intrinsic part of sound, and as I said near the beginning of this whole thing "bad phase" is no different from plain old "bad sound." It's one thing to say you shouldn't worry about a particular phenomenon, it's another altogether to not explain it. All my "phase"-related posts are just trying to explain what it is an how it works, not to make any judgments about good or bad.

> Quote:
>
> Originally Posted by GULCH OF ROT View Post
>
> Hi all
>
> ...would it be bad to take my vocal signal through the Digital Vocal effects processor I use to do live shows...

No, not at all. Replicating your live show is never bad, assuming your live show is good. When asking for advice on recording (as opposed to music) forums, it is good to provide direct links to the material in question. Just referring everyone to Myspace links in your sig is a little annoying.

That said, the material on the myspace page for Gulch Of Rot sounds okay for what it is.

The guitar sounds extremely fizzy and weak, but it's hard to avoid that with such extreme gain settings. “Punch” and “dynamics” intrinsically require a difference between quiet and loud. If the guitar is just as loud when it's in-between notes, then it turns into a wall of fizz. To put it bluntly, the vocals sound like a joke. I hate to say that, because they are clearly capable within the genre, and frankly, not everyone can sing like that, but cookie monster/gollum is getting a little tired.

If I haven't already pissed you off to the point of no longer listening to me, I would encourage you to check out some old soul music: If you listen to the song “Disco Inferno”, there is a similar kind of vocal “fire” on some of the high notes, specifically on the first syllable of the word “satisfaction.” Similar stuff can be heard on Sam and Dave's “Soul Man”, The Temptation's “Ain't to Proud to Beg”, and John Lee Hooker's “Boom Boom.” If you need a more metal example, then you could do worse than to check out Blind Guardian's “Imaginations From the Other Side” album, which combines massive “fire” vocals with real melody that transcends genre and era. That “fire” in the voice that lends drama and power to emphasis notes turns a little silly when it is used on every syllable. Similarly, a guitar that is always loud is ultimately indistinguishable from a guitar that is always quiet.

The difference between an organ or chorus and a drum is that an organ/chorus produces a sustained, consistent sound, whereas a drum produces a sharp, short, punchy sound. If everything is high-gain, constant volume, then everything sounds quiet unless you're playing to a very specific audience that always cranks the volume. And I daresay the audience for steady-state death metal is small and growing smaller.

It frankly sounds silly and cartoon-ish. Count Grishnackh is no longer scary, just a pathetic example of nerd rage compared with, say, late-90s hip-hop, which has both sonic impact and a genuine sense of implied visceral violence, based not on abstract metal coolness, but on real survival instinct.

My job as an audio engineer is clearly not to help people kill each other. But good art always requires stark drama, and one of the beauties of intense art is the catharsis that we never really achieve in real life. The soldiers who went to die at Thermopylae, the unimaginably dedicated lovers of ,50s pop songs, these are people who had a purity of purpose that we can all envy, even if none of us are willing to follow in their footsteps.

Imagination allows us to be heroes in a classical sense. Drama, excitement, brutishness and sexual tension have no genre. It's all about tension and release – no release is the same as no tension. A hummingbird and a chainsaw are only differentiated by playback volume.

> Quote:
>
> Originally Posted by GULCH OF ROT
>
> Yep and Mag
>
> I do not understand stand what you mean by putting a direct link to the songs but I will look into it...

Just anything that links directly to the song in question, even if links to a myspace page player or whatever. Something so that the listener isn't being re-directed to another page to hunt for the song in question.

To expand on my earlier response, maybe in a more helpful way, very heavy nu-metal-ish stuff is possibly the most difficult kind of music to engineer in a satisfying way. It's not very hard to get an "accurate" recording of a four-or five-piece rock combo, but what works at 110dB live is often vastly less satisfying at TV volume. There is no way to achieve the gigantic, visceral crush of sound without big SPL levels. And this is a very real problem when you're doing music that is dependent upon sounding louder than hell.

With a choir or a chamber orchestra or jazz ensemble or a folk duo you can often just stick a reference mic in front of the band and simply capture what they actually sound like – that could be the album, right there. You could do the same with a metal band, but very often it will sound pretty bad, unless you're playing back at bone-crushing volume. A flat-line saturated sound such as a full-on metal act turns into steady-state fizz when played back at sane living-room levels. Instead of sounding powerful and ferocious, it sounds weak, buzzy, and a little silly. Which exactly the opposite of what you want from heavy guitar rock.

There may be a saving grace with doing niche music made for specialist genre fans, in that they might be expected to always crank the volume, and to have good playback systems, and they might be somewhat inured to and more forgiving of the realities of recorded music in the sub-genre. And if that's the basis of comparison that you care about, then maybe how it sounds in a 300-watt car stereo is all that matters.

I don't know how far down this road anyone wants to go, because it really does start raise artistic questions of what constitutes musical power and impact, but the fact is that the furious thunder of double-kick mayhem, massively over-driven mesa-boogies, and ferocious gutteral vocals turns into mushy, papery fizz at non-deafening sound levels. And played side-by-side with some geriatric blues-based riff-rock that has space between notes and that allows enough milliseconds for individual kick drum hits to develop into actual thumps, it tends to sound even less powerful.

I over-stepped the bounds a little in my last post, and verged into critiquing your musical choices and performance style, and I apologize for that. You're working in a difficult genre – both difficult to play, and very difficult to record, and your band and your recording are both very capable.

> Quote:
>
> Originally Posted by DerMetzgermeister
>
> ...I appreciate any comments you have, about the mix, the arrangement, the performance.. anything.
>
> Thanks.
>
> Http://stashbox.org/523480/Licantropo.mp3

First impressions: I do not think you're getting precisely the sounds you think you're getting, particularly with the guitars. The high-gain stuff is begging for delay or multi-tracking. And maybe a little less gain. Maybe a lot less. And the low-gain stuff might benefit from a little more tubey fire. A little distortion on "clean", melodic guitars can really make the harmonics sing and snarl. Similarly, aggressive cleaner-tone playing is vastly under-rated.

The sound quality is decent, but the vocals sound a little hesitant/unrehearsed. Kind of a clenched, nervous, "hunched shoulders" sound in the "clean" parts. "Fire" parts might be over-doing it a little, and might benefit from a little more pitch and a little less "gruff." There might also be some pitch issues somewhere, either in the vocal or in the accompaniment.

Mix thoughts: Drums and bass are way too quiet, IMO. Might want to punch up the drums with some distortion/exciter/smacky compression. Vocals too much proximity effect, inconsistent dynamics on the "clean" parts. The processing on the vocals is a little weird, and sounds a bit like it's trying to alter the performance, especially in the "fire" parts.

If you're open to arrangement suggestions, you might consider more internal contrast among the song sections, e.g. trying the melancholy type-O-negative-style vocals over a tougher "space marines" kind of backing track, and giving the gruff "power vocals" a little more breathing space. Contrast is the key to creating that kind of cinematic drama and excitement. Everything spacey and vague sounds no more or less boring than everything overloaded and roaring.

It's not bad, but it still has a "bedroom demo" vibe of an incompletely-realized musical vision coupled with somewhat disconnected sonic elements that seem to have been worked out in isolation. I would recommend really focusing on building a foundation from drums and bass that "vibes" right, both in rehearsal and as the first step in building a mix. If you have a good rythm section, then you can put almost anything on top of it and it will sound good. But if you start with the "showcase" elements (guitar and vocals), it's a bit like trying to build a car from the paint down.

> Quote:
>
> Originally Posted by onewayout
>
> Glad to see you back Yep....Can you help me on getting this sound on the modern rock guitar mix? I have very strong guitar tracks and I can clearly hear there is something I'm missing when getting this vibe and mix.....Here is the song I'm mixing...
>
> http://stashbox.org/manage_file/5229...%202%20MP3.mp3

Broken link, but I think I got it. In stashbox, try using the "forum code" link:

http://stashbox.org/522962/Rush%20Hour%202%20MP3.mp3

That said, the mix sounds great. The guitars in particular show judicious gain and exactly the right "size." The drums sound great.

Are there going to be any vocals?

Foundation of the song

To expand on some of the above replies, the importance of starting from a good "foundation" cannot be over-stated. Usually this is drums and bass, but it might be anything.

The foundation is the sonic elements that the audience "feels" more than hears. It's the parts of a mix that bypass the ears and communicate directly with the hips, the lower spine, the hairs on the back of the neck, and the subconscious spirit/psyche.

Do you want people to get up and dance when they hear this song? Do you want them to get that mad, headbanging adrenaline rush? Do you want them to slip into a hypnotic, contemplative state? Do you want a dreamlike sense of remembrance and nostalgia? Do you want their hearts to swell with inspiration, patriotism, parental love, or pride? The songs "yesterday" and "back in black" both have the same number of syllables in the title/chorus, and could be sung to the melody of the other. But it wouldn't work. You don't have to like either of those songs, or even remember what the lyrics are (I don't), but pretty much anyone who has heard them will, I think, know what I'm talking about when I say that each has a distinctive "vibe." Even if I just heard the backing part played on a steel drum or hurdy-gurdy, I'd instantly recognize not so much the notes as the "vibe." It's probably unrealistic to expect every song to be as iconic and immediately evocative as the two examples cited above, but they're not all that unusual, they just happen to be well-known songs whose lyrics would sound preposterous if reversed, picked off the top of my head. That "vibe" is a bit like warm sand underfoot, or the smell of a wood fire on a cold night, or the experience of walking indoors from bright sunlight. You only need a reminder, and it conjures up a visceral experience, even if you can't precisely remember the sensory aspects, you remember what it "feels like." The beauty of music is that it allows for very specific and complex evocations, more than can be put into words, more than can be captured in a picture. A bit of music can convey the same sensation as looking across the car at a lover and knowing from the precise angle of her jawline that something is keeping you apart, but not knowing what. It can make this moment feel like Friday afternoon before a long weekend in summer. It can provide catharsis and a sense of release that we never really get in a life of pressures that just continue. It can convey a glimpse of the divine or it can be sexier than sex itself. It can provide a therapeutic purging of the rage and pain of daily sufferings by working through intensified versions of them. It can also just be a silly and playful experiment in sensation. At its best, music puts the listener exactly in the same state of mind and being as the composer/performer, and achieves a supreme human connection that is perfect and valuable in and of itself, regardless of the message conveyed.

I think anyone who loves music has had the experience of hearing a piece of music, and not necessarily remembering the song, but still remembering and wanting to re-experience the feeling it created. That doesn't come from tube preamps and million-dollar drum rooms and perfect reverbs. Those things can help, by making sure the conveyance of the message doesn't get in the way, and by providing just the right candlelight and wrapping paper or whatever, but they don't equate to saying something worth expressing in the first place. And done badly, over-producing and over-working a sentiment can strangle it.

It's good to give performances and to make records that sound "professional." But it's more important to convey something meaningful. Ideally, you'll do both. But be wary of the tendency to over-focus on the desire to sound and perform like a "real" or "professional" artist. The elemental, subconscious experiences that you can convey are vastly more important than technical perfection or "professionalism." Showing off your stylistic range can be fun, if you have it, and it can also be a great way to grab the audience's attention from time to time. But it also gets boring fast when done constantly.

I'm a great fan of the game of baseball, and like every baseball fan, I revel in the constant psychological and athletic tension of the grueling 162-game season of a sport that never lets up, where nobody ever comes close to going undefeated, and where the best of the best are only successful a few percentage points more often than people relegated back to the minor leagues. A lot of people think baseball is boring. It certainly lacks the dramatic intensity of rugby or gridiron football, and it is vastly slower-paced than soccer, hockey, or basketball. But at its most intense, baseball has the finest examples of explosive athleticism in sport, and its slowest and most grueling, baseball has an intensity and a precariousness matched only by the best suspense movies, if you understand the game.

My point with the above digression is to set up the following statement: I can't think of anything more boring to watch than a tee-off home-run derby. What makes baseball exciting is not the raw ability to hit a ball off a tee 500 feet. That's the athletic equivalent of music made to impress friends and parents.

The glory of the home-run, or even of the bloop infield single, is the ability to hit a deceptive 95 mph pitch thrown by someone with a nigh-superhuman ability to make the ball move in ways that seem to defy physics, and to hit the ball in such a way that nobody can get to it before you can run to first. The pitcher must find a way to deliver the ball inside a narrow window such that the sharpest eyes and fastest reflexes in the world don't know where it's going to cross the strike zone, and must do it with a rapid delivery that prevents stolen bases and that forces the batter to evaluate the motion of the pitch and commit to a swing when the pitch is still 30 feet or more from the plate.

Watching grown men hit balls with sticks is not, in and of itself, a worthwhile pastime, even if they can hit them very far. Certainly not something worth watching for three hours a day, six months of the year. Similarly, a three-octave range alone does not make a good singer, and fast fingers do not make a good player. It is the human connection, the inner physical/psychic intelligence, that makes these, my two favorite spectator sports, worthwhile.

You can sit in your studio and launch home-runs all day, and that's fine, if that's what you're about. Your parents and friends and other musicians might be wowed by your prowess. But if you want to actually play the game, then you need to go up against the greatest competitor of them all, which is the human spirit. And the only way to do that is in real-time, against the blindingly fast and deceptively twisting pitches of thought and emotion. You have to hit that vibe in the sweet spot and send it for a ride.

[begin page 17]

> Quote:
>
> Originally Posted by DerMetzgermeister
>
> Thank you very much.
>
> I do agree with everything you said but I have some doubts. Care to explain? It has compression, eq and reverb, I think some delay too.. what struck you as weird?

Only that the vocal sounds a little out-of-place, kind of "floating on top" of the song. It's hard to tell what the processing "should be", but the vocal sounded a bit like it's not on the same stage as the rest of the band, if that makes any sense. This is often a symptom of a guitar-centric mix.

If I had access to the raw tracks, my first approach would be start over with a good mix of the drums and bass, using whatever dynamics and eq were necessary to get them to "sit" together, geting them as close to "finished song" as I could, especially since the vox and guitar are both on the atmospheric side. Then I would pull up the vocal, and get that to "fit" within the drum and bass mix. Then use the guitars to fill out the sound wherever.

Without having a clear sense of how much can be done with the drums and bass, it's hard to say for sure where I would start with the vocals, but it would almost certainly involve some broad lower-midrange cut and some unified reverb with the guitar and drums. I would almost certainly try to get a "bigger" sound from the drums, with a more pumping, smacking eq. On the version I heard on my laptop, it's very hard to tell what the bass sounds like, but there may be some pitch conflicts between the bass and vocals (even if everything is "in tune"). Bear in mind that, traditionally, bass is the loudest instrument onstage, although usually the least present. Weak bass is not usually a hallmark of a good mix.

> Quote:
>
> I don't get the space marines reference.

You know how, in science-fiction movies, when there is a gruesome special-effects battle going on, they always seem to have some kind of blood-pumping metal/industrial/symphonic backing track? I always think of stuff like Rammstein or White Zombie or Drowning Pool as "Space Marines" music.

More to the point, what I meant to convey is that sometimes the best picture of melancholy comes from a juxtaposition of vital, full-blooded, and energetic elements against a striking patch of emptiness and ennui. The way that maybe a photograph of a single flower in full-bloom against a decrepit old farmhouse is more evocative than rotten weeds in front of the same. Or that a skull or a rusty pickaxe in front of a scene of gray decay is sometimes not as effective as the same in the midst of a lush garden or a party.

I am specifically reminded of a project from a few years ago where the band had written what I thought was a very cool song, but one that was also very oppressive and downbeat in tone and tenor, and was also long, almost six minutes. It was also by far their most dramatic, hard-hitting, and lyrically accomplished song. On first pass, it seemed like the obvious "single" and their signature track.

But the mix just didn't fit together right, and it seemed to sound sort of dreary and boring no matter what we did... I brought in a female session singer I knew who had a very soulful R&B-style delivery, thinking that she might sort of liven it up in a Lords of Acid or KMFDM kind of way, and after a few mediocre takes, she took off her headphones and said, "you know what? It's just too heavy. It's all thinking, there's nothing to sing to." And she left.

And she was right. It was immediately obvious to me as soon as she said it. It wasn't even a song, per se, it was an essay with a backing track. I sat down with the band the next day and told them: There was nothing to sing along to, certainly nothing to dance to, nobody would ever play this song at a wedding or a party, or even a funeral. Nobody would ever pop it in the car after a hard day at work, it was just oppressive and downbeat and grim and unpleasant to listen to, although still very good and very accomplished both lyrically and musically. It was a work of art for considered living-room listening, not a pop song.

I still thought it was a worthwhile artistic effort, but grim, dreary, and intellectualized were its nature. There was no way to turn it into a musically exciting mix without re-working the song itself. The band decided to keep it as-is, which I agreed with given the context, but it was a lesson both to them and to me. To the band, it was a lesson in the limitations of the studio process.

To me, it was a lesson in arrangement. If I could have gone back to the very beginning, to pre-production, when the song was still being fleshed out, I would have suggested inserting some major chords, or at least heroic fourths, or key changes at certain points. Not because the song was bad, and certainly not to turn it into a chirpy bubblegum ditty, but to break up the relentless, desolate intellectualism of the thing.

To make the negative aspects stand out in sharper relief, instead of drowning in their own churning grind of exhausting nihilism. To set that rusty pickaxe or rotten old farmhouse against something alive and vital, to show the contrast, the way that a love story about a farmhand is more romantic than one about a dilettante, the way a heroic tragedy depends upon the hero achieving success and then squandering and ruining it to have the same pathos.

The lesson to me was that unbroken and uncontested negativity could be as bland and as blank as the vapid and chipper cheerfulness of top-40 teenybopper pop. It's all about contrast. I keep learning that lesson, in different ways.

> Quote:
>
> One last question. What do you think about the EQ?

I'm not even sure what to say with the current mix. It seems okay, if a mix with disappearing drums and bass can be said to have good eq. And again, what YOU think is vastly more important than what I think. I might have all the wrong preconceptions. I certainly have different tastes than a lot of music fans do, which is easy enough to gather simply from looking at my CD collection, which looks nothing like a list of top-sellers.

I can't tell you what sound to go for, all I can do is offer advice on how to get there quicker. So it's hard for me to evaluate the route you took without knowing where you wanted to end up. It would be much easier for me to offer specific advice if I knew what YOU think is wrong with the mix, assuming something is. (If you think your mix is just fine, then please don't post here asking for reviews!)

[Stopped 5-25-09, Thread #641]

> Quote:
>
> Originally Posted by thalweg
>
> ...
>
> P.S
>
> Not to digress from the technical aspects of this thread, and maybe it requires a new one, but I would be grateful if you could perhaps spill some brief thoughts on song writing. Whats your approach? Do you have a specific instrument you like to write with? What about from other notable musicians you‘ve worked with..any experiences to share there?
>
> Cheers
>
> Thal

Yeah, that‘s probably a topic for another thread.

That said, it‘s not specifically songwriting advice, but the best piece of general artistic advice I ever got was from jazzman Bill Dixon, when I was nineteen and full of piss and vinegar and wanting to set the world on fire.

It was something to the effect of: “Don‘t try to be innovative. Don‘t try to be anything other than good. If you are an innovator, then you will innovate whether you try to or not, and if you‘re not an innovator, then you can still be a very good musician without making ass of yourself trying to be something you‘re not.“

Kind of a more pointed version of the old saw: be yourself, and don‘t try to be something you‘re not. A little surprising but also more forceful coming from a relentlessly innovative and cutting-edge guy like Bill Dixon. A similar and related note comes from Wynton Marsalis, who said something like: “There is no evolution in music. We‘re never going to 'get beyond' Bach or Coltrane. We‘re all just adding to the mix, and there is no need to be 'better than' to contribute something worthwhile.” Both of them speak to the tendency to over-intellectualize, which might or might not be applicable to anyone else, but it certainly was (and probably still is) relevant to me. It‘s really important to just play, in every sense of the word, and to make sure that the primary focus of your musical life is to have fun. Because if you‘re not enjoying the creative process, then how is anyone else going to enjoy the result?

Make time to get away from the computer, clear your head of tips and tricks and theory and structural considerations, and just play the kind of music you want to play for its own sake and without outcome objectives or goals, even if it‘s derivative or the same thing you always play. This kind of raw, pure, unstructured creative state is the soil and fertilizer for imagination. It‘s probably why you started playing music in the first place. If it‘s not the center of your musical life, then how can you not expect your creativity and inspiration to dry up? If music is the product of a methodical series of chores and academic exercises, made by a glassy-eyed technician hunched over a computer, then it‘s going to show in the results.

A skill producer or arranger might listen to a good song and say, we could break up the monotony by adding a bridge after the second chorus, or by moving the key of the verse down a fourth, or we could add some tambourine or percussion to liven up the last section, or thin out the guitar riff until the pre-chorus, and all that kind of stuff. And that stuff can push a good song over the top, but I don‘t think it can make a bad song good, and I certainly don‘t know of any way to write anything other than outright album filler with such techniques.

The writing has to come from a place of inspiration, I think. And the best technique I know of to fire inspiration is just to play, and to open up your mind or spirit or whatever. If you sit down right now and start playing one note over and over again, I guarantee it will not last more than a few minutes before you start getting ideas and playing something more interesting. Some of them might be similar to ideas you‘ve already played, but again, if you keep at it, new variations and juxtapositions will start to present themselves. There is something magical in the resonances of musical notes, a kind of "music of the spheres" that is already in the tones, and in the "beats" of the harmonics. It will guide you, if you allow yourself to be led.

And in a world where 90% of all songs on the radio are structurally the same, and where half of them even use the same chord progressions, a song doesn't have to be groundbreaking to be good and worthwhile, it just has to be inspired.

> Quote:
>
> Originally Posted by DerMetzgermeister
>
> ...Vocals pitchy, hesitant, etc-> Kidnap some close relative of the singer and send him sliced body parts until he get his shit together and work in his technique for once... I'm all out of ideas there...

All singers should practice. When a musician is learning a song or a part, there are at least three distinct "stages" of learning.

Stage one is just getting the basic mechanics down, where the musician is still making mistakes, missing changes, requires a pause between different sections, hitting wrong notes, etc.

Stage two is when the musician has learned the part and can generally play it all the way through without having to stop and without making many obvious "mistakes." They may not get it exactly right every single time, but they can technically go through the mechanical exercise of playing all the notes in sequence and more or less in time.

Stage three is when the musician has really and fully "got it," and can rip through the whole thing, with muscle-memory on auto-pilot, just hearing and flowing, and playing the song expressively and creatively, not just in a mechanically accurate way, but it a full-blooded musical way. The musician might have it down so that they can improvise fluidly and musically, or they might simply play the part as written, but with more ability to control the sound, texture, dynamics, and "feel." I think anyone who plays an instrument is familiar with the difference,better than I can describe it.

Better, more accomplished musicians tend to get to stage 3 a lot faster, while beginners often have a hard time breaking through the "student recital" stage 2, but everyone can eventually get there with practice.

Now, the trick with singers, especially with pop/rock/hip-hop singers who are not usually singing stuff that is exceptionally challenging in technical range/music/complex tonality sense, is that they tend to start out at stage 2 right from the get-go. Most people can basically sing through a typical rock song on the first pass without many egregious "mistakes," at least to whatever their current skill level is.

The above fact, plus the fact that practicing singing is often a little embarrassing, means that an awful lot of part-time vocalists never get past stage 2, because they rarely or never feel the need to just practice, beyond maybe singing in the shower or the car. Any singer who wants to break through to the next level, of delivering powerful, professional, emotionally and musically focused vocal performances HAS to practice.

Having a decent voice and basically hitting the right notes at the right times is no different than a student musician with a decent instrument who has just learned to plod through the mechanics of new piece. It's not "bad" or "wrong", per se, but it's not necessarily the kind of performance that people are going to pay money to spend their Friday nights listening to. This is probably the single most common problem with amateur bands — not BAD singers per se, just weak, un-polished singing. And no act can ever be better than the singer. When you go out to see local acts in a small club or bar, most of them, now matter how good the musicians, still sound kind of blah and uninspired, mostly because the singers are just kind of going through the motions. Either that, or the singers are trying to make up for it with sound and fury by yelling, posturing, or "acting" more than singing. They're not necessarily BAD, but they kind of tend to go in one ear and out the other.

But when that one band comes up with the singer who is actually on-point, who is confident and capable, who delivers a performance that is decisive and musical, who sings clearly and with controlled, appropriate dynamics without mumbling or overloading the mic with proximity effect or resorting to tuneless yelling, who nails the pitch effortlessly and decisively with a focused, controlled delivery, then THAT is the band most people are going to remember the next day, because that's the band that people are able to sing along with.

> Quote:
>
> Originally Posted by junioreq
>
> I can‘t thank you enough Yep! Not to hijack or anything. But I have to run out and havent read the previous posts.
>
> What about the land of“direct recording“. Example:
>
> We use Ezdrummer for drums. If you listen to Ezdrummer, the kick has a lot of room ambiance on it. Now, we can replace it with a kick using drumagog, or we can do several steps to make the kick seem tighter. The only trick I know really is using eq with slow attack, fast release, think there is something that can be done with an expander as well.
>
> Not sure if I really have a question, but a lot of us struggle with the tools we have like Ezdrummer – If we don‘t like the overhead sound, we kind of have to deal with it – And thats where I spend most of my time. Just getting the tools to sound right in the first place.
>
> Are there any tips, or go to methods and like smashing these square pegs in round holes and making what we have to work with – sound like what we want? Ya know, we can‘t just run up and put a different mic on kick.
>
> What are some ways to mold and clean up sounds?
>
> Hope this makes sense..
>
> ~Rob.

I think I‘m replying to text that was later deleted as“not relative” (not relevant?) but it certainly is relevant, in fact it cuts to the heart of the whole question.

Everybody always has to “deal with it” where “it” means the sounds available to you. The dividing line between “sound” and “music“, if it exists, is something for philosophers to debate. The sound and the music are the same, especially in modern non-classical music. Ever since the electric guitar first allowed talented but technically mediocre musicians to express ideas almost entirely through sonic texture that could not be conveyed on a score sheet,“sound” has been a foundational element of popular music. Maybe not in academic texts and music theory courses, but certainly in the real world of everyday listening and enjoyment.

Singers used to have to be“good” to become famous. Now, partly or entirely tuneless “vocalists” can get by entirely on stylistic and sonic delivery. And so on with any instrument. Whether this is good or bad is almost beside the point, because it is certainly true. Good musicians are still good musicians, but sometimes bad musicians with a good sound can now be good musicians, with a whole lot of stuff in-between.

Modern songsmithing is very often a matter not simply of writing the music and then getting instruments to play it, but building the music itself upon the sounds. Whether it‘s samples or real instruments, a boomy, rickety-sounding vintage drum kit with a kick drum that sounds an actual note and that rings out on every hit long enough so that you can step out for a smoke and come back before the drum hit decays is never going to sound the same as a modern Tama or DW metal kit where the kick is a sharp click with a rapid thump that dies in a nanosecond. And neither is going to sound the same as an 808 drum machine.

None of these are good nor bad, they‘re just different sounds. You could stomp on a cardboard box. I think Benny Benjamin claimed that was the best kick drum sound he ever got. And he was a guy that played basically one of two drum beats on every song he ever played on, but probably recorded more number one singles than any other drummer in history. And somehow, all those records sound different and alive, even with the same nominal beat. It wasn‘t in the notation, it was in the sound he got and the way he worked the decay and the texture of the spaces in-between.

You can start with a drum beat or pattern, and try to find a sound that fits it, or you can start with a sound, and then construct a beat around it, or you can actually function like a musician and create in real-time, in response to the sound and everything else. Some of the best amateur drummers working today play pickle buckets and pie pans in subway stations (I‘m not kidding). They get a sound and a vibe that blows away half of the top 40.

It‘s not about the quality of the samples, per se, it‘s about how you create a vibe and a flow from the sounds that are available to you.

If you work with sampled drums, the first rule is to forget about putting any effort into trying to make them sound “realistic.” That is a stupid waste of time, especially in a world where half the top 40 is obviously fake drums, and where half of the records with real drums are trying to find ways to sound fake. Nobody cares whether you use real drums, and certainly nobody cares about your skill in making fake drums sound real. People care about the music. Even drummers or audio snobs who bitch about drum machines still get down when the music compels them, in between complaining about fake music.

> Quote:
>
> Originally Posted by thalweg
>
> ...Earlier you talked about/quoted someone as saying music having already exhausted past innovative efforts and we‘re relying now on variations of whats already been done. Do you really believe this to be the case?

If you‘re talking about the Dixon/Marsalis quotes, I don‘t think they implied anything like the characterization that you‘ve given them. I think Bill Dixon in particular would object to the notion that we have "exhausted past innovative efforts" and that we are now just constrained to "what‘s already been done."

Neither point was quite analogous to "there‘s nothing new under the sun." Dixon‘s point was that you can be just plain good, and that being good is a good and worthwhile thing in and of itself, regardless of whether you‘re doing anything demonstrably "new" or "different." Bill Dixon is a legendary master of innovation and someone who creates music that exists as completely outside the constraints of conventional tonality and time-signatures as anyone alive.

Marsalis‘ quote, as I read it, was not meant to convey that there is nothing new to add, but instead that EVERYTHING is potentially valid, even just variations of stuff that has been done before. "There is no evolution in music" doesn‘t mean that there WAS evolution but that it has now stopped, only that stuff like Beethoven‘s ninth or "Love Supreme" will never be rendered "obsolete." The best music that currently exists will never be "bettered," nor has it ever been, for as far back as we have recorded music. Everything worthwhile is adding to the mix. The world of music moves outward and inward every day, but not necessarily "onward" in the sense of continual improvement.

To cite another favorite quote, this from Duke Ellington, "There are only two kinds of music: good music and bad music."

As to my own thoughts on the "state" of the art and business of music, I can‘t think of many things more boring and wanky to talk about, and I already fear that this thread has veered too far into "yep thinking about music."

Good recording practice has almost nothing to do with my taste in music. The artistry of music is what I love to experience, and to listen to. But the technical aspects of audio are much more interesting to talk about. IMO, writing about music is like dancing about architecture, or something like that. People who can‘t write, interviewing people who can‘t talk, for the benefit of people who can‘t read, and so on. I much prefer to talk about the technical details and let the big picture stuff speak for itself.

PS — I will add this:

There is a certain kind of musician who always "gets an A+" but who nobody really wants to listen to. There is no need to name names, we all know examples. The kind of people who say all the right things, who play all the right things, who know all the right stuff, who are playing music to impress other musicians but whose music is never going to be played at a wedding or at a good party or such stuff.

Nobody ever falls in love to their music or is compelled to break down and weep or to jump up and dance, it's music for music nerds. And that's fine, in and of itself.

But I think a lot of people who fell in love with music once upon a time one day when it DID break their heart, or when it bypassed their brain and took over their hips and compelled them to get out and shake their groove thing like a retarded epileptic, or when it took over their central nervous system and flooded their body with adrenaline and endorphins... I think a lot of musicians sometimes lose sight of that, and get caught up in trying to make music that would get an A+, instead of making the kind of music that made them love music in the first place.

> Quote:
>
> Originally Posted by thalweg
>
> Fair enough re: Music business...and the evolution of the art form. Completely understand...I was hoping that you may have had an alternative viewpoint. No sweat..
>
> Thanks again.

Well, here's the thing: there's not much to say that really matters, IMO. Music is useless. If I need a new refrigerator, I might want a $10,000 viking or sub-zero or whatever fancy thing, but I can still refrigerate my food with a $300 generic from Home Depot.

If I want to hear the Beatles or Mozart or whatever, I can hear them for the same price as anybody else. Nobody goes into a record store to buy a Tom Petty record and instead decides to buy some other band that's almost as good as Tom Petty because it's $3 cheaper.

The only "competitive advantage" in music is to be better, whatever that means. People in the developed world buy X number of records per year or whatever and they have to like your music better than the other music that they'd spend their money on. That's about all there is to it. And there is no way to tell anyone what anyone will like better. I don't personally believe that various theories about how to game the music business are very useful from an aspiring artist's point of view. All you can do is to be as good as you can be, and hope that people respond to it.

Having said all that, there are some 6 billion people in the world, last I checked. If you can do something that, say, one-tenth of one percent of people respond to, then you have a potential fan base of 6 million people. Which is a lot. If one percent of those people are willing to pay, say, $15 for a CD, then that's $900,000 potential revenue from a single album. And that's one percent of one tenth percent of all the people.

Unfortunately for musicians, we live in an era where the most active population of music fans (say, 16 to 35 year olds) often does not pay for music. They pay for iPods and high-speed internet connections and bigger hard drives and faster computers, but they expect the content itself to be free. On top of this, opportunities for live performances of popular music are rapidly diminishing in favor of Djs and jukeboxes.

To cite one anecdotal example, a personal acquaintance of mine went on a world tour as the opening act for NIN a few years ago. She spent a year playing in front of 100,000-capacity stadiums all over the world. She headlined side-shows that sold out more tickets than her band has ever sold records in the respective countries. She made a net income of $15,000 (not a typo) that year, most of it from selling a song to a Belgian fruit-preserves commercial.

The roadies made more, the venues made a fortune, the promoters, agents, and managers made money, but the music itself is regarded system-wide as a disposable, replaceable commodity. She came off tour and went back to work at a thrift shop to save up for the next tour. This is an act with multiple MTV videos, an act that has played Radio City Music Hall, an act that has been featured in Rolling Stone and Spin and so on... Millions of fans, but not many who are willing to pay for the actual content. They'll pay for the nightclub experience, for the iPod and for the internet, for the computer and backup hard drives, but not for the music.

The band is currently on hiatus, and almost certainly over unless a PBS special or some such ever decides to fork over some cash to reunite them. She is now trying to find a way to get a job that will pay stuff like health insurance and some kind of retirement savings. This is someone who has fans all over the world, looking for temp jobs. (If any intrepid googlers figure out who I'm talking about, please do not post the name of the band... these are friends who I am sure do not want their name popping up in this context from google searches.)

An example that has already been in the news, so I don't feel bad naming names, is the 80s teen diva "Tiffany" who was some years ago fired from a retail clerk job at Pier One imports in Nashville (a discount furniture and knick-knack store), because she was unable to perform the job duties due to being overwhelmed by fans. My friend Tim (also a musician) was the unfortunate supervisor who had to tell her that she was just too famous for the job.

A lot of people have the impression that if you‘ve ever been on TV, someone must have handed you a million dollars and a Ferrari, but it‘s not so. The people who pay for talent these days are not fans (who generally just pirate music, for good or for ill), but ads and movie soundtracks. The way to make money with music is selling music to pepsi commercials, no longer selling music to music fans, because music fans don‘t pay for music anymore.

In terms of "art", the most expensive artist at auction right now is Van Gogh, a guy who in his entire life sold only a single painting, and that to his brother who paid all his bills and put him up anyway.

> Quote:
>
> Originally Posted by PAPT
>
> ...
>
> If everyone stopped making commercial music tomorrow life would go on.
>
> If the garbage men stopped picking up garbage tomorrow we would have a huge problem on our hands instantly.
>
> We live in a society with very silly priorities.
>
> Make music for your enjoyment.

That‘s certainly a valid way to look at it. A case could also be made that talented and dedicated artists, given space to pursue their own artistic visions, contribute far more towards what makes life worth living than (theoretically) more replaceable trash collectors do.

But in any case, what philosophical conclusions to draw from the fact that someone can provide meaning and artistic joy to millions of people, without recompense equivalent to a garbage collector, is far beyond the scope of this thread.

My point was meant to be a practical one, not a philosophical one. If music is without value, then we might wonder why people spend real time, money, and effort on piracy and media players, but even if we assume that the content is valueless, and that music from pepsi commercials is just as good as anything else, and that no musician ever deserves to be supported for her creative contributions, the practical realities are the same.

[Stopped 6-6-09, Thread #661]

Being an independent artist

A flip-side to the fairly cynical outlook above: There has never, ever been a better time to be a **very** independent artist. A dedicated hobbyist with a spouse, kids, and a day job, who makes music in their few spare hours per week, can now reach an audience to a degree that would have been nigh-impossible 20 years ago when mailing cassettes was the only way for a non-touring, no-airplay musician to be heard (remarkably, that did happen — there were people who would actually lick stamps and send $5 bills to musicians who would then mail out a cassette.

And some of them were really good. One of my all-time favorite records is "It Came From Jay's Garage", just a cassette of a bunch of bands who were friends with a guy named Jay).

What is evaporating is opportunities for mid-tier artists to get by as musicians. As live music venues have dried up, they have increasingly turned into "play for free" if not outright "pay to play." I'm not talking about getting rich and famous, I'm just talking about the ability to earn enough money for gas, Ramen noodles, and sterno cans to cook on in order to make it to the next gig in a live-in-a-van tour. Enough to pay for a shared rehearsal space and a couch to crash on. Which is precisely how most of the best bands developed their talent and their sound.

And all the stuff I've been talking about, about crafting and refining good music in the real world, and THEN recording it well... all that starts to fall by the wayside the more that musicians lose the real experience of just playing and working out material night after night. It becomes more and more about disposable drum loops and one-hit albums with a bunch of filler, because it's very hard to take more than one or two ideas to fruition when you have to do everything yourself, all without bothering the neighbors and in between doing the laundry and paying the bills after work.

A dirty little secret of the modern music scene (in the US, anyway) is the fact that almost all "working" musicians of a sub-mega-star level are now people who have some kind of family money or support network. Either that, or they are people working in some kind of lucrative part-timish field, often computer-related, that allows them to devote a lot of time and resources to a fairly time-and money-intensive hobby.

And this, to my way of thinking, is a serious problem. Not that "rich" kids can't create good music, but when there is no path left for the talented and dedicated kids who can only afford a bass and an amp, who are willing to put their all into it and who have genuine talent, but who don't have parents to pay the rent, who have to get up in the morning just to live at poverty-level, and who, no matter how talented, cannot get a paying gig (I mean, even $50, four times a week), then we have excluded probably 70% of the population. Including some of the demographics that have traditionally been the most fecund and talented...

It is a common complaint among musicians everywhere that the top 40 is soulless and vapid. But if you live in the US or the UK, spend a couple hours listening to pop stations in continental Europe or Japan, where post-rock music is more traditionally the domain of hipsters and club kids, and you will be begging for the lyrical depth and soulful intensity of Pink or N*SYNC. I wish I was kidding.

Maybe the internet is bringing us to a place something more like music was pre-radio, and pre-recording, where most people's exposure to music was from small-time, after-work pub and folk musicians, except where the "local" is global, just intensely genre-fied. And maybe that will lead to a place of small networks of connected fans and musicians with closer relationships, or something. Maybe the whole notion of "professional" popular music was a transient artifact of the last 70 years or so, an era where recording and transmission was too expensive for amateurs, but cheap enough for people who did not merit commissioned symphony performances. Maybe the new royalty is mega-corporations who sponsor pop mega-stars instead of symphonic composers, and the new corner pub musician is the after-work cyber-hobbyist who plays a sequencer instead of a fiddle. I don't know. But it doesn't seem to me like most people are using the internet to connect with smaller artists.

Whether piracy is right or wrong seems increasingly irrelevant. My personal belief is that it always wrong to take the product of someone else's work without their permission, but it's pretty clear that as long as people CAN get music for free, they'll take it. And the modern music "industry" is almost entirely piracy. For every record sold, there are probably a thousand pirated copies, and not just among poor or casual music fans, there are plenty of people with expensive computers and ipods and high-speed internet and cable TV and so on who listen more or less exclusively to pirated music. The savviest music hipsters are often the most prolific pirates, with thousands of dollars invested in computers and sound-systems and mp3 players and whatever, and zero spent on music. Whether it's right or wrong, it seems, at this point, plainly inescapable.

In the meantime, my acquaintance from above is looking for retail or data-entry work, if anyone knows of an opening. Unfortunately "millions of fans worldwide" doesn't qualify one to operate a cash register, apparently...

> Quote:
>
> Originally Posted by stupeT
>
> Sorry, I still hope you were just kidding.
>
> #1: Club stuff, since Disco, Elektro, Tekkno and House... is made to switch of the brain. No lyrics at all or stupid repetition in phrases only. It's a very old trick, like meditation or humming when dancing around a camp fire. Any meaningful lyrics would be counter-productive.
>
> #2: If I would print here for you the very best 10 phrases from lyrics of the true "European post-rock" musicians in native German, French or Italian you would not even be able to recognize the brilliance, the subtle message, the great jokes, the in-deapth truth.
>
> It is really better for you to regard Pink as the state-of-the-art lyricist – and please stop talking about continental Europe's lyrics.
>
> The internet as such is breaking the neck of the major record companies by breaking their monopoly distribution chain. That hurts. But it was absolutely necessary. The future of music will be very different from how it was from the 1950s to 2000. But it will be much better. And in many aspects it will be closer to what it was in the past. Music's first task is NOT to create a handful of super stars making super money – and leave the most musicians behind.""

As Lokasenna said, I was talking about commercial radio music. I have heard absolutely fantastic music from continental Europe, Japan, and all other corners of the world. The best music has nothing to do with locale, and I did not mean to imply for an instant that music from North America or the UK is categorically better than music from anywhere else.

Sorry if I gave that impression.

But I have heard a fair amount of stuff on mainstream pop radio in continental Europe that would not even pass muster as an advertising jingle in the US in terms of soul and artistic integrity (and American standards for advertising jingles are pretty low). "Big in Europe" and "Big in Japan" are somewhat regarded as backhanded compliments in the US music scene, implying that the artist is somehow akin to David Hasselhoff (who has been something of a standing joke in the US for decades but who had a very successful career in Europe).

My comments were regarding the top-40 commercial radio hits, not the quality of club music and certainly not regarding the quality of independent or artistic music, which in my experience, produce brilliance about equally anywhere, these days. Perhaps even more often in corners of the world that are not New York, LA, London, or Nashville, with their entrenched scenes and genrefied hierarchies.

On the world wide web, it is sometimes hard to contextualize things, especially for people in the US, for a whole lot of reasons (not the least of which is that so much of the english-language internet is Americans talking to each other). There is a widespread perception among "hip" Americans who have little experience abroad that "American" music is bad, commercial, and corporate, and that "foreign" music (or film, or whatever) is "better." My sense is (and my point was) that this perception is at least in part due to the fact that only the very best foreign entertainment manages to penetrate the extremely competitive American market, so small sample size leads someone who has heard only very little from outside his country to believe that it is representative of the homogeneous quality of the rest of it.

In truth, I think that the very best artistic entertainment is about equally distributed. If the US creates more bad entertainment than most countries, it's probably just due to the comparative size of the corporate entertainment industry in the US (and I certainly think a case could be made that, for example, Bollywood puts out at least as high a proportion of mindless crap as Hollywood does).

Perhaps most to the point, and maybe somewhat sadly/ironically, a lot of the stuff that is "big in Europe/Japan" (but implicitly not big in the US) is the exported dregs of the American entertainment industry. IOW, I didn't mean to imply in a jingoistic sense that American-created entertainment is in any sense better than anything else, just that soulless singsong crap is universal. And my statement was particularly for the benefit of americans who think that everything on the radio in Europe is the kind of innovative, artistic college-radio stuff that makes it onto the charts across the Atlantic. If I over-stated the badness of European commercial radio, then I apologize.

The role of the producer

> Quote:
>
> Originally Posted by DerMetzgermeister
>
> To Yep.
>
> When commercial bands do a record, who is usually the person who decide when, how and what studio "trickery" apply?
>
> I'm talking about those subtle touches that are not a part of the technical side of mixing but more of artistic choices. Those things that you can't detect at first hearing but nonetheless shape the whole "vibe" or feeling of a track: The doubled or multi-layered vocals, reverbs or delays used as special effects, weird sampled sounds, fades, extreme eq, etc.
>
> I'm curious because that can't count exactly as writing & arrangement but still are elements of a song, sometimes very important.
>
> Usually what's the dynamic? The producer/engineer makes the decisions? The bands or artists make suggestions? Are often this "touches" a source of disagreement between artists and producer/engineers?
>
> And for an amateur band that records and produces itself, do you recommend to stay away from those pitfalls and focus mainly in just achieving a clear recording and competent mix, or anything goes if that's part of your goal?
>
> Sorry if this is off topic.""

The role you are thinking of is that of the producer, who may be a member of the band, or may be the whole band, or may be an outside person brought in to manage the process of creating a record (see the "Producing Yourself" spinoff thread in this forum for more details).

Usually the band comes into the studio with songs that are either fully-written or partly-written based on the real-world instrumentation of the band. I.e. if there are two guitar players, a singer, a bass player, and a drummer, then those are the parts written. And they are usually not very "arranged" — that is to say, each guitar player basically has a verse part and a chorus part, and they each play their part through three times or whatever.

It is the role of the record "producer" (and that might be the band or a member of the band) to manage the creation of record — to listen to and evaluate this material that has been worked out in a rehearsal space or club tour, and to figure out what it's going to take to make this into a good record. That might be nothing — it might just be recording the band live in a woodshed. Or maybe the producer decides to bring in a professional arranger and hire a string section and horn players and a choir and record each string of the guitar separately or whatever. Maybe the producer just suggests a couple of breakdowns or key changes to liven up one or two of the songs that are in danger of turning monotonous. Maybe the producer re-writes entire songs, tells the band to reduce the number of syllables in the lyrics, hires a drum teacher for the drummer, and replaces all the bass parts with keyboards. It all depends on the nature of the project, who has control, what the budget is, and whose "vision" is driving the project.

The engineer is just there to do the recording and mixing, to operate the mics, processors, and mixing console. The engineer's job is technical, and most good engineers do not try to do anything beyond capturing and mixing the sound in as high quality as possible. The engineer typically follows the instructions of the producer.

Some producers actually do some or all of the engineering, a lot of them play instruments and/or contribute writing or arrangement ideas. Some of them don't know a compressor from a noise gate and can't carry a tune or keep a beat, they are just project managers who find the right people and who know how to say what's working and what's not and how to keep things on time and on budget. Some of them find old churches to record in and set up tapestries and incense and collect all the band's cell phones and give them pot and mushrooms and eastern philosophy books in an effort to spur creativity, some of them are martinets who take over the project and tell the band to stand aside while they make a record, some of them are baby-sitters and cat-herders who just try to make sure that something productive is happening every day, some of them are schmoozers and wheeler-dealers, some of them are just former musicians or engineers who know what it takes to get a big recording project completed, some of them are creative "fifth members" of the band with a special expertise in the studio, and some of them are just budget watchdogs that the record company hired because they don't know who the band is or whether they are capable of completing anything.

In all cases, the "producer" is whatever person is ultimately responsible for getting a record made, and is usually the person who is responsible for the difference between a well-recorded rehearsal and whatever ends up being the finished record (even if that is just a well-recorded rehearsal). See the other thread for more.

[begin page 18]

Specific monitor recommendations?

> Quote:
>
> Originally Posted by TedR
>
> Yep, is their any specific model/brand of monitors that you like in particular ? At maybe 2 or 3 different price points ?"

That's a really hard question for me to answer, because I do not have anything close to comprehensive experience with lots of monitors.

I still like NS10s, but I think that's only because I'm used to them and have a good feel for what is wrong with them (basically everything but the midrange). They are certainly not worth the prices they now command on eBay. On the expensiver side, I have been really impressed with the big two-woofer ADAMs. I can't personally justify that much money for speakers, not having much significant income coming from recording.

Almost as good is the JBL LSR series, which is a little closer to mid-market. I have liked the ones I've heard a bit better than (I think) similarly-priced Mackies, which are also fine, and popular, but a bit indistinct in the low end to my ears. I would usually rather have a higher cutoff frequency with better accuracy than have a lower-range speaker that turns tubby or one-note-bass-ish at lower frequencies. YMMV.

On the cheaper side, Guitar Center was selling EMU PM-5s for $100 apiece a couple years ago, and I bought five of them for a surround setup. They're not super-loud or super-low, but they are outstanding speakers down to 55 cycles or so. I do not know if they are still available at that price, but if you can find them, grab them.

On the super-cheap end, you could do worse than a pair of the old (passive) Radio Shack Minimus 7s, which are kind of a poor man's NS10. The lows are surprisingly good for such tiny speakers, but the cutoff is still pretty high by 2009 standards. Scrounging on ebay can turn up a pair for $50 or so, I think. I have also never heard a Tivoli Audio system I didn't like, and while the regular retail prices on them often approach bona-fide monitors, places like Target or Sears occasionally have blowout sales where you can snag a stereo system for $40 or so. Similar bargains might be found among other inexpensive "audiophile" systems.

I don't really have much experience with the cheaper Behringer or KRK monitors, but my guess is that they are vastly superior for monitoring to any home stereo made by Sony or whatever.

In any case, when evaluating monitors, I would encourage you to use a familiar CD and to listen carefully and systematically to every frequency range. Are the bass notes clear and distinct? Could you tune a bass guitar to the bass notes from this speaker? Can you hear how hard the drummer is hitting the open hi-hats, or do they turn into trashy hash? Can you hear the kick drum as a distinct and real instrument, as opposed to a vague thump? Do acoustic instruments have a clear and varied dynamic profile, with distinct transients and a natural and realistic decay into audio "black space"? Is the overall sound natural, complete, and without "holes" or "hash" or "peaky" frequencies, especially in the midrange? These are the things that "good speakers" but "bad monitors" tend to fail at, and they can make or break your recordings. Trust your instincts, and keep your skepticism about you — you have a better ear for these things than you think you do.

Non-native English

> Quote:
>
> Originally Posted by stupeT
>
> You're welcome.
>
> I just wanted to emphasize that in "continental Europe" there are NO native English speakers. That makes our native lyrics hard to understand – and judge – for the native English speakers. On the other hand if we dare to use English it obviously must sound "funny" or ridiculous to native English speakers ...""

On a side note, I wish more international acts who choose to sing in English were less self-conscious about sounding naïve in terms of grammar, accent, and idiom, and more willing to just say what they mean without trying to sound "American." I think any native english-speaker who works with international acts as an engineer or producer has experienced the bands that are self-conscious about their English, and who are eager for lyrical advice. But some absolutely fantastic songs have been written, performed, and recorded by "foreign-sounding" english-language bands.

Whether the predominance of american english as the global language of popular entertainment is a good or bad thing is a topic for another thread, but one thing that american english has going for it, and that I think a lot of people don't fully realize, is that it is an extremely fungible, utilitarian, and variegated language. There are a thousand different dialects of english in any big American city, and hybrid languages such as "Spanglish" and strange idioms and unconventional grammar are part of everyday life in the US.

Strange constructions, pronunciations, rapidly-changing slang and idiom, and so on are not only readily understood but commonplace in American english. The complexities and confusing spellings and grammatical rules of English are not in spite of, but because of the readiness of English to absorb foreign words and constructions. And for the most part, American English-speakers understand each other pretty well, even when they are almost speaking different languages. And Americans (and most British) are not French, nor Russian, where a flub or a mispronunciation is going to cause offence or misunderstanding.

Moreover, the global entertainment consumer base has had zero trouble adopting spanglish and ever-evolving hip-hop dialect and so on. The people who matter in terms of english-language global entertainment generally do not care much about your accent, grammar, or idiom, they care about the substance of what you are saying, and english, as languages go, is fairly forgiving in that respect.

English, and particularly American English, is a sprawling hybrid, a hodge-podge of spelling, pronunciation, and grammatical rules, a utilitarian catch-all that readily adopts foreign as well as made-up words and constructions, whatever expresses the concept.

Nobody much cares whether you ask "What is your name?" or "What your name is?" or "What your name?" or even "You who?" or "What name?" People in international cities at least do not associate unconventional grammar with low intelligence. Indeed, some of the best english-language poetry in modern popular music comes from either non-native speakers or (for example) hip-hop artists who defy conventional grammar.

Also, I really like "Rock you like a hurricane" even though it doesn't make a lot of sense.

Monitors deteriorating with age?

> Quote:
>
> Originally Posted by Marah Mag
>
> ...I think that, within reason, being familiar with a set of monitors can be more important than some objective quality they may or may not have...""

This is only true as long as the "flaws" of the speaker are not fatal to monitoring. A lot of which I discussed earlier in this thread, but for example, a lot of "good sounding" bookshelf and home theater systems simply cover up and take over massive ranges of the sonic spectrum in ways that are, I think, impossible to "get used to."

> Quote:
> Originally Posted by Marah Mag View Post
> Let me ask you... do or can speakers ever go bad, eg., with age?...

Yes, although with legitimate studio monitors is almost a theoretical concern. Assuming a magnetic moving-coil driver, what happens is: you have a fixed magnet with a coil of wire wrapped around it. The coil of wire is attached to a flat-ish “cone” that is big enough to displace air molecules. The cone is held in place by a flexible suspension surrounding the cone.

When you pass electrical current through the coiled wire, it generates a magnetic field, that wants to either push in or out in relation to the fixed magnetic filed of the speaker magnet. An alternating current (like audio signal) will cause the coil to move back and forth, and that will in turn push the cone in and out, creating positive and negative air pressure that we hear as sound.

Over time, the suspension may either become looser or stiffer, as humidity, drying, sunlight, temperature, air contaminants, age and mechanical excursion take their toll. The speaker cone itself may start to flex and become “floppy.” The fixed magnet may also tend to lose some of its charge, although this happens excruciatingly slowly in terms of component life-cycles. The wire coil or the structure it is affixed to may likewise start to develop a semi-permanent charge from sitting in the magnetic field.

If you have ever left a sturdy cardboard box in a dank basement or humid attic for a year and come back to find it flabby and splitting at the seams when you go to pick it up, you can imagine what happens to cheap paper speakers over time. As the mechanical resistance of the air-moving driver changes, so changes the relationship of input electrical signal to output sound. This is where tales of speakers sounding better after being “broken in” for a time come from — in the early audiophile days of the late 1950s, it was often true that speakers came from the factory with stiff suspensions that would mechanically “relax” after some excursion (e.g. playing loud music for a period of time).

But these days even consumer stereos are usually made with synthetic or coated driver cones and man-made suspension materials that start out with almost zero resistance and stay that way for a long time. The foam-type suspensions that are now commonplace are as likely to disintegrate with age as they are to appreciably change in terms of their mechanical resistance. And speaker cones are typically made to survive the next ice age. And the ceramic or other specialized materials that are used in decent speaker magnets last a lot better than cheap supermarket magnets that start to fall off the fridge after a couple years.

And I would expect all of the above to be true even of inexpensive entry-level speakers that could reasonably be called "studio monitors" these days. The materials and construction are not terribly expensive, and the markups above cost to produce are high even with Behringer stuff. Which leads me to something else I've been meaning to touch upon...

Why is some audio gear so expensive?

The short answer is labor. As soon as you go from having unskilled workers stamping out circuit-boards by the thousand to having skilled engineers hand-build, test, and calibrate something, you have increased the production cost by an order of magnitude. I think that's easy enough to understand.

So why not just have the skilled engineer build the first unit, and then have another skilled engineer develop drawings and manufacturing procedures to copy the original? Well, that's basically exactly what Mackie did by copying Neve preamp designs and converting them to pc boards in the early 90s, and then what Behringer did by copying Mackie designs and building them in China some years later. So shouldn't Behringer be just as good as Neve, at this point? Not necessarily.

Let's take a look at a single component to see whether there might be merit in having a skilled engineer hand-build small-batch electronics. Let's take the simple potentiometer (or knob, in common speech). (With some technical over-simplifications...) You can buy knobs from electronics catalogs for less than 50 cents apiece in bulk. Their typical construction is a braided, flat copper wire or "brush" with a sort of frayed end, that twists back and forth across a strip of metal that is wider at one end and narrower at the other, or something similar. At the narrow end, there is more resistance, and at the wider end, there is less resistance. So far, so good.

So what's the problem with these cheap knobs, and why is there also a market for $100+ "military grade" potentiometers with the same resistance values? For one thing, the "brush"-type knob is very poor at repeatability. Depending on any number of factors from the contact of the strands to the ambient temperature or humidity to the amount of tarnish on the metal contacts, two identical brush-type knobs might output significantly different values at the same settings or even on different days or depending upon usage. Moreover, this design is subject to dead spots and interference (crackly knobs, or knobs that cut out in certain places, or knobs that have no difference between, say, 4 and 8, and then immediately jump in value at 9). They are also fairly crude and do not present solid connections, and resistance could develop due to tarnished or dirty contacts, and capacitance could develop in the gaps between strands and metal.

These tiny discrepancies might be tolerable in clock radios and possibly even for one-person hobbyist studios, but if you are depending on the knob to determine the location of a guided missile launched from a hundred miles away, then obviously a percentage or two is a pretty big deal. Similarly, if you're running a commercial studio that costs $50 per hour in real estate and overhead alone, and you spend 20 hours on monday and tuesday setting up 20 tracks of drum sounds, each of which might through 6 potentiometers, then you do NOT want to come in on Wednesday morning and have to re-calibrate 120 knobs or track down a crackle. Moreover, you don't want to have to deal with inconsistent or uneven knobs in the first place when you have to set up 120 of them just so.

Solution? Use better knobs. Maybe they are stepped attenuators, or use calibrated spring-loaded rollers instead of brushes, or whatever. Problem is, in order to be truly worthy of directing guided missiles, these things can't simply be stamped out in an automated factory. Someone skilled needs to actually sit down and calibrate each and every one of these knobs. And in a world where there is a finite quantity of electrical engineers, that step suddenly costs real money. Whether you do it yourself, or whether you pay someone else to do it, or whether you simply bundle it into the cost of manufacturing, the fact remains that somebody who has the skill set to be doing something much more valuable and productive than testing knobs or matching resistors needs to sit down and actually verify the consistency and repeatability of these things, or at least hand-match resistor values.

Perhaps this process could be automated? Almost certainly. But inventing, programming, building, testing and calibrating a machine to do this excruciatingly specialized task is mind-bogglingly expensive. And we're not making toasters or flat-panel TVs that will sell millions of units in Best Buy. Specialty studio component manufacturers like Tube Tech or Thermionic Culture (for example) might move 20 units a month. There are only so many obsessive recording studios out there. And their products usually have a slow and expensive person-to-person sales and distribution network through professional audio dealers that spend many hands-on man hours on every sale and account. This is not stuff that could simply be farmed out to illiterate Chinese peasants looking for a factory job at $2 a day.

And all this is only talking about a simple knob. Just one component. The skilled electrical engineers who run a four-person operation obsessively building boutique studio gear are people who could be making a lot of money in the corporate world, with sick leave and vacation time and retirement and health care and job security and so on. For them to quit that life to build preamps in the garage is a big risk for themselves and their families, and there is no reason for them to do it if they cannot charge enough markup to make it worthwhile.

Is Behringer (or Mackie, or Presonus, or whatever) "good enough"? Sure, it's fantastic for the money. Could someone make a hit record with it? Absolutely. Would you want to use it to run a large-format commercial studio where you were expected to churn out a new hit record every two weeks, with bands you've never heard of and might not even like, while paying downtown rent in LA or NYC and charging accordingly? Maybe not.

That is what makes "professional" gear "professional." Not necessarily that it always magically produces results that are categorically better than anything that could have possibly been achieved with Behringer or Mackie or whatever, but that it is designed for everyday use by actual working professionals to meet their real-world requirements.

Professionals do not generally have time to see whether six month's worth of tweaking their favorite band with free plugins can match the results of simply using top-quality gear to begin with. Professionals need to make every act that walks in the door sound like a million-dollar rock star, and they need to do it every day, day in and day out. It's their job.

[Stopped 6-17-09, Post #694]

Note by Smurf

PLEASE READ!

I have got a few emails about how I am doing this PDF, so I thought I would address the questions / comments here...

I am doing this to document yep's Thread & Ideas, not all of the extra comments & arguments. I will include others posts if the relate to the flow of yep's points & examples, but not all of the "other side of the coin" type of posts.

This is why I am not documenting the ENTIRE thread. I will leave that to someone else. I am just archiving yep's posts for my own info, and decided to share it with others.

If I leave someone out, or some view out, or some point out, it is NOT because I am "choosing sides", or not willing to "relate the entire story", I just feel they add nothing to the information that yep is posting, nor to HIS way of doing things.........so I apologize in advance if this upsets anyone.

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Thank You.