Why do your recordings sound like ass? 2

Started from 12-24-2009

Post #1126

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Quote:
Originally Posted by Black Shadow
Another way to do this is by zooming in very closely and applying tiny fade-outs and fade-ins (and then duplicating the edited part). You won't hear the fades and audio clicks are avoided.""
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There is nothing wrong with this approach, and in some technical sense it's probably "ideal", but in practice it can be a PITA to try and find a good edit point where both waveforms are both exactly at a zero crossing. More to the point, I don't think anyone has ever noticed an otherwise "good" edit done with a crossfade, and automatic crossfades are certainly very easy to do in REAPER (and most other DAWs).

Quote:
Originally Posted by GregHolmes
Ah, yes, I remember it well... :-)

But I was thinking more about tonal and performance discontinuities when cutting and pasting takes, rather than techniques like splicing at zero crossings or cross fades.

The performer may "dig in" or speed up at the end of a verse that leads to a chorus, making it harder to use that take as a verse which leads to another verse or bridge.""
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Ah, well, that is a whole different problem. And a very interesting one.

Everything has a tradeoff, and one of the huge tradeoffs of productions that start out with the intent of being a "computer" recording is that a lot of the benefits in terms of flexibility and easy editing are, to varying degrees, dependent upon somewhat limiting the creative/artistic aspects of the performance.

I'm guessing that most of us have either heard of, experienced, or imagined an ideal recording scenario where the artist simply performs a whole boatload of material, some awesome, some less awesome, and then an engineer goes in and splices together all the awesome stuff to manufacture a 3-minute virtuoso performance.

The problem is exactly the above. For starters, this approach almost necessitates recording to a click or drum loop or something. For seconds, even with a consistent tempo and key, the "awesome" performance from yesterday may not necessarily mesh well with the awesome performance from today.

Maybe yesterday the singer was dragging the beat a little, singing a little more soulful and breathy, using slightly vaguer pitch. And maybe it sounded great. And maybe today, we had an extra cup of coffee, and we're pushing the beat, singing full-voice and dead on-pitch, with the whole "American Idol" voice, and it also sounds great, just a different kind of great.

Now when we go to splice those two together, we end up with some parts sounding mumbly and pitchy, and other parts sounding over-bearing and strident. Worse, the edit points don't quite add up. One overlaps the other, and the tempo feel doesn't really "mesh".

Every so often, a producer/band gets lucky and finds a perfect "mesh" (the Doors' epic "The End" was spliced from two semi-free-form takes, for example).

But more often, doing a specifically "computer-based" recording (one with lots of double-tracking and spliced takes) usually means somewhat restricting take-to-take musical expression and freedom, and starting from a place of doing every take "perfectly", or by-the-book. So the singer and everyone else has to kind of "decide" what the feel and delivery and expression is going to be like, and then try to deliver every take as a sort of mechanical karaoke exercise. Instead of delivering the music the way they are "feeling it" at that moment, they should try to deliver it the way it's "supposed" to sound, whatever that means.

For good or for ill, this tends to mean a somewhat more robotic, "casio keyboard" vibe to the performances. Whether this is a good thing or a bad thing is a topic for debate, but it raises an interesting point:

It's practically a certainty that performances such as those heard on, for example, classic Rolling Stones or John Lee Hooker or Velvet Underground records could have been done with a modern multitrack production approach. The individual performances are simply too "loose" and too interpretive to expect clean double-tracks on all the guitars and vocals, etc.

There is a big textural difference between the smoky, rambling, almost sloppy delivery of a Rolling Stones track and the mechanically "perfect" precision of post-80s rock (anything from Bon Jovi to Offspring or whatever). Def Leppard's producer wanted everything to sound "perfect" and "thick" and "big". He got that sound by stacking multiple tracks on top of each other. The way he achieved this was by getting the band to play every take exactly the same way.

That kind of approach was a turning point in rock and pop music production. The upside is much "bigger", more "perfect" and "polished" recordings. The downside is the loss of that smoky, soulful, "in the moment" imprecision that happens when real musicians are just "feeling it". Neither is better nor worse, they're just different ways to entertain people.

But there is an important decision that gets lost in the evaluation process.

If the Rolling Stones had never cut an album, but instead stepped into a time machine and walked into a modern studio today, and wanted to record, there is a very good chance that the studio might start out by, for example, asking Keith Richards to play a riff the same way twice, and then compare to see whether it sounds better double-tracked. And chances are very good that the engineer/producer (and the band) would decide that ONE measure of ONE riff sounded better, bigger, more "polished" when two tracks were playing than one.

Then they might try the same with Keith Richard's voice. Turns out that looping a measure of his voice with and without doubling reveals a much thicker, more polished sound when double-tracked-- it downplays some of the bleatiness, and beefs up and smooths over the reedy, imprecise tonality.

Similarly, what happens when we replace Bill Wyman's old bass with a new active pickup alembic or music man or whatever-- punchier, stringer, more articulate bass sounds. Solo'ed, one measure at a time, it sounds a lot more dramatic.

So we decide to record the band to a click. We tell Keith Richards to pick an arrangement and performance style and stick to it exactly, on every take (since we need to double-track it). We probably also decide that it would be better if he played power-chords, or doubled the parts a fifth up, since that sounds a lot "bigger" still, when we listen to each measure one at a time. We get the young Mick Jagger some singing lessons and train him to sing every note full-voice, on-pitch, and with the same timing in every verse. We double-track and edit all the vocals to get the most "perfect" sound on each individual note. We zap that fat old flabby bass sound for a modern, transparent, slinky bass sound. And so on. And listening to solo'd tracks one measure at a time, each of these sounds like an unquestionable "improvement" to everyone.

And at the end of the process, the world has a Rolling Stones discography that sounds a lot less like the Rolling Stones and a lot more like Def Leppard or the Offspring.

This might or might not be a net improvement, but it's certainly a very significant change. The big difference is the methodology of looking for "better sound" in terms of isolated, measure-by-measure decisions, as opposed to looking to accurately "capture" the sound of a very talented band.

Which is better and which is worse is a matter of preference, but the important lesson is that we do not necessarily achieve "the same but better" results by making evaluations in isolation. Splicing and layering and multi-tracking everything not only changes the fundamental sound, but the musical approach. Keith Richards is not two guitar players playing the exact same thing every time, and Mick Jagger is not singing three tracks plus a whisper track every time.

The whole sonic texture and vibe of the Rolling Stones we know is two rambling guitarists playing off a fat, subsonic, meandering bass, with an inconsistent, reedy, bleating singer who sounds like he's jumping all over the place. If we replaced that with a tight, deep, modern bass sound; thick, constant, precise guitars; and a smooth, consistent, layered vocal, all tracked to a click and playing "perfect" renditions, it would no longer sound like the Rolling Stones. And the thing is, there is no way to tell what you're gaining or losing with simple measure-by-measure A/B comparisons.

Quote:
Originally Posted by sissyneck67

http://soundcloud.com/mice/the-devil-and-the-pilot

So I'm trying to hit that next skill level, and figure out what's wrong. Here are what I hear as problems:
- Overall I think the song is too heavy in instrumentation, it's not 'breathing' anywhere
- the banjo is sticking out too much, and may need to be removed entirely, I'm not sure I like what it adds.
- the backing acoustic guitars are too heavy, creating this sort of wall that I don't

think is adding anything
- the beat is off

Would you be willing to tell me what changes you think would benefit the song, or any thoughts you have?

I'm very fine with a touch of lo-fi, I'm not looking for crisp perfection. But I do want it to be....better.""
=====

I think you're on exactly the right track with your ideas.

The banjo does seem to be a bit of a third wheel.

The piled-up draggy quarter-note jangle-beats sound like something from a dirge, like when they march child slaves through the mines in a musical or something. Something needs to be doing something different. You might be very surprised at how effective a lo-fi, slightly funky or uptempo drum loop can be with this kind of material. You can use some telephone eq, mild distortion/saturation, and some reverb to get a ghostly effect, like it's not really there.

As an experiment, try muting the tambourine, and then dialing in the guitar sound to *sound* like the tambourine with steep eq to isolate the upper mids and heavy compression of the trashy/saturation variety. I doubt whether that is exactly the sound you're looking for, but I think it might help to re-orient the way you're approaching this in a helpful way.

A similar experiment you could try with the banjo part is re-recording it with a very heavily overdriven electric guitar, using a deep reverb and/or delay. Counter-intuitively, heavily overdiven guitar notes, when turned down quiet in the mix, can often sound very dreamy, haunting, and faraway. Especially sparse, sustained single notes. If you can't or don't want to re-record the banjo, just try hitting it with some heavy compression and guitar distortion and see what happens. I'm not sure if you'll get enough sustain to get that wailing effect (might want to try a low cut first).

I might try and "fake up" that harmonica sound a bit, possibly with some autotune/vocoder type effect, and maybe some synthesizer-type extreme eq, maybe followed by pitch shifting or octave doubling up a fifth or something. Also sweeps, glitchies, and granulation-type effects might help.

I think one of the things you're running up against is that the instruments are distracting from the atmosphere. The vocals have that airy, dreamlike, ethereal quality that I think you're going for, but the instruments sound rather quite tangible and "made by the hand of man", in comparison. One way to try and change that is to make the instruments sound less clearly identifiable, less realistic, more like memories of sounds than like sounds themselves.

I kind of feel like, if angels or ghosts were going to play music that we could hear, they'd do it not by picking up a guitar and plucking strings, but by making the radio or TV warble and modulate through static, or by making the guitar strings resonate without actually being plucked, or by sending echoes of sounds that don't have a source-- that kind of thing.

I'll also suggest that recording might get a lot easier if you have some kind of percussion and bass tracks to build from. They need not be slamming or hard-hitting, just something rythmic and a tonal bed to lay the other sounds on top of.

Have fun!

PS-- less is more, to cite a cliche.

It sounds like this track might have a touch of "pile-on-itis" where one thing then another keeps getting added on to try and make up for whatever's missing from or wrong with the other stuff. If the ingredients aren't right, the solution is not to use more of them.

Not sure how helpful that is, but that's what I got.

RE: the acoustic guitar stuff...

What everybody else said.

There actually was some pretty extensive stuff on acoustic guitar earlier in the thread, and since everybody else has the same search box as I do, it will be a lot easier to reply to specific questions about what's working/not working than to simply start quoting or re-writing earlier stuff from an already bloated thread.

The recap is that a lot nearfield/farfield issues come into play when trying to self-record acoustic guitar. What the player is hearing if often completely different from what the audience would be hearing. Also the fact of the guitar resonating against your chest cavity means that headphones don't necessarily isolate they way they would with keyboards or electric guitar. Also, a mic pointed in the soundhole is usually about the worst way to record acoustic guitar, but often the first resort of beginners.

Lots more detail earlier in the thread.

Quote:
Originally Posted by mymymetrocard
...How often would yep/anyone recommend replacing drum heads? 'when they break' has previously been my thoughts on the matter...""
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Definitely more often than that!

As heads stretch, they lose their elasticity, which significantly affects the sound and playing feel of the drum. New, good drum heads "pop". The stick bounces of the head with a satisfying "crack". Old, worn-out heads sound and feel dead, like hitting a piece of rotten fruit, or rubbery, like hardened jello.

Heads that buzz or rattle, or that have to be tuned super-tight to sound okay are past their prime. Any drum head that is dented or that "bowls up" when you take it off the drum is long past the point where it was bouncy and elastic. If the coating is wearing off the head, then it's past the sell-by date (the reason the coating is wearing off is because the head is stretched and deformed, and the coating cracks and peels off like old vinyl seat covers). Old or bad heads sound boingy and/or dead.

Now, with that said, drum heads are expensive. And some brands of heads are better than others. If you're a cheapskate, my recommendation would be to keep two or three sets of heads: When you set out to make a proper record or to play a major show, buy a new set of drum heads (make the band split it, and offer to also split new guitar strings and a pack of tea for the singer). Then when the record is done, put the old crummy heads back on and use them for practice. Set aside the "good" heads, and save them for gigs worth changing heads for. Next time a recording project or "major" gig comes up, buy new heads again, retire the old crappy ones to replace breakages, and make the older "good" ones your everyday heads.

Do some googling for "longest lasting drum heads" or whatever before you buy-- some heads are definitely better than others, and I don't have the expertise to tell you what to buy. The cheap, stiff plastic heads that come with most drums suck-- they break easily and they only sound good for about the first 20 minutes of playing. You can almost stretch them out to sagging point just by pushing on them with your hand. You want tough, bouncy, elastic heads for good sound, not cellophane.

Quote:
Originally Posted by mymymetrocard
...they really are expensive tho...""
=====

Drum heads are absolutely one of those things that should come from the "band" expense account, assuming the band shares expenses. Just add $X/month or whatever to the rehearsal space rent, and use that purchase strings, picks, cables, drum heads, etc.

It's not right for the singer to have to buy nothing but earplugs while the rest of the band has to pay for disposables simply because of they instrument they're attached to. A band should be run like a business: the expenses should get paid before the shareholders (members) do. If you're playing out and making any money, that should go into the expense kitty, and stuff like batteries, drum heads, mic cables, new tires for the van, etc should come out of that account.

Of course most bands are somewhat money-losing affairs in their salad days, but the same principle should still apply. If there is a shortfall, then everyone should split it, unless the band exclusively "belongs" to one member (in which case *HE* should pay all expenses, since the rest of the band is really just employees).

edit: sorry for the way OT

Quote:
Originally Posted by rhkk
...here is my question:
I am trying my best to level match right after this step. (pre-mix) So after a few passes I can get it sounding pretty good volume wise. Then I start to add some delay of reverb etc... and it changes the volumes (or at least the perceived volumes) My first reaction is to always reach for the faders to fix this issue but I'm getting the feeling I should probably leave these alone for the most part since I already did my level matching. So is it a matter of going back and forth between

effects and then adjusting faders over and over again? Or should I first try to fix perceived volume issues with eq and maybe compression? I guess the best way to ask this question is: How much should I be changing individual track volumes after I level match? (I feel like I might be doing it too much)

THANKS AGAIN!!!""
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If I understand your question correctly, the problem is that reverb is changing the apparent level of the relative tracks (?)

For starters, if you're using insert reverbs (i.e., dropping a reverb effect in the vocal track), then that is certainly changing the real and apparent volume of the vocal track. If before, you had just vocals, and now you have vocal + reverb, then you've probably increased the vocal track by 3dB or so on the meters. moreover, your *perception* of the vocal has probably increased even more since the vocal now has longer tails and a "bigger" sound (OTOH, it could sometimes sound "quieter" since it's pushed "further back" with the details washed out).

If you're using a reverb bus, same thing applies.

I would suggest setting up a reverb bus basically as soon as you start mixing. It doesn't have to be anything you end up keeping. But it's practically a certainty that you're going to want some amount of ambiance on at least some of the tracks, so it makes sense to create a simple bus to send them all to, in varying degrees. This is partly a hardware-centric workflow from the days when you only had one reverb box in the studio, that may be unnecessary in a plugin environment, but that's how I like to work.

To get really detailed, my suggestion for building a mix is something like this:

1. Throw up all the faders to zero and create a reverb bus that everything goes to, and just flip through presets to find a decent patch, typically a "room" or "chamber" or "tight hall" or maybe a plate or something like that. Just something that sounds like a generic, unobtrusive reverb. Don't waste much time on it, it's just a placeholder.

2. Turn down all the faders.

3. Next turn up the kick drum. Dial out the reverb send to zero. Get the kick sounding good and punchy (compression, eq, etc), vibing the way you want (sending a little back to the reverb is okay, but the less the better). You want the kick to end up at around ~12dB peak, but don't sweat it too much, just keep the levels low and adjust your speakers if you want more volume. Try to keep the kick drum as tight and short as you can, within the context of the material. Go for thumpy and punchy rather then boomy and huge.

4. Bring up the snare drum, and get it to sound good with the kick (compression, eq, distortion, etc). Some reverb is probably in order. Ignore the levels, just get the snare sounding appropriate to the kick drum. Again, go for as tight and focused a sound as makes sense for the tempo and material, but don't hesitate to throw a bunch of it to reverb if sounds right.

5. Bring up the bass fader (probably no reverb on this one). Use eq, compression, distortion, reamping, amp simulators, whatever, to get the bassline sounding good with the kick and snare. The kick, snare, and bassline are the critical elements of your rhythm section in a typical rock/pop mix. You want these three instruments combined to be hitting about -10dBFS peak on the master out at this

point, which gives you just about the right amount of headroom that you should be able to add all the other instruments and have the overall peaks coming in at slightly under 0dB, which means you shouldn't have to worry about levels, metering, or clipping for the rest of the mix: you can just mix by ear and tweak/limit/normalize the final output a hair at the end. IOW, from here on, basically ignore the meters and fader positions: just mix by ear and do whatever "sounds" right.

6. Bring up the lead vocal fader, and get it sounding good with the kick/snare/bass mix that you have going. Probably a combination of eq, compression, and some stuff to the reverb send. You may want to drop in an extra "vocal only" reverb on the vocal track, like a short plate or delay to fill and thicken the vocals. If you are using double-tracked vocals, bring them both up and do whatever you meant to do to get them sounding like whatever you want your double-tracked vocal to sound like (again, we can tweak/finalize the ultimate reverb later--the reverb bus is just a placeholder effect for the overall "ambiance")

7. The above are your most important instruments, so review, and make sure they all "fit" together, sonically. It should actually sound like a decent mix at this point, if a bit sparse.

8. Bring up your most important rhythm/backing instruments (guitars, piano, etc-- the "chords"). Get them to sound good in context with the existing kick/snare/bass/vox mix. This might be a lot of work. Use the reverb send to taste. When in doubt, make the "chord" instruments defer to the above. The tighter you kept your drum sounds above, the easier this will be.

9. Bring up your "sweetener" tracks: backing vox, tambourine, synth pads, congas, whatever. use eq and compression to fit them in wherever there is room.

10. Go back and re-visit the overall mix, and tweak the reverb send effect as desired.

One thing I did not include in the above step-by-step is where to bring in the other drum tracks besides kick and snare. This is a judgement call, mostly depending on how "important" the other drum sounds are to the song. And it's sometimes a split decision. Fills and cymbal splashes are usually, but not always "sweeteners" that fit in around step 9. Hi-hats or repeated tom patterns are usually more like rhythm instruments in step 8. Sometimes the OH or room mics are the basis of the drum sound, and are the first thing you pull up. So a lot depends on how you recorded. The whole idea is to get the most important stuff fitted into the mix first, and then make the less important stuff fit around it.

In any case, the larger point is to introduce reverb as a distinct element early in the mix process.

Quote:
Originally Posted by rhkk
...I had a rendered vocal track that I had already eq'd the extreme lows and extreme highs off of. I knew I had done that but thought it would not hurt to do it again since now I was going to get more detailed with the eq settings. In other words I assumed it would not make a difference to cut the lows and highs twice. I kept thinking the vocal was sounding thin especially when put in the mix so I finally decided to bypass the eq and WOW it made a difference. So I took the low/high cuts out of the eq and it sounded great. So unless I am insane I am concluding that cutting out low/highs twice makes a difference but I am not sure

I understand why? The only thing I can think of is that the eq cuts still have "curves" on them so you are not completely cutting out the extreme highs and lows on the first rendered track, so to do it twice accentuates the cuts too much?...""
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I guess something I should be careful of is to say that cutting lows or highs is never a good idea, unless they need to be cut. I made some posts earlier where I suggested cutting lows and highs, not because lows and highs are bad, but because there is a common tendency to track with too much of them.

When you're dialing in a single track of vocals or whatever, there is a tendency to go for the biggest, most dramatic, most full-spectrum sound possible on each track. This tends to produce a project full of tracks that are hard to fit together, with too much on the extremes. Hypothetically and ideally, you'd have tracked everything perfectly and it would need no eq at all, but that's pretty unusual. So I suggest experimenting with low and high eq cuts, because I suspect a lot of people reading this thread might otherwise never even try it, and it can often make a huge improvement.

That said, it's by no means a "recipe" for good sound, and if your tracks don't benefit from it, you shouldn't do it.

To your specific question, there is nothing right or wrong with doing any kind of eq in multiple stages vs just one stage. Whatever works. If it needs more eq, eq it again. If it doesn't, don't.

Quote:
I also have a question/comment on "super compressed" music. I certainly am not an advocate of Overly Compressed music by any means. I LOVE DYNAMICS! That being said I always read that people get fatigued by listening to OC music which I agree with especially at loud volumes. But if you listen to these songs at lower volumes would the opposite not be true? I mean isn't it "fatiguing" to listen to uncompressed music at a lower volume because you can't really hear everything in the mix?""
=====

No. It's important to understand what we mean by "volume". "Volume" is a quantifiable but still subjective measure of the average sound level of a thing. If you hear one hyper-compressed piece of music at an average sound level of 55dB SPL (very quiet) and another uncompressed, highly-dynamic recording at an average sound level of the same 55dB SPL, they will both have the same *average* loudness, but the dynamic, uncompressed one will have significantly louder and more natural-sounding instantaneous peaks.

HOWEVER, the uncompressed version will have to be played at higher gain settings (a higher "volume" setting on the volume knob) in order to achieve that same perceived loudness.

This is confusing, but important: the "volume" knob on your stereo does not have fixed settings for perceived sound level. It only adjusts output gain. Try this experiment (seriously) to see what I'm talking about:

1. Put a DVD or Blu-Ray movie in and start watching it with the volume at a comfortable level. Be sure to turn all audio settings to flat and to turn off any kind of auto-leveling or other compression on your player and speaker system. (There are exceptions to every rule, but these movie soundtracks are usually left pretty

dynamic, with a -18dB or lower average signal level).

2. Now, without changing anything or adjusting any volume settings, eject the DVD and put a recent commercial pop CD in the same player. Start playback.

Unless you picked really unusual examples, or have some kind of compression or auto-leveling feature turned on, the CD will sound LOUD AS HELL, like 2 to 4 times louder or more than the DVD movie. If you are a sane human being, you will immediately reach for the remote and turn down the volume. And similarly, if you put the movie back in, and it sounds too quiet, you will pick up the remote and turn the volume back back up. Which is what everyone does.

In short, the signal level embedded on the digital recording has NOTHING TO DO with how loud or quiet your playback is-- you control that entirely with the volume on your remote control or knob or whatever, same as every other listener everywhere in the world. The only difference is, if the recordings is dynamic and uncompressed, you'll still hear whatever average volume level you prefer, but the peaks and transients will be musical, dramatic and exciting.

In all seriousness, the peaks in the uncompressed version will be LOUDER than a "hotter" mix, which will simply sound flat and steady-state, because you're not adjusting your volume control according to some arbitrary measure of peak signal level, you're adjusting based on average sound pressure level as you perceive it at the listening position.

If the compressed mix has only 3dB of headroom, then you're hearing your preferred listening level with peaks 3dB louder. If the uncompressed version has 12dB of headroom, then you're still adjusting the volume to get the same average sound level, but now the peaks are 12dB louder: More impact, more thump, more punch, more drama.

I really encourage anyone who is unclear on this point to go back to the early "golden ears in one easy step" parts of this thread. This is the number one cause of mixing frustration and bad sound in beginner recordings: You stay up all night "adding" stuff, and everything you add to the signal makes it sound "louder", but then you come back the next day or hand it to a friend or listen on a different system when your ears aren't blown out, and you turn it down to a sane listening level, and the whole thing sound like a flat, distorted, tunnelly vortex of shit.

What happened was, you spent all night doing what teenagers with under-powered car-stereos do: you kept turning it up louder, and every time it got louder it got a little bit more distorted and your ears start to fry out a little more, so you turn it up a little louder, which makes it flatter and more distorted and more fatiguing which saturates your ears even faster, so you keep turning it up more and more.

Except you're not turning up your speaker's volume knob, you're just loading up the digital signal more and more, which means everyone who ever hears this recording is going to hear the same thing we hear when we pull up alongside a deafened teenager with an under-powered car stereo: a clipped, fizzy, distorted, buzzy mess.

And if you took that teenager out of the car, or took the recordist away from the computer, and put earmuffs on them for an hour to let their ears recover, and then brought them back to the playback system and compared the sound they WERE listening to with a dynamic and un-compressed version at the same AVERAGE level, they would immediately hear the improvement 100 times out of 100.

But don't take my word for it: just do all your A/B comparisons at the same AVERAGE PLAYBACK SOUND LEVEL (NOT the same gain setting on your volume knob, and NOT the same peak level on your digital meters), and compress as much as you want.

The whole problem comes from using the compressor instead of the speaker level control. If you compress peaks by 6dB, compare that with the un-compressed version PLAYED BACK 6DB LOUDER (since this is what the listener will do, including you, tomorrow) and see if it still sounds better. You can't make a fair comparison between a compressed and an uncompressed recording unless you ears are hearing both at the same average volume.

Quote:
***Originally Posted by flmaso**n*
Went back and tracked down the "golden ears in one step" part of the thread as review.

Was curious if this was ever revisited, the part about "lots of tricks and psycho-acoustical funny business"...

Given that Hard Rock/Progressive/80's rock et. al (i.e. the guitar god era stuff) is what I've been chasing, thought these tricks should be something I learn to quote in my sleep, LOL!""
=====

Better I think than examples of *good* heavy rock production is to hear something *bad*, that sounded good at the time. Forum-goer Deltones above offered this example:

Quote:
Damn... if there is one thing that exemplify what Yep wrote above, it's this video. It's brutal, but not in a good way.

http://www.youtube.com/watch?v=ZDR7v3Vb1qo ""
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Go ahead and click the link, and think about the following (setting aside whatever you think about Cannibal Corpse):

- The person who videoed it, at the time, thought they were hearing something worth recording (and they undoubtedly were)

- The reason the sound sucks is not because of a problem with the recording equipment. If the recordist had been using a primo mic and recording interface instead of a cellphone camera, it wouldn't necessarily sound any better, it would just be a more hi-fi recording of suck (assuming the same recording technique of a mic held up near the front of the audience).

- Most importantly, this music DID actually sound good at them time, to the listener (let's set aside opinions on death metal here, and just pretend that we're talking about heavy, powerful, punchy being good, and fizzy, weak, papery being bad).

So what happened to all the power, aggression, impact, size, and weight? How did one of the world's most notorious death-metal bands, playing through a super high-budget soundsystem at a large festival, end up sounding like a kazoo with some trebly clicks in place of kick-drum hits?

Simple: it got quieter. In all seriousness, play that same youtube track through good speakers or headphones cranked up to the threshold of pain, and it suddenly sounds like a death-metal band should sound.

Loudness effects are like a dynamic eq that turns down the highs and lows and that suppresses dynamics the quieter the sound gets. This effect is also nonlinear: that is, it becomes more exaggerated the further we get from the target "original" sound volume.

The good news is that it is almost entirely a one-way street: things that sound good quiet tend to sound even better when loud (although the reverse is emphatically not true). So while the above Cannibal Corpse live video only sounds good loud, most Cannibal Corpse ALBUMS actually sound pretty good whether quiet OR loud ("good" in this context meaning "the way a Cannibal Corpse Album should sound", regardless of whether you like the band or their sound).

To your specific question, an 80s prog rock sound is certainly different from Cannibal Corpse, but the same principles apply. In fact, the same principles apply to electric blues or Irish folk or anything at all whose live effect depends to some degree on dynamic sound, which pretty much means all music.

It may be surprising to think that all music is played to about the same peak volume. Be it Death Metal or acoustic folk, when any band or musician plays live, they tend to play right up to the threshold of pain. An acoustic guitar or grand piano easily hits instantaneous peaks that match jackhammer levels at close distance, but because they are instantaneous, they don't hurt the ears or our hearing.

Something like a death-metal or heavy rock band tends to play live at steady-state levels that come close to jackhammer volume, with instantaneous peaks that *just barely* exceed the threshold of pain. And at that real-world volume, a tiny step over the line has a big impact, since it leaves a stinging "slap" that is felt viscerally in the body and on the eardrums.

Perhaps ironically, that means that an acoustic folk duo might very well have MORE dynamic variation than full-throttle death metal band. The former is playing at delicate, conversation-level volume, with big, plucky, sharp transients, while the latter is playing at full-saturation, full gain level with transients that barely register. When everything is already turned up to eleven, where do the transients go?

So what happens when we play these musics back, at living-room volume level, is that the acoustic folk duo sounds like a conversation-level band with sharp peaks, deep lows, clear highs, and dramatic musical dynamics, while the death metal band sounds like a fizzy, midrangey, flat-lined kazoo. Which is the opposite of how most metal bands would prefer to sound. The difference is that we recorded the folk duo as they sound, and then played them back as they sound, but we recorded the death-metal act as they sound, and then played them back maybe 50 dB quieter.

This can be hard to wrap your head around, but it's important:

The folk duo sings and plays at conversation-level volume, with dramatic transients and peaks that occasionally hit pretty high on the dB scale. We can keep those dynamics, or we can compress them, but either way the recording is going to sound natural and normal when played back at conversation-level volume, because THAT'S WHAT THEY ACTUALLY SOUND LIKE.

The death-metal band, on the other hand, plays at threshold-of-pain volume. The musicians (and the front rows of the audience) either wear earplugs or are going deaf. If all we do is record that sound and then turn it down to living-room volume levels, then all the size, power, and visceral impact disappears. More technically, the highs and lows drop out and all we hear is the upper midrange: that fizzy, nasal, "kazoo" sound, with clicky drums.

So the question is: how do we make stuff sound loud and powerful when it's played back at conversation-level volume? In a sense, the entire first third of this thread was about that topic, so please go back and re-re-read if you either didn't read it or skipped over the specific instructions.

The answer is NOT more compression-- flatter sound is the opposite of what we want. It's also not about cranking the lows and highs to offset the fletcher-munson effects: that just produces a dull, blurty sound with hissy fizz instead of the kazoo-like midrangey effect: out of the frying pan and into the fire.

Instead what we need to do is really to re-construct the entire sound from the ground up, to create the big rock "effect" of loudness without the benefit of volume. Which makes heavy rock one of the most challenging genres to record, but also one of the most rewarding: there is nothing like the awesomeness of getting those screaming guitars and howling vocals and churning bass and pounding drums to come through at shopping-mall-speaker volume levels.

Step one is to use gentler, lower-gain, and more midrange-friendly sounds than the band would use live. For starters, take the gain knob on the amp and turn it down about halfway between "clean" and your typical live sound. Same with any distortion or overdrive pedals. The higher the gain/distortion settings are, the less "spank/chunk" you get on the pick attack at lower volume.

Similarly, and perhaps counter-intuitively, turning down the bass/low eq on the guitar amp usually produces a more powerful and dynamic low-end. Cranking the bass knob tends to send a ton of low-frequency stuff to the saturated preamp stages where it gets saturated, distorted, and compressed into a sludgy muck of mwahhy hum. Turning down the low eq knob on the amp tends to open up the bottom end, and to allow a cleaner, more "fender-y" thump and spank to punch through the midrange crunch.

And then we come to drums and bass: God help us all. Unlike guitar players, who tend to be obsessive about tone, many of even the best drummers have little or no sense of sound quality at all, possibly because they are deaf from playing drums. They'll come in with drums that creak and rattle, with loose hardware, old heads with dents and worn-off coating that you can press down on like a mattress, and they'll often have completely bizzare notions of what their drums are supposed to sound like that have no relation whatsoever to how they sound in the real world. Like, they're playing a wood piccolo snare (that sounds great), and they want it to sound like a deep explosion (hello sample-replacement).

Similar with heavy-rock bass-players, especially if they are used to playing through an under-powered rig. There is a reason why a 300W+ 8x10 bass amp is the customary accompaniment to a 50W 4x12 half-stack guitar amp: the bass needs to move a lot more air to sound as loud. A bass player who is used to playing on under-powered amplification (as many are) is often very difficult to record. In fact, next to vocal coaching, improved bass amplification is one of the biggest-difference makers that typically happens when a band gets signed.

All this stuff comes back to the "golden ears in one easy step" section of this thread. There are many pages of iterations, but the bottom line is that you need to make recordings that, every step of the way, sound as good or better than comparable recordings do at comparable volume.

Quote:
Originally Posted by noon
...The 'boom' is a big part of the effect. It sounds fine on my monitors, home stereo, headphones with a good bass response. But it buzzes smaller speakers, like desktop or car...""
=====

Unfortunately you may be forced to make a decision, and to compromise one way or the other on this.

Smaller speakers, especially small, cheap speakers cannot produce very low frequencies at volume (actually, they technically *can*, but it requires so many other sound-quality compromises, and so much expense that such speakers basically don't exist).

There are several ways that manufacturers address this limitation, but one of the most common ways is simply not to: The speaker will effectively work as a mechanical eq that cuts everything below a certain frequency, and it does this automatically. And that's usually a surprisingly effective approach: you may not get the chest thump or shaking floors of a nightclub or cinema action movie, but baritone vocals, guitars, and lower-order harmonics from bass guitar and typical kick drums actually come through pretty good even on speakers that only reproduce 100Hz and up. Even a speaker system with a 200Hz cutoff can sound pretty good with most pop music, although it won't have much in the way of visceral low-end "power".

Poor speaker systems will do a bad job of this cutoff, and will create distortions as the speaker excursion "flattens" on powerful bass notes (better small speakers will still excurse all the way through the low-frequency cycle, but their small size simply won't displace enough air to create audible sound). Bigger problems tend to come from the amplification and signal circuit, which on cheaper systems tends to be under-powered, meaning that deep lows will overload the circuit and distort the *whole* signal, even if they aren't actually reproducing the lows.

This is one of the major reasons for doing commercial mixing on small nearfield monitors, and at low volume. If you know you're making club mixes, or cinema soundtracks, or specialty "audiophile" records, you are free to do whatever you like, frequency-wise: you can count on good playback systems. But if you're making records for commercial release, the hope is that this will be heard not only on high-end systems, but also on 2" TV speakers, shopping mall PAs, car stereos, cheap earbuds, etc.

These kinds of sub-par soundsystems are simply not going to deliver subsonic "thump" and "boom". Which means that deep/subsonic elements are either not going to be heard at all, or are going to be heard in somewhat compromised fashion. Unfortunately the "compromises" that can help them make their way in the world of bad soundsystems are typically ones that may also somewhat mitigate the experience for listeners who *do* have good, full-range systems.

That's where the art and skill of good engineering comes into play:

Simple ways to move subsonic, low-bass sounds into the "small speaker" range are stuff like eq'ing up the second-order harmonics; using distortion/saturation/exciter effects to bring up illusory harmonics that create the impression of deeper sound; using compression to sculpt the attack of the higher-frequency portions of the sound element to make them sound "bigger", or to substitute low-mid "thunk" for lower "thump", etc.

Of course all of the above also mean somewhat compromising the sound and true effect for listeners who do have good sound-systems, but a skillful engineer can *sometimes* thread the needle such that the alterations are minimal, or even so they sound like a net improvement on all sound systems. But not always. Sometimes you have to make a decision.

Quote:
Originally Posted by noon
yep & nerdfactormax,

Thanks. It's somewhat reassuring to think I'm attempting the impossible.

Please take a look at this graphic display of my section [brighter] superimposed over a big drum bit from a commercial CD [dimmer].

The other recording exceeds mine in amplitude at virtually all frequencies, and yet it produces a much clearer sound. Note, the left hand scale is +10 db, so that highest peak is about -8.6.""
=====

For starters, you're doing it wrong. Turn up your track or turn down the commercial track until the "average" curves are the same. Then compare them. Unlearn the notion of peak level as meaningful anything.

Second, your graph looks awfully "peaky". In particular it seems to have declining peaks at 60Hz, 120Hz... and some really big dips. In short, without actually hearing it, it "looks" like it might be a poor room or recording, but there's not really any way to tell from looking. You certainly have a much more midrangey recording than the commercial one.

Last but not least, this chart tells us nothing about dynamics. Yours might be "quieter" because you have big peaks. In that case, just turn up the volume.

You're thinking too hard. You really need to re-read until you wrap your head around the difference between peak and average level, because right now you're comparing apples and orangutans.

Quote:
Originally Posted by sweetbutt
I first want to thank yep and everybody for creating and building this thread.
it's a rare gem.

I'd like to ask something that's been poppin in my mind lately something I want to improve

it's about monitoring-trusting what you hear

but in the very late stage of the process. by this I mean when your mix sounds great at home and sounds "ok" with JUST a bit too much midrange in my car (but any "pro" record sounds balanced..ranging from slipknot to the sea and cake to brian eno), sound great in my mp3player again, sounds a little less balanced in Michelino's home stereo and again sounds nice in my mom's home stereo and so on...

the situation is where you're just about very c.l.o.s.e. to make it sound "consistent" thru several playback systems.
but YOU KNOW it actually doesn't yet.

do you have any suggestion for this particular point of the process?

what I find a bit tricky is that being very close to the final result, spotting those little issues (usually freqs issues...a little midrangy...a little harsh...low end a little too pronounced...etc..) goes towards a trial and error thing I want to minimize and eventually overcome

I just want to trust my ears more and more.""
=====

Man, there is so much stuff I do without even thinking about it that it is sometimes hard to put into words.

Try this, with either a single track or a full mix: turn your speakers or material way down, until all you can hear is the ugly (if it's really ugly, then ugly will usually be the last man standing as you turn down the volume.)

Now start to sweep an eq with a -12dB cut and 1.2Q or greater until the ugly disappears. Or, even better, but somewhat harder, is to take a +12eq BOOST and sweep it around until it's as ugly as you can make it. The Louder=better phenomenon makes this one harder, but if your ears are trained, you'll hear the "ugly" jump out at certain frequencies.

Another way to try this (especially with saturated, full-spectrum sounds like electric guitar), is to turn down the speakers until until you can barely hear the material, and then sweep a -12dB-cut around until the sound disappears. When you turn it back up, whatever was most prominent will be gone. And often, that will be the worst parts: the honk, mud, brittleness, etc.

Always process a little bit less than sounds good. A good rule of thumb is "halfway". If a 10:1 compression ratio sounds good, try setting it to 5:1. If a 6dB boost or cut a particular frequency sounds good, try backing it off to 3dB.

One of the skills that only comes with experience is knowing how things will sound at different volume levels, with different degrees of background noise, on different playback systems. But you can accelerate that learning curve exponentially by monitoring at extremely quiet levels, by monitoring from the next room with door partially closed, by monitoring with a blender or vacuum cleaner running, etc.

Monitoring in demanding conditions such as the above quickly reveals the weakness and ugliness. Dialing out the stuff that sounds bad reveals the stuff that sounds good.

9 times out of 10, if you have, say, some midrange ugliness in the guitars, turning the guitars down to the bare threshold of hearing will leave nothing but the ugly. Same with vocals or anything else. If you sweep an eq cut around until the sound disappears, or if you sweep an eq boost until it's all you can hear, you will identify the offender.

Quote:
Originally Posted by Gizzmo0815
Holy crap that worked like a CHARM. Thanks for this tip Yep, it's right on the money. With a little tweaking and a lower volume I'm finding it much easier to break out what the "bad noises" are compared to the good ones.""
=====

Quieter monitoring = better strikes again!

Even though pretty much every second post in this thread is re-stating it, I cannot stress enough that that the fastest route to "golden ears" is to listen at low, level-matched volume.

All the loud prettiness disappears, and all the ugliness tends to make itself known at low volume.

Quote:
Originally Posted by sweetbutt
...
ok..we've lowered the volume considerably:

apart from evaluating an evident ugliness lying in the recording, I'd like to go a bit more into details, nuances of this process..

maybe is there a way to examine the low end?
or the mid range?...""
=====

I feel like I'm starting to re-state stuff from earlier in the thread a bit, but here are some thoughts, in no particular order, about monitoring quietly:

- Monitoring at low levels tends to depress the lows and highs and to exaggerate the midrange (discussed at length earlier, see "Fletcher-Munson" effects). Conversely, hyped lows and highs tend to create a "loudness effect" (see "smiley eq")

- Louder volumes tend to sound better, lower volume tends to sound worse (also discussed in more length earlier, see "speaker salesman")

- Stuff that sounds good at low volume *usually* sounds even better at higher volume, while the reverse is not true (see almost the whole first half of the thread)

- Many if not most if not all common beginner mistakes come from confusing "louder" with "better". E.g. move a mic closer, the sound from the speakers gets louder. Apply some high-eq boost, the sound gets louder. Add some reverb or delay, the sound gets louder. Add compression, the sound gets louder. Double-track, the sound gets louder. Add some low boost, the sounds get louder. Add another instrument, the sound gets louder. And it is almost impossible not to hear "louder" as "better" unless you are being very careful and/or are fairly experienced at detecting these things.

So there is this big illusion at work throughout the decision-making process. And the beginner spends all night TRYING to work on mic placement, just like everyone says, but lo and behold, the "best" placement always seems to be the mic placed as close as possible in the brightest possible position with the biggest low-ed proximity effect. They slave over the eq, and have read all the stuff about how small changes and cuts are usually better than big boosts, but it all seems like a lie when cranking the lows and highs just makes the speakers pump and sounds so much "bigger" and "better" and more powerful. And they have heard all these people talking about how over-compression kills sound quality, but somehow the harder they hit the limiter, the bigger and fatter the sound seems to get.

And of course, they keep running up against the headroom limitation, so they keep turning down their master output, and then it sounds all weak and quiet again, almost worse, so they bump up those highs and lows a little more, try adding some reverb, maybe it needs some more overdubs, let's try just a little more compression... NOW it's starting to sound good again, but it's also clipping, so turn it back down. It's weak again. Repeat. Moreover, it's human nature to continually turn the volume up a little throughout extended listening: our ears get saturated, just as our eyes do in bright light, and we always seem to want just a little bit more light when examining something closely.

Come back the next morning and play back the tracks at early-morning volume, only to find that your all-night "improvement" session has turned the whole project into a hissing, nasal, lifeless vortex of shit.

Well, part of that is that you were gradually burning out your ears, and part of it was that you kept confusing "louder" with "better". One very simple way to solve both of those problems is to simply constantly keep the speaker volume as quiet as you can stand it. Keep turning the speaker volume DOWN, in other words, and force the underlying audio quality to sound better.

Pirates wore eye-patches for naval combat: covering one eye allowed them to go from bright sunlight to below-decks, flip the patch to the other eye, and still see in the dark. If you need to get up for a drink of water or a trip to the loo in the middle of the night, you can achieve the same effect by covering or closing one eye while the lights are on. Then you'll still be able to find your way back to bed in the dark. Realtors will tell you, when selling a home, to turn on every light in the place: a bright interior looks bigger, shinier, cleaner, and more welcoming, while a dim, shadowy interior looks smaller, grimier, and more cluttered.

Monitoring at low level is the same idea. It simultaneously keeps your ears fresh and sensitive, and it also strips away the "flattery" of loudness.

The point is not to tell you to make this decision or that decision, the point is that loudness clouds your decision-making. Here is an example:

A lot of recording advice tells you to look for "ugly" frequencies by sweeping a sharp eq boost around the spectrum. This is old hat and the easiest thing in the world for an experienced engineer, but it gives beginners a lot of trouble: they crank up an eq boost, and it seems like EVERYTHING sounds better with that sharp boost.

And they're RIGHT: it does sound better, in a straight, instantaneous A/B comparison. Just sweeping an eq around is the same thing a wah effect, or a nightclub DJ using a filter sweep to highlight every portion of the drum loop or whatever. It's sweeping a spotlight over a city street, and when you zero in like that, there's some cool and interesting texture EVERYWHERE, even the nasty parts are kind of cool and gritty.

What the "sweep an eq to find ugly frequencies" tips leave out is that the engineer has already identified the ugliness he's trying to find. He's just using the spotlight to zero in on it, not to simply look at each square inch one at a time to see whether he likes it or not.

So how can a beginner, with no way to know what to look for, do the same?

Well, the easiest way is usually to turn the track down to the lowest volume you can stand. Chances are, a whole bunch of stuff will jump out as sounding not-that good, so start with the most obvious. First things first, and second things not at all, since the second things will become the new first things once the most important stuff is attended to:

Maybe a disappearing/reappearing bassline, or vocal part, for example. Some compression, some fader-riding, some eq, whatever: forget about sound quality: we need to hear the bass and vocals, so just get them audible. Okay, now you can more or less hear the whole part even at extremely quiet volume.

But now maybe the compression or fader-riding has turned some of the quieter notes up and some of the louder notes down, and maybe you've got a lot of flabby, woofy, cheap-sounding lows and some notes that are just a pointy metallic upper-midrange or whatever: it's audible, but all unbalanced. So, keeping the volume as low as you can stand it, find those flabby/woofy frequencies with an eq sweep and drop them by a few dB (again,using the "halfway rule"-- adjust until it sounds like a clear improvement, and then bak off to halfway). We might need to re-visit the compression as well: maybe you solved the audibility problem just by cranking the compression and now it sounds flat and lifeless, so maybe we need to go back and do a bit of fader-riding or envelope-drawing, and look for a "tighter" attack/release/threshold combination might give a bit more punch and a bit less flab. Whatever. Same with those graty, raspy, edgy highs: if you turn the volume down way low, they'll jump right out, and a quick eq sweep will tell you where they are, once you know what you're looking for.

Now we're making pretty drastic improvements in the first 20-30 minutes, instead of spending all night just randomly turning stuff up and down. And the coolest part is that this learning tends to "stick" and permanently improve your ears pretty quickly. I.e., now you can turn the volume back UP, and see how the lows hold up at a healthy playback volume, but you'll still "hear" the changes you made.

Not sure how much that answers your question.

Quote:
Originally Posted by ringing phone
Well...I was thinking about why my recordings sound like ass...and on top of some rather mundane issues...my recordings sound like ass because basically I am not a very good player...""

=====

I suspect there are a lot of people with a similar problem.

The good news is, once you start practicing for sound, as opposed to merely practicing the mechanics of playing the notes, improvement comes in giant leaps and bounds. Your "sound" can often catch up to your "mechanics" within a week or two.

That said, it's really, really important to start with the basics of instrument setup, etc. A cheap instrument with a good setup can often get 99% of the way to a great instrument. That said, it's helpful to have a basis of comparison. Every musician should, from time to time, try out a bunch of different instruments, even if it's just trying stuff at the guitar store that you can't afford to buy, just to have a sense of what the different tools are good at.

A lot of musicians, especially guitar players who practice electric guitar "unplugged" and without accompaniment, can spend a lot of time and effort learning difficult passages, etc, and can become nominally very competent players, but you plug them in and set them up to record, and they just don't sound good.

Part of it is voicing and finger technique, part of it is poor dynamics control, a BIG part of it is often sloppy or variable timing: slowing down on difficult passages, speeding up on easy ones, pausing at awkward chord transitions... AAARGH! That stuff ALWAYS comes out sounding amateurish and bad, no matter how cool the material is.

You don't actually have to be a very good musician to be a rock star. But you should sound good, even if you're just playing quarter-note strums of three-chord progressions. In fact a lot of hit records have been made that way.

Quote:
Originally Posted by adXok
Captain Damage,
I oppose your opinion. In fact a Mix made with (on) field monitors should have a warning label:

!!! Not recomended listening this mix with a headphones !!!

And nowadays a majority of people using Walkmans (USB), iPods and so many mp3 Players listen thier music on what? That is right - on headphones. Good, bad, excellent, in-ear, opened, semi-open, closed... doesn't matter. You get used to listen to what headphones you have in hand.""
=====

Given that essentially every single commercial record ever made was tracked, mixed, and mastered with speakers and not headphones as the primary monitoring system, I think it's pretty safe to say that you can make records on speakers that will sound good on headphones.

I've spoken on this topic elsewhere at more length, but in my opinion, it is very, very difficult to make records with headphones as the primary monitoring system. Nothing ever seems to sound the same on any other set of speakers or headphones as it did on the phones you tracked or mixed on. Meanwhile, the reverse actually works quite well: records made on good speakers in a good room sound good on pretty much anything. Headphones always seem to have a sort of "one-way glass" effect, for a lot of reasons that have been discussed and debated endlessly.

Everybody WANTS to find that magical pair of headphones that can replace speakers, and god knows that people and manufacturers are trying, but for now, it's pretty safe to say that nobody who has the option chooses to monitor through headphones instead of good speakers. It's not like digital vs analog, or plugins versus hardware, or any of the other raging debates where there are lots of credible professionals on both sides: it's such a complete consensus in favor of speakers, even among people who would rather use headphones, if they could (which is pretty much everyone-- we'd all rather be using headphones if the results were the same).

that said, if headphones are working for you, then there's no reason to let anyone tell you they're not. Somebody has to figure it out first. And headphones do certainly have a role in the studio, for checking details, careful corrections, etc.

The trickier question is headphones VS *bad* speakers, or really bad rooms. That's harder to say.

As I said at the very beginning of this thread, I think the single most important investment any studio can make is in room treatment and decent monitors. There is a work-around for pretty much anything else, as long as you can trust what you hear. My 2 cents, anyway.

____

Quote:
Originally Posted by adXok
My God!
Yep, if you were making just an audio or video samples of what you are talking about this thread could be more clear, meaningful and... short!""
=====

I've actually thought about this quite a bit.

The hardest challenge is controlling the listener's playback level. In a sense, you don't need my advice or examples or anyone else's. All you have to do is listen to the radio and you can hear the entirety of commercial recording arts for free. Everybody knows what a great record sounds like. And these days you can often find somewhere to download the raw tracks.

The hard part is closing the gap, connecting the dots that show how someone knew which sounds to get, how to capture them, how to process them, how to balance and mix them, how to make them sound more good and less bad, etc.

There are also a metric ton of variables in the listener's playback system that I don't know how to control for. Making "perfect" tutorials for the four people who might actually bother to sit down in front of a calibrated, reliable monitor system and work through them systematically is hardly worth the effort.

I need to figure out a way to illustrate stuff like loudness effects and 2dB eq changes and transparent listening in a way that will translate across lots of playback systems and environments, and that will be usable and digestible in small chunks. Otherwise it will sound like me twisting a bunch of knobs that sound like nothing, and then having a final mix that sounds way better. I have experienced exactly this problem firsthand, trying to listen to audio examples, etc, on laptop speakers or in the car or whatever. The problem is the playback system is already hashing and compressing and re-eq'ing everything, so it sounds like nothing. Meanwhile, if I'm actually going to fire up the studio and sit in the chair, I'm not going to spend that time listening to a CD of somebody making 1dB changes to pink noise (maybe I should, but I don't).

Suggestions are most welcome.

Quote:
Originally Posted by flmason
...But is it just me, or does the above point completely contradict the oft heard theory, "...you need to get the sound you want in the room, coming from the amp before anything else... if it ain't happening at the amp... it ain't happening...""
=====

Yeah, it kinda does.

Here's the thing: if you take a super-accurate omni-directional mic (the Behringer ECM800 is a good one for cheap), it WILL sound just like what it hears, IF you play everything back at the same volume through good speakers.

IOW, if you set up your live sound, and then put an omni reference mic roughly in the same part of the room as your head is, and hit record, you'll get a pretty perfect recording of what you were hearing while you played live in the room. But only if you turn up the monitors to deliver the same volume level as your half-stack. And that requires some pretty powerful monitors-- you'd need something like 1,000 watts of clean stereo gain to reproduce the volume level of a 50 watt half-stack.

Try this experiment with your "live" sound:

- Get it sounding the way you want at live volume.

- Now turn down the master volume control to about conversation-level, or about as loud as you expect most people to listen to music in their cars, or in their living rooms or whatever.

- Now, assuming you plan to have two tracks of guitars in your recording, turn it down half as loud as conversation level (we have to fit two tracks of guitar in that volume, remember).

- Assuming once again that we also need to include bass, drums, and vocals in the mix, turn your amp down to half of the above volume, so that we are now at one-quarter of conversation-level volume. By this time, you'll probably be hearing more of the pick and string noise than amp sound, so you'll need to get someone else to play the guitar: that's what your "live sound" sounds like in a recording. I bet you have all the same complaints that you have when you stick a mic in front of your amp: weak, fizzy, muddy, etc.

Volume changes the sound, a lot. If you record any really loud sound, say a gunshot or a firecracker, and play it back at living-room volume level, it sounds like a midrangey little pop, like someone popping bubble paper. Similarly, if you record yourself popping bubble paper and then play it back at ear-splitting volume, it will sound like a cannon. If you crinkle a cellophane candy wrapper in front of a mic and then play it back at high volume, it sounds like a roaring inferno.

If you simply put a reference mic in front of a jazz trio, you'll get a pretty good recording, because they're usually playing at something close to living-room volume. That similarity of volume works in your favor in two big ways:

- One is that the band is actually playing a sound that already works at normal listening levels. So half your work is already done by the musicians themselves, since what they are playing and how they're playing it already sounds musically and sonically appropriate, if they're any good at what they do: the drummer is playing with a light touch, working the little pop and sizzle, the upright bass is working the speech-level woodiness and finger tone, the piano or guitar is playing clean, delicate, articulated lines, etc.

- The second is that the texture, impact, and frequency balance doesn't change much when you play it back at normal listening levels-- it sounds the way it sounds.

On the other hand, if you left that exact same mic up in front of the exact same stage, and swapped the jazz trio for a heavy guitar act that plays at earplug volume, the exact same things are working against you:

- The band has a sound that is built around loudness: the drummer is bashing every kit piece and putting out explosive, bone-shaking SPL, the bass is pounding out massive floor-rattling impact, the guitar is screaming a wall of ear-splitting saturation... The whole band is using volume as a musical effect, which is not going to sound the same at conversation level.

- When you turn everything down, the sound changes drastically: the explosive drums turn to little paps and pops, the crashing cymbals sound clanky and tinny, the roaring bass turns to warbly mud, the howling guitars become a fizzy little nasal thing, and the whole thing goes from gunshot to bubble-paper.

The ocean and the sky are some of the awesomest things to look at, but make for some of the boringest pictures: a 4x6 print just doesn't capture the size, depth, or hugeness unless the photographer is quite skilled at knowing how to frame the picture in such a way as to achieve a sense of scale.

And that's what recording loud music is all about: creating a sense of scale, an impression of loudness and sonic size in a small sonic space.

Quote:
Originally Posted by fly
I have a few few questions :

Does the difference in volume playback mean that I'll never get anything close to the tone I want with a 40W amp?""
=====

You're thinking too hard. It means that things sound like what they sound like, and you can't make them sound otherwise by arguing or proving logically what they "should" sound like. That's the problem with relying on recipes and gear reviews and everything else: you have to use your ears to hear what you're actually recording, you can't just "do everything right" and then cover your ears and hit record. (I mean, you *can*, and a lot of people do, but then you'll be complaining about no matter how much you follow the instructions, it still doesn't sound good)

Quote:
I hear about the proximity effect often, but only for singing . It seems to be the norm to place a mic just next to the grill cloth on amps . Aren't guitars affected by the proximity effect as well ?""
=====

proximity effect happens anytime you use a directional mic close up. Whether it's good or bad is an open question. One thing proximity effect does it to boost the highs, lows, and detail, and somewhat increases the "loudness" and fletcher-munson effects. So it might actually HELP your quieter "recorded" sound to sound closer to the louder "live" sound, compared to a more accurate and neutral recording. Or maybe not.
____

Quote:
Originally Posted by SuprchrgedSi
...What happens when we're recording instruments that use the same frequency band and harmonics but we need multiple parts, i.e. moving guitar harmonies like Brian May or Metallica?""
=====

Wow, awesome question.

First off, you are right to eschew panning as a fix for this. In fact I like to start with mono mixes and gradually pan them out as it nears completion, for a whole lot of reasons. While this method is certainly not mandatory or universal, it is safe to say that a mix that depends on good stereo is not going to translate well on a lot of systems.

One thing you might notice if you listen to harmonized guitar parts by bands like Queen, Metallica, or Thin Lizzy is that the harmonized parts tend to happen either when the singer is not singing, or when the singer is singing along with the harmonized parts. I daresay this is a deliberate artistic and arrangement decision.

Earlier in the thread I talked about the role of the bass, and said something like: "whatever instrument is playing the lowest note is the bass" in reference to the ROLE that the bass plays in setting the tonal foundation. And "note" may have been too strict a term, it's more like whatever melodic line is playing in the lowest register is going to define the tonal movement of the song, whether deliberate or not. So it's usually good for that instrument to BE deliberate about setting the tonal foundation. I.e., if the bass is playing lead, then something else is playing the "base", and it might not be doing it right.

Something I haven't really talked about is the role of the "lead", which is every bit as important. The "lead" in a piece of music, asit is heard by the listener, has nothing at all to do with what you name the band members in the liner notes. You can call anyone you want the "lead" vocal or "lead" guitar, but the listener is going to hear the instrument that is actually playing or singing the musical lead.

The "lead" part in a piece of music, or in a section of music, can be roughly defined as the part a person would sing, hum, or whistle if they were "singing the song" (or humming or whistling the "tune"). In classical music it would typically be called the "melody", which can be a bit confusing since there are often multiple things that could be considered melodies going on simultaneously, but most conventional compositions have melodies, and then they have THE melody. The rest effectively amounts to harmonies and accompaniment, and the listener usually has little trouble identifying and humming along with THE melody.

In jazz, pop, and popular folk music, the most common type of notation is a "lead sheet", or a musical notation of the vocal melody, along with chord symbols to indicate the approximate accompaniment. In early rock and roll, country-western, and a lot of folk music, the instrumental solos tend to be basically either riffing on or outright playing the vocal melody while the singer takes a breather.

The "lead" is usually the most conspicuous material in the upper midrange. Even if the singer is a baritone or deep bass like Johnny Cash, the rest of the instruments tend to leave space for the upper-midrange articulation of the vocal. Similarly, if there is a bass solo, it's almost mandatory that the guitars etc either drop out, or play very minimal accents, so that the listener can hear the articulation of the bass clearly.

An old rule among studio and tour musicians is not play over the lead. Meaning, don't play in the same register as the singer while the singer is singing: you can hear this pretty clearly in most pre-Beatles music: the accompanists either stay out of the midrange entirely, or play only subdued accents or harmonies while the singer is singing. Almost any horn section is a good illustration: they might play short stabs or have a low-register chord or "pad" during the vocal, then they blare out the cool stuff between vocal passages.

Human hearing is conditioned for survival to immediately notice two things: one is sounds in the register of a crying baby (the same register most lead vocals and instrumental solos, not coincidentally), and the second is movement or change.

Repetitive sounds, or sounds in the low registers or higher birdsong registers, tend to recede in our conscious awareness. This is why you can have a churning saturated guitar riff playing over and over, and a slippery, funky bassline, and wailing high-frequency improvisation, and still hear the singer clearly. It's also why, when you move that wailing improvisation into the vocal/crying baby range, or when that churning, saturated guitar or organ part starts playing complex or non-repetitive stuff, it immediately becomes distracting and hard to fit in the mix. It now sounds like two crying babies: the listener can't hear both and doesn't know what to pay attention to.

So the simplest solution is the arrangement: only have one lead at a time. If three guitars are playing a harmonized lead, what is the point of having a rythm guitar part? The drum accents will still poke through just fine, as long as they are loud and dynamic, which they should be anyway, and a three-guitar lead is a perfect time for the bassline to recede into a simple tonal foundation part, and for synths, organs, horns, etc to play a supporting role of pads or stabs. And there is usually no reason to try doing a harmonized guitar lead at the same time as the singer is singing anything more than oohs and ahhs.

That's the way people did it in the days when they actually had to play the material live, and when the only mixing was what the musicians could achieve by altering their parts and playing technique, and it's almost certainly still the best way.

That said, a lot of modern music requires the mix engineer (either live or in the studio) to perform a lot of the functions that used to be performed by either the arranger or by the musicians themselves.

But the same principles still apply: something has to be playing the lead. Just as the lowest instrument is setting the tonal foundation whether deliberately or not, so the listener is going to hear something as the "sing-along" or "hum along" whether we intend for them to or not. The listener is GOING TO HEAR a "melody", and typically it's going to be whatever displays the most sonic movement in the "crying baby" range.

If there are multiple instruments or sonic elements all playing competing melodies with motion in that range, then it tends to sound like an amateurish cacophony of musicians who aren't listening to each other, who are all just playing on auto-pilot. In that sense, *some* commonsense arrangement decisions are always going to be necessary.

A lot of modern guitar-driven rock and top-40-style pop and dance music kind of turns the old rules of arrangement on their head, often to pretty cool effect. It's not unusual to hear a riff-rock band where the vocalist sings fairly monotonous steady-state type stuff and where the sonic interest and motion is provided by killer guitar riffs, and similar stuff often occurs with dance-type pop music, where warbly synths and cool loop textures come to the fore against an accompaniment of multi-tracked vocals with a lot of the mids scooped out. This is a very different approach to conventional pre-beatles pop arrangements where the vocal melody was always the obvious front-and-center lead that the other instruments left room for.

That said, and for reasons that are hopefully becoming fairly obvious, it's almost impossible to a fit an expressive, improvisational jazz- or R&B-type singer over a furious thrash-metal riff-rock band. That kind of vocal needs room and sonic space to breathe, at any tempo, and it's hard to tap-dance in a mosh pit.

Bringing this all back around to harmonized guitars: guitar harmonies are almost certainly going to be your lead instrument, whenever they occur. And they *are* going to soak up all the midrange and push everything else to the background, but that's kind of the point of harmonized guitars, isn't it?

My guess is that your problem is not with the harmonized guitars, per se, but with other stuff that probably shouldn't be trying to play at the same time as the harmonized guitar leads. A lot of guitar-rock bands tend to fall into a sort of musical auto-pilot: you have a verse riff, a chorus riff, and maybe a bridge or pre-chorus or whatever, and all the musicians tend to become soft of loop-machines just playing the pattern over and over.

A smidgen of thought put into arrangement can give a ton of new life to static songs. Parts SHOULD drop in and out. Breakdowns (where one or more instruments stop playing) are a fantastic way to draw attention to different parts of the sound (have a cool bassline that seems to disappear behind the guitars? just drop the guitars out for a measure or a verse, and everyone will get to hear the sick bass chops and coolness. When you bring the guitars back in, it will all sound even better, because now the audience will be aware of the cool bass beneath the surface).

Similarly, in the interests of musical taste and discretion, don't try to have harmonized guitar leads playing against a competing vocal lead. Just make them separate sections of the song.

Last and least, if you've made it this far, then I suppose you deserve to hear the secret trick for making everything fit together when nothing does: run all the tracks through a distortion pedal and scoop the mids. I'm not joking-- vocals, drums, bass, synths: just run each track through distortion. This is how 90s industrial bands got massive tracks of busy, layered guitars, synths, and vocals to all sit together and sound loud, and it can still be heard on tracks such as Kevin Rudolph's "Let it Rock". Won't it sound distorted? yes, it will, and it's a really good way to make music that will sound over-produced and dated, fast.

Quote:
Originally Posted by flmason
...

2) I'm not sure about the gunshot thing. I've been involved with firearms most of my life. Many of the vids on Youtube, shot with any old video camera sound like real guns to me, and I'm sure few of those folks are sitting around engineering their vids to "sound like a badass gun", LOL!...""
=====

That's because those cheapo mics were distorting/overloading/compressing the loud sounds. Which is exactly how to make a pristine high-headroom recording of a gunshot "sound" loud.

"Hi fi" recordings, such as can be heard in documentaries, behind-the-scenes movie clips, etc illustrate the "little pop" effect better.

PS-- respectfully, it might be helpful to move some of the guitar gear arguments to a different thread. Not that it's a bad discussion, but some of it might not be directly relevant to the whole "how to get the best sound from what you have" thrust of this thread.

Gear is certainly relevant to sound, but as I said in the beginning, there are a million places on the web to discuss it, and the more that gear comes into the picture, the more these kinds of discussions tend to peter out or drift into unfocused opinion boards...

Quote:
Originally Posted by PooFox
this is truly a rockin thread, but i just couldnt resist reanimating this comment from the first page...

(stuff about garbae-in-garbage-out omitted...)

i think this is really only true if you want the sound at the other end to sound like it did when you recorded it. ive successfully turned my hand whacking the front of a $10 mic into a fairly convincing and epic drum of doom...""
=====

I don't think there is any contradiction there: "Garbage" is stuff that you don't want, and don't have any use for.

The relevant bit for the recordist might be re-stated as: if you don't like the sound you're recording, fix the sound.

IOW, the "danger" is recording stuff that sounds bad, especially just by blindly following rules or "recipes", and somehow expecting it to sound good later.

That's one of the reasons why I started with all that stuff about testing and trusting your signal chain: it's less important to have a great signal chain than it is to trust the signal chain that you have. If you know that you can take a CD of "Kind of Blue", send it out through your speakers, record the output with a mic, and end up with a pretty good-sounding recording of "Kind of Blue", then that knowledge (hopefully) gets you off the train of constantly thinking that you must need a better this or that.

Maybe you would benefit from having a better this or that, but you'll always benefit more by getting over "gear anxiety" and focusing on the fundamentals. Size of the boat versus motion of the ocean and so on.

Quote:
Originally Posted by PooFox
... limitations are only bad when you accept them perhaps?...""
=====

Interesting post.

There is a VERY widespread notion among accomplished recording artists and producers that "limitations are good", even though almost none of them willingly choose limitations, other than in fairly abstracted ways (e.g., they might decide that they're going to do a record all live in one room, but that doesn't mean that they won't spend a year on tracking with a million dollars worth of rented amps).

There is a notion that has been articulated by the football coach Bill Parcells, and also by George C Scott's character in the film "The Hustler". It's the notion that people will lose, if they have an excuse for losing.

If I'm an unsuccessful but talented musician, it is fairly easy for me to construct a narrative where my talent is legitimized by saying that this or that famous rock star got there by virtue of having a million-dollar producer or a bunch of fancy and expensive preamps, etc. And that narrative makes it easy for me to feel not only that I can't compete with those records, but that I don't have to and shouldn't be expected to.

This is a fork in the road of anyone attempting anything ambitious: My point with the "trusting your gear" stuff is to divert AWAY from the "I can't achieve that" road and ON TO the "my gear can do it" road.

Dr Dre didn't have the advantage of million-dollar gear when NWA broke big. Bruce Springsteen recorded at least one album on a cassette 4-track. Practically any of the producers of the legendary records of yesteryear would have killed to swap with rigs with half the people on this forum.

Limitations are neither good nor bad, and they're not even limitations. You can buy a $50 mic from behringer that will record fool-the-ear accurate recordings (albeit with a bit of hiss) using almost any phantom-powered audio interface on the market. That means, in a very literal sense, that if you can make good-sounding music, all you need is a $50 mic to record it. Frankly, for that matter, an

SM57 plugged into an XLR-to-phone plug transformer will pretty accurately record whatever you point it at.

The problem is not that the gear is failing to record the music, the problem is that musicians are not making their music sound the way they want it to sound, and they're blaming the mics and preamps and so on. Which makes it easy to keep up the narrative that the problem is not with the singing or writing or playing or arrangements, it's that we don't have the right tube preamp or whatever.

My point is not to beat up on musicians, but to liberate: there is a massive industry telling you every day that your talent isn't good enough, that you need this, that, or the other: that great-sounding music is not only insufficient, but impossible to achieve-- everything has to be dependent upon recording gimmicks and tricks and vintage this and tube that and phase-linear, opto-circuit doodads... you'd think that good music didn't exist until 50 years ago.

Quote:
Originally Posted by PooFox
...ive never heard of an XLR-to-phone plug transformer. whats that do? i always felt like sm57 was lacking in low end reproduction...""
=====

It's one of these:

http://www.radioshack.com/product/in...ductId=2062443

And it does exactly what it looks like. It was a staple in the days of 4-track portastudios that only had 1/4" inputs. It doesn't help sound quality.

Quote:
Originally Posted by flmason
Hi Yep,
At some point I don't know how you can divest the two. Guitar, electric guitar, specifically big rhythm electric guitar is so endemic to the popular music of the past 4-5 decades that it eventually has to become a topic...""
=====

Guitar is certainly a topic, and how to record it and mix it is certainly a worthwhile topics in the context of this thread.

My only "rule" that I asked at the beginning (way before I thought this would get 180,000 unique hits) was that we leave off debating this gear vs that, and focus on techniques, not because gear doesn't matter, but because there are a bajillion places on the web to read about and debate gear, and not a whole lot (that I knew of) to focus specifically on technique.

My hope is that this thread has been and will remain just as elucidating and useful regardless of the kind of gear you use. That's a different thing from saying that gear is not relevant. I've expressed some pretty strong (and rather controversial) opinions on gear elsewhere on these forums and others.

I have had specifically in mind throughout this thread that budgets vary, and that, to a lot of people, $50 is a lot of money. That's kind of been my informal threshold: if it requires more than $50 expense to try/experiment, leave it out. Of course, to some people, even $10 US is a lot of money. To others, a grand or two here and there is an incidental expense. Everybody's different.

But there is an awful lot of stuff that is important and valuable to know, that has nothing whatsoever to do with how much money you spend. Nirvana famously made an album for $600. Bruce Sprinsteen made one on a cassette 4-track. Countless "world music" acts from poorer countries have made great-sounding records on a shoestring. I have heard some church choirs recorded into garageband or a digital portastudio with a couple of cheap mics that sound devastatingly beautiful.

So it's possible to set aside the gear debates and still talk about how to make good recordings, which is what I'd like to do here. I'm not a mod or an admin, and I couldn't ban posts if I wanted to (which I don't), but I would encourage you to link to good discussions of "best guitar amp for recordings" etc, rather than taking them here.

____

Quote:
Originally Posted by flmason
I once saw a video ad for some distortion plugin where the demonstrator started with a fairly bland pop sort of song and gradually added some distortion to each track. At the end it was considerably more "produced" or "pro" sounding.

Just curious, as you are presenting it, do you mean each individual track distorted separately, or just the master two channel mix?""
=====

either or both.

Quote:
Went and listened to the "Let it Rock" track over on Youtube. Couldn't really pick out heavy distortion, like say a Rat pedal would do. So I'm guessing we're talking something a little less extreme?""
=====

You might be surprised. Try running a vocal or drum track through a distortion pedal with a full mix playing, and you might be blown away by how much distortion you can get away with before it starts to sound "distorted". That vocal probably has as much distortion as a typical AC/DC guitar track does.

____

Quote:
Originally Posted by Smurf
A quick question, all of the "aural exciters", like BBE, X-Cita, etc, is this not what they do? Don't they add slightly distorted harmonics to the top end of the signal, or have I read wrong about these units?""
=====

A lot of them do a lot of stuff. But the basic "harmonic exciter" as first popularized Aphex pretty much does what you said. It rolls off the high end and then regenerates it with harmonic distortion. The idea is that this newly-generated high end will be better phase-aligned.

All such effects should be used on a purely "try it and see" approach, with a healthy dose of caution against fatiguing hype effects.

Quote:
Originally Posted by sweetbutt
...
yep what would you say about those "polite" and balanced low ends?
what to look for to achieve them?
or whatever passes in your mind

and also
if I raise the bass knob and, in a great, balanced song the low end (i.e. kick-bass) remains locked and firm, and in a not so great, balanced mix a kick starts to "bump out" a little too much over the bass line..is it a lack of compression or a matter of Eq in the kick track? or something else?""
=====

Given the slightly vague terminology, it sounds like your car has a bass-light system, and that the mixes you like are generally well-made records.

"Polite" bass is a good way to put it. The "obnoxious" bass is probably over-compressed and over-hyped. If you check it on good monitors and/or a spectral analyzer, it's probably hyped and compressed to "thump" or "boom" artificially at some frequency in the 100-300Hz range. IOW, the mix or mastering engineer is probably trying to "improve your stereo" by faking big bass.

Next to "ringing phone" HF distortion, this is my biggest pet-peeve in modern music production: records where the bass is already overloaded and over-compressed, so that it's impossible to get it to sound good, even on a good system. And ironically, it tends to sound even worse on a bad system (which is what it's trying to correct for).

You know when you pull up next to a car and hear this buzzy low-frequency "BUUHH BUUHH BUUHH" from an overloaded cheap stereo? What's happening is a combination of compression, saturation, and "one note bass" (discussed near the beginning of the thread). Sometimes this is just a fault of economics: cheap stereos, from the preamps, through the power amps, through the speakers simply don't have the power-handling to reproduce powerful, slow-moving, current-sucking, speaker-excursing low-frequencies at volume. And I can hardly fault a teenager in a '93 Tercel for not being able to afford a thousand-dollar stereo. Their poor little stereo is doing its best, and if the best it can come up with is to reproduce all bass notes as flatlined, distorted, 200Hz square waves, then you have to sympathize.

Similarly, some more expensive systems deliberately do something similar, albeit usually more tastefully, without the obnoxious overload. My TV Bose surround system has more than a touch of "one note bass" but it's a lower, less discernible frequency (probably around 50 cycles or so), and it's not distorted or buzzy, more thumpy, and can actually be quite enjoyable to listen to (although it would be terrible for monitoring on). It creates a bit of illusion of lower-frequency production than it actually delivers.

What drives me nuts, though, is when producers build "one note bass" effects into the track-- it does exactly what you're talking about: overloading under-powered systems (which already do that anyway), and even worse, making it impossible to get a natural, smooth-sounding, dynamic low end even on systems that are capable of it.

note that this distortion that I'm talking about is not the rich fuzzy growl of an Ampeg SVT, nor the burpy, mellow fatness of a flatwound P-bass through a vintage preamp, it's the dull, lifeless BUHH BUHH BUHH of a cheap stereo, embedded right there in the record, probably through multiband compression applied to compete in the "loudness race". They flatline the bass right up to the limit of digital clipping in an effort to make it sound "louder", which wreaks havoc on cheaper stereos that only add more distortion in a struggle to keep up with the massive waveforms, and that negates good stereos by making the bass sound flat and dull to begin with.

Once again, use good monitors, and level-matched listening, focusing on the midrange at low volume levels, and you won't have to worry. Your car stereo won't reproduce the ultra-lows, but what it CAN reproduce will still sound good at your preferred tone-knob and volume settings, because the underlying material is well-recorded.

____

Quote:
Originally Posted by sweetbutt
could you briefly explain me why the midrange? (even tho you we discussed about the low end...find this really intriguing)
midrange seems like it's a big key factor in a "congruent" mix.

if you were behind the console, evaluating that midrange at low volume levels, could you point me 3 main characteristics of it?

1- it has to be _________________
2- it has to be _________________
3- it has to be _________________

what I miss is that what exactly I should be looking for after I lowered the volume and listening to that midrange.""
=====

Just good sound.

Have you read this thread from the beginning? I wish people would start at the beginning, and reply to older posts if they are unclear. Honestly the most useful and detailed stuff in is in the first half of the thread, and a lot of the last half is kind of wankery and philosophizing (me as much as anyone).

The importance of the midrange and low-level listening were e3specially covered in a great deal of detail, and argued over, including lots of screenshots of spectral analysis of various popular recordings, and quite a bit of detailed technical and psycho-acoustical theory.

It seems like this thread is kind of starting over again, but with lower quality. If you want my short opinion, it's to monitor at low volume, focus on the midrange, and make it sound good. If you want details on the hows and whys, I wrote some dozens or hundreds of pages earlier in this very thread. If some parts of that aren't clear, feel free to reply to older posts for clarification.

Quote:
Originally Posted by sweetbutt
I actually printed the whole thread in a nice paper-copy

I'm goin thru it very carefully. as I previously said it's a rare gem.

I saw some spectral analysis shots when I first flipped thru the pages but I didn't get there yet..
I was aware of this so I was trying to ask something the more specific possible but I didn't manage then

so I apologize to ev'body if I was asking stuff already mentioned

I'll write back when I get to the end of this enlightning lecture""
=====

Just to clarify, my point was not that everyone must necessarily read all my posts from beginning to end, only that this thread started some time ago, and I did start it in a specific order and sequence in order to cover stuff from roughly most-important to least-important. So by all means, if you start at the beginning and reach a point where it's no longer useful/interesting, drop it and start a new thread. Alternately, if you hit a post from page 3 that you disagree with or don't understand, feel free to dispute/question it. But I don't know that it's very helpful to have this thread turn into an "ask yep for advice" subforum.

Quote:
Originally Posted by GregHolmes
I believe anyone can learn how to sing, if they are willing to listen, critique, and change. Some people can do that for themselves, but most will need a teacher.

Most importantly, you need to find your own voice before you copy anyone else.""
=====

Yeah I agree with this. Or at least, most amateur singers could sound a LOT better than they do. People who put a ton of effort into practicing their instrument and tweaking sounds and shopping for gear often put shockingly little effort into improving their singing voice.

And the vocal is definitely the most important part of any song. The band can never sound better than the vocal track. If the vocal sounds weak, hesitant, nasal, muffled and tuneless, then nobody's going to hear anything else-- the whole track sounds crippled.

And contrary to popular theory, most of the singers on the charts are actually quite good singers (even if they were hired for their looks). Yes, Christina Aguilera comps 100 takes, and yes, it's auto-tuned, but that's to perfect an already great performance. People point to stuff like that as evidence that singing on modern pop records is all studio trickery but then ignore that their favorite technical metal band comps 100 takes of guitars and layers six tracks of every part and so on (or whatever).

Watch Pink's performance from the grammys, or any of the live shows on cable TV or whatever: sure, the songs might be kind of dumb, and yes, these people are often selected for their looks and not just for their talent, and yes, they enjoy the benefit of a million-dollar processing (and probably auto-tune), but if you took them offstage and immediately handed the same handheld live mic to your average garage-band singer it would be an embarrassment.

the reason I did all the stuff at the beginning about trusting your gear is exactly to put a stop to all the mental excuse-making that holds people back. When I encourage you to stick your best mic in front of your best speaker and try to get a reasonably accurate-sounding recording of your favorite CD (which you can, I guarantee), that's a nice way of saying that the mic is not the problem.

There is a lot of ego-protecting that goes on in these kinds of discussions: pros with million-dollar studios watching their business evaporate lash out that the cheap gear revolution and call mackie mixers "shit on a stick" or whatever, and amateurs who have maxed out the credit cards on fancy preamps smugly join them. Budget amateurs, meanwhile (often the same who are quick to deride expensiver gear as placebo-effect waste of money) lament that pop stars are just hired for their looks and made to sound good with fancy machines.

My point is not to make anyone feel bad, it's that the ego-protection is holding you back. It's not just the hours and dollars wasted on trying out a bajillion different plugins, it's the mental erosion of your focus on and confidence in what's important.

It's amazing how much further you can get by putting one foot in front of the other, than by thinking, arguing, and experimenting with the best ways to get there.

Quote:
Originally Posted by Lowell Mather 5150
Yep,

Considering high passing/hi pass, what are your thoughts on the evolution of this techique say from, the sixties up to 1994 - I'm speaking in terms of individual tracks, but also the stereo mix could be relevant. After I started researching and mixing more, it seems like some guitar tracks in particular are high passed rather liberally - others, judiciously, especially when the guitar is out front. But even when the guitar is out front, it seems like some guitar tracks (I'm talkin Van Halen 1984, i.e. Drop Dead Legs)are barely high passed. I know the material as a whole is relevant, but when it comes time to hi pass, if it all, no matter what instrument, what has been the evolution in terms of the technique from the time periods I proposed.""
=====

I honestly have no idea.

Home stereos and transmission medium (especially the shift from AM to FM radio, and then the move from LP/cassettes to CD) have generally expanded the frequency and dynamic range available to most listeners, and production techniques have tended to follow suit, but only a little (up until the recent "loudness wars" reversal, anyway).

For reasons described in detail up-thread, a super-extended frequency spectrum tends to become self-defeating past a certain point: extreme low-frequencies have a way of robbing headroom and mucking up the speakers and room acoustics, while extreme high-frequencies tend to exaggerate hash, hiss, and digital artifacts on the playback system, often producing a more veiled, LESS clear sound.

Moreover, most musical instruments don't really use or benefit much from frequencies at the extremes of human hearing. High C is what, like 8kHz or something? And a bass guitar bottoms out at somewhere around 35Hz, I think, and even then sounds practically atonal at the fundamental. I mean, try tuning anything to 35Hz pure sine wave and see where that gets you.

As for individual instruments, it's impossible to say anything. You might record he exact same player, guitar, and amp, with the exact same settings, using an AKG 414 mic pointed straight at the speaker cone and 6" back from the grill, and I might record the same with an SM57 shoved right in the grill off-axis, and we're going to have massively different frequency profiles without even touching an eq.

Which is exactly why it's pointless to say "for best results, cut guitar by 4dB at 200Hz" or "to sound like Eddie Van Halen, cut X by Y".

The recipe for recording guitar (or anything) is simply: get the guitar/amp sounding as close as possible to the way you want it to sound, then select the most appropriate available mic and placement for the sound you're after, then use eq to compensate for the stuff you were unable to correct with the amp sound or mic placement, then use additional effects or processing as necessary at mixdown to achieve the most flattering and enjoyable balance of sounds.

Not sure if that answers your question, but that's what I got.

____

Quote:
Originally Posted by flmason
...Anyway, it is a thread, not someone's personal backyard or property, so with all due respect to all the effort Yep has expended, really, what's wrong with people getting into specifics?...""
=====

There is nothing wrong with specifics, and I certainly don't own this thread, but I do think there is a certain usefulness to topicality. This thread is really, really long, and unless you're willing to go through some hundreds of pages of text, it's not a good place to look for advice on guitar gear.

I wager it would be more useful for people reading this thread not to have it devolve into a debate on guitar gear, and that it would be more useful for people who care about guitar gear to read through an easily searchable and relevant thread on the topic than to dig through this monster.

I mean, why not have ongoing pages devoted to conflicting opinions on the best bow rosins for string instruments, or the best ratios or metallurgy for horn coatings,or the best action/responsiveness/material for kick drum beaters and pedals? Because those topics are certainly debated as endlessly as guitar tone. But they have little to do with why your recordings sound like ass.

I suspect it would be more useful for everyone to start a "why does your guitar tone sound like ass" thread, and there to explore all the ins and outs. My guess and my hope is that this thread's usefulness is for those who LIKE their guitar sound, for example, but who are still unsatisfied with their recordings.

In any case, you are certainly free to post whatever you like. I'm not trying to be the thread-Mom, just trying to keep things on-topic.

Quote:
Originally Posted by flmason

....The debate was really about the "all in the fingers vrs. equipment" issues. ...""
=====

There has never been any such debate. If you run a search, the only references to "all in the fingers" have been your own attempts to dismiss a considerably larger school of thought.

There is MASSIVE debate and consideration given to mouth-pieces in the brass instrument world: cup size, bore size, material, coating, treatments, etc and so on. And, like guitars and guitar amps and guitar effects pedals, it remains an open-ended and ongoing discussion, and probably will forever. Likewise, there is huge and significant debate over the effects of metallurgy on tone: whether lacquered brass sounds better or worse or even any different from silver-plated or gold-plated or whatever.

Nevertheless I suspect that there is still a usefulness for a discussion of good recording techniques that sets all that aside.

Similarly we could get into drum sizes, rim contours, laminate vs solid wood, stick sizes, beater heads, head materials, cymbal thickness, head coatings, etc and so on. And god help you if you get into organs and pianos.

Yet it seems that only guitar players think that their own gear selections are critical to the recording process. And it's a bit tedious, and is reflective of the general wankery and self-importance of guitar players.

If you want a good electric guitar sound with no effort and no debate, buy an all-tube Marshall half-stack with a spring reverb tank and either a Les Paul Standard or a Stratocaster American Standard, and then pay for a proper setup, and keep fresh D'Addario, Ernie Ball, or GHS strings on it of the heaviest gauge that makes sense for your style of playing. Record it with a Shure SM57 plugged into a good preamp for loud/saturated signals such as a Neve or API. If the guitar sound is bad, it's not the fault of the gear.

There are a million other alternatives that might be cheaper and just as good, or that might be marginally better for certain genres (i.e. Les Paul or Explorer with GHS boomers, EMG pickups, and a Mesa Triple-Rec for Nu Metal, or a Tele with groundwound strings through a Twin Reverb for twangy, growly clean tones, or a

Gretsch archtop with flatwounds through either a Roland Jazz Chorus or an old Gibson amp for mellow jazzy stuff, etc). You can also achieve quite a lot with a cheap guitar and plugin or digital effects, or any number of different amps. Plus there are lots of manufacturers making alternatives that are arguably even better (Carvin, PRS, Bogner, etc), as well as knockoffs that are sometimes superior to the originals, some cheaper, some costlier. But frankly, compared to legitimate orchestral instruments, an electric guitar rig is not very expensive, and you can pretty easily buy the "real deal" that superstars use for a couple grand (try doing that with a violin, or a piano).

The point is, some guitar player somewhere has got awesome sound from completely unexpected places, and plenty of other guitar players have got crappy sound out of exactly the same rig as (fill in your favorite guitar player).

Moreover, nobody except guitar players gives a shit about that stuff, any more than guitar players sit around discussing whether Mile Davis' horn sound on Bitches Brew sounds like he was using a 7B or a 5C mouthpiece, or whether Miles were playing a silver-plated or laquer-coated trumpet, whether he had brass or titanium or Monel valves, etc. Nobody in this kind of thread wastes much time asking what the coating was or the beater density was on Bonham's drums, or what brand of piano string Rachmaninov was playing on, or what particular Hammond model organ or Saxophone reed or whatever was used on X recording.

Those discussions have a place, and those things do make a difference. But it's tedious and distracting to mix them up with how to make good recordings.

____

Quote:
Originally Posted by BenK-msx
reminds me:
my brass playing sounds like ass because i barely practice
was pretty ok at it once, sob... (bach 5 for me)

yep just reminded me of that - sorry.""
=====

Surprisingly, in the brass world, nobody is much impressed by how fast the trumpet player can work the valves or what metal the trumpet is made of. In fact simple fast trills are about the easiest thing to play on a horn. But we immediately blame the player if the trumpet sounds like shit. Probably because anyone who ever casually blew into a friend's trumpet and produced either silence or a sound like a dying goose realized that it takes skill to make good and musical sounds.

Players of electric guitar, OTOH, tend to be used to a world where, if you plug in the right effect pedal, anyone can pick a string and have it sound like an awesome note. Which makes them tend to blame the pedals if the notes don't sound awesome.

____

Quote:
Originally Posted by rhkk
Can I ask a mixing/recording question???

After "fixing" my room for modes etc... I found a good deal on a nice pair of monitors. They are made by Audix and retailed for $650.00 about 10 years ago. I really like them and can really hear even more of my mixes good and bad. (unfortunately I hear a lot more hiss then I did before). Since I have switched it appears I have a problem. When I mix at a lower volume I can get everything to sound good together. I am happy with my mix until I turn up the volume and it seems one or two things start to really stand out as being too loud in the mix. Usually vocals or drums. I may just not be used to these monitors or maybe my ears suck, but either way it is a little frustrating.

Questions: Could this be that I am not hearing correctly at lower volumes? Could it be the monitors? (I think unlikely.) Could it still be the room? Is this part of the mixing process and I just have to get used to it?

I have to admit I have not had a lot of time to try to figure it out myself. I've only had the monitors for a few weeks and probably have only used/heard them for about 10 hours.

I could post a mix if that would help but I thought I would ask to see if there might be a simple answer first.

THANKS""
=====

I want to reiterate that there is no "magic recipe". The important thing about building a bridge is that it stays up, and the important thing about making a record is that it sounds good. In either endeavor, there are a whole ton of inter-dependent principles and practices that can be employed, but the best engineers, the ones who create the best, most efficient, least clunky, most elegant results, are the ones who adapt the tools and techniques to the situation.

Fortunately making records is vastly easier than making bridges, and someone working at home on a hobbyist budget, using commonsense practices, can pretty easily achieve adequate results, and can often achieve great results.

The whole "monitor mostly at low volume" thing is a technique that can often make things easier, but it's not a rule. It's worth noting that the very reason why it often works is the same reason why it's not perfect: volume changes sound. It stands to reason that it may change instrument balance.

Soundtracks for cinema offer the mix engineer a great advantage: most cinemas are calibrated, not only to play back at the exact same volume that it was mixed at, but often on a soundsystem calibrated to certain standards of quality and frequency response. In effect, every audience is hearing exactly the same sound that the mix engineer heard: this makes mixing a much easier project.

When those movies go to DVD or Blu-Ray, they are typically remixed and re-mastered to reflect a somewhat less forgiving range of playback equipment: the dialog is usually turned up for clarity, the subsonic effects are turned down, the dynamic range is compressed for living-room systems, the frequency and ambient information is re-worked with consideration for less "dead" rooms and less "flat" soundsystems than cinemas offer, and so on.

When those movies go to broadcast or cable television, they are often re-mixed yet again. Now they not only have to "work" sonically on living-room DVD players, but on pizza-parlor TVs and in the bedroom at whisper-volume while the wife and kids are asleep, and they have to compete with flat-lined, loudness-maximized commercials, etc.

Each stage above requires progressively greater compromises in terms of realism, drama, impact, dynamic variation, and sound quality. But luckily for the movie mixer, each stage has pretty clear and specific goals and instructions.

The music mixer has a far more subjective role to play. In theory your mix should both be a perfect and idealized "cinematic" record of the band's performance, but it should also "work" on a 2" speaker in a pizza parlor or late at night with the kids in bed.

But there is good news: the broadcast industry is "helping you out", with quite a bit of technology devoted to making all music sound roughly the same. Radio stations, bar jukeboxes, TV soundtracks, nightclub PAs, etc will all run your tracks through some degree of processing, typically multiband compression and exciters. After all, they just as much interest as you do in making sure every track sounds good.

Nevertheless, it is an unfortunate reality that the music mixer, unlike the cinema mixer, cannot rely on calibrated soundsystems to exactly reproduce what she is hearing (it is also an unfortunate reality that few music mixers get to work in a cinema: a giant room with soft, dark walls, 10,000 watt calibrated speaker systems and full-range subwoofers is a luxury afforded to very few professional control rooms, never mind home studios).

I would never ever discourage anyone from making the best-possible recording they could make, but a certain amount of practical necessity often has to intervene in definitions of "best".

I would encourage anyone reading this thread to adhere to her own definitions of "best", since sound quality is ultimately a subjective thing, but in practice a certain amount of versatility is usually called for: a recording that sounds good on only one playback system or in only one volume range is probably not ideal.

In my experience, mostly dealing with rock/pop music and some acoustic and jazz-type stuff, the ratio of listening volume during mixing that I have found to be most useful is roughly:

- 50% extremely quiet, like as quiet as you would ever listen to the material, minus one, especially for technical adjustments and overall instrument clarity

- 30% moderately quiet, like conversation-level or slightly less

- 15% "regular" listening volume: loud enough that you have to turn to face somebody to hear what they are saying

- 5% "cranked" volume: big, nearly concert-level volume to hear how the lows and dynamics hold up.

None of the above are any "better" or "worse". But for me, I usually find that working at lower volumes leads to mixes that still sound pretty good when played loud, with minor adjustments, whereas mixes that sound good loud do not necessarily sound good quiet. IOW, I tend to get better results across the spectrum by doing most (but not all) mixing at low levels.

When it comes to tracking, I usually monitor a step or two louder than while mixing, typically at something closer to regular listening volume.

I'm not sure how helpful that is, but that's what I got.

Quote:
Originally Posted by Gizzmo0815
Yep,

If you'd be so kind as to help me with a philosophical question on mastering.

I've been reading a lot recently about the theories and concepts behind mastering. I find that often it's difficult to find a true definition of the term...""
=====

Now THIS is an easy question: "mastering" is the job of taking the finished audio recordings and preparing them for reproduction.

You take the final tape or 24-bit stereo wav file or whatever it is that represents the "finished" studio recording, and from that you produce your glass CD master, your pressing disc for the LP plant, or whatever will be used from now until forever to produce the copies that will be sold at retail.

This highly-technical step is important for wide commercial releases: the creation of the final master should ensure that every CD player (or record player, or mp3 player, or whatever) is going to reproduce this correctly, is going to understand the transitions, pauses between tracks, track numbering... that the needle isn't going to skip tracks because the grooves were cut outside tolerances, that older CD players aren't going to unleash a full-scale buzzy clipping at track transitions, that there aren't any digital errors that are going to make the CD unreadable by some brand of CD player, and so on.

Creating a fault-free duplication master is not a terribly difficult job, but it's one of those things that is worth getting right before you stamp and ship a million copies.

That's what mastering is. But I'll bet that didn't really answer your question. More in a minute.

Mastering part 2:

Step into your time machine and set the dial to sometime before, say, the early 80s. Sometime before digital reproduction was common. Now step out of the time machine and go visit a record-mastering lab, the kind of place where they engrave the discs that will be used to stamp actual vinyl records.

The people who work at this facility have a considerably more demanding and technical job than the modern "mastering engineer" has. For starters, the "record" they are receiving if usually a pile of reel-to-reel tapes. The tapes may have been recorded on different machines, at different speeds, calibrated differently or incorrectly-- who knows what (hopefully there is good documentation). The final "track listing" is a piece of paper that tells them what order to put the songs in.

Analog transfer between different mediums is not an exact science, especially when the starting mediums are not identical to begin with...

For example, a track cut at 30ips will sound more detailed but hissier than a track recorded at 15ips, for example. And your pile of tapes might include stuff that was tracked at 15ips but mixed down to 30ips, or who-knows-what. Maybe with varying types and degrees of noise-reduction, and so on. So for starters, there is a high probability that the various album tracks will all have noticeably different amounts of hiss and SNR. Unless you want the album to "pop" and/or noticably change hiss levels at every track transition, you are almost certainly going to have to do some work on this.

For seconds, if any of the tracks or "finished" mixes were done at different studios, or at different times, or by different engineers, or on different tape machines, then there is a very good chance that these tracks have never actually been listened to in sequence before: changing a reel of tape takes time, and the probability is very high that many tracks will have serious instrument or frequency imbalances. E.g., that the snare drum or bass or vocals will be much louder on one track than another, etc, or that some tracks will sound much more midrangey or bass-heavy or whatever. Not "bad", mind you, just different aesthetics, different approaches that will nevertheless end up sounding jarring or distracting on a finished album. One of the most obvious examples is the mellow, steady-state ballad that takes up the whole dynamic spectrum and that ends up sounding twice as loud as the hard-hitting rock or funk track that needs headroom for big dynamic swings. This is a pretty obvious opportunity to improve the overall album by making some adjustments.

For thirds, there are some pretty serious technical considerations when "mastering" to vinyl records (less so for CD). Magnetic tape heads are different from metal needles weaving their way through record grooves, and stuff that plays just fine on tape might not remain centered in the groove, or might cause the needle to skip, or might cause one groove to cut into another, and so on. There are a variety of "rules" for cutting vinyl, and making a tape mix fit into those rules may require a certain amount of eq, stereo field manipulation, and compression on purely technical grounds, completely setting aside aesthetic considerations.

(pause): you may be starting to get the notion that mastering engineers, in the old days, basically HAD to do a certain amount of sonic re-processing to the "finished" mix, just as a routine part of getting the tracks fit for reproduction. And you may also be getting the impression that some of this processing blurred the line from strictly "technical" functions into more aesthetic/subjective decision-making. This would be a correct impression. (un-pause)

For fourths, the mastering engineer has a very privileged high-level view of the overall recording, in a number of ways: 1. She has never heard these tracks before, and has no emotional or personal investment other than producing the highest-possible sound quality; 2. She might work on a hundred professional recordings per year, without the tunnel-vision of a producer or band who might have only done 3 records in their whole career so far; 3. She is typically working in a very technically "pure" environment, a well-made listening room with just a few high-quality processors, removed from listening couches and arguing band members and a million knobs and faders to keep track of, removed from racks of noisy gear and comb-filtering from a giant mixing console under her face and the wear-and-tear, stress, and ear-fatigue of a working studio...

In short, she has the "critic's privilege": she gets to listen to and analyze the results, completely removed from the process, emotion, turmoil, doubt, and complexity of having made the thing. Couple that with an expert ear, top-flight equipment, and hundreds of album's worth of experience, having heard the results of her own work on the radio, in nightclubs, on home hi-fis, in car stereos, on headphones, etc... she is in a *very* good position to spot any number of obvious flaws or opportunities for sonic improvement.

More to come...

Mastering part 3

So still in our time-machine, and considering all of the above, it is not surprising that a number of mastering engineers would emerge with reputations for not only producing technically correct and error-free reproduction master copies, but who also added an extra layer of polish, professionalism, and "magic" to the records that came across their desks.

Even just the rudimentary, technical basics of noise-reduction, setting the song levels appropriately, setting appropriate pauses between tracks, and doing the fade-in/fade-out and tuck-and-tails correctly could make the "master" sound noticably more polished and professional than a simple spliced-together tape of the various mixes.

Add to that the mastering engineer's ability to "fix" problems with the source mixes, such as a too-mushy kick drum on this track, or a too-loud vocal on that one, etc, and the "master" could often come out sounding dramatically better.

But none of that is actual "mastering". It's "pre-mastering" or stuff that the engineer does before the technical work of creating the error-free master copy. In fact, it is typical that the mastering engineer would send this tape to the producer for review before cutting the master pressing disks. In other words, all of the "magic" of the mastering process happens before the "mastering" even begins.

This is an important distinction: the mastering engineers of yore did pull out their equalizers and compressors and stereo controls to "improve the mix". They pulled out those tools to get the needle to stay in the groove. While they were at it, they also used those tools to get the noise levels and track levels balanced, the crossfades sounding natural and balanced, and the overall frequency balance consistent from one track to the next. While they were at it, they took advantage of obvious opportunities to improve the mixes, especially "problems" revealed by the new stereo and frequency balance.

the net result was often records that sounded better than the original studio recordings.

More in a minute...

Mastering part 4

If you have read through the above, you might already have a sense of where this is going. Certain mastering labs and individual engineers began to emerge who had a reputation for adding a full letter grade or two to the sound quality of records that they worked on. Send your tapes to one of these guys, and the masters would come out not only technically correct, but sounding better than they went in.

Now get back in your time-machine and flash-forward to 2010: "mastering" CDs, mp3s, etc is a fairly trivial, automated, and software-run task. On a million-release CD, it's still worthwhile to have it done by an expert, but the cost is negligible.

What has remained is the notion of "golden ears" mastering engineers (really pre-mastering engineers) who put an extra layer of sonic spit-and-polish on the mix before the CD factory runs it through error-checking and the stamping machine. In fact, in a purely digital universe, a pure "mastering" lab doesn't even need to own a pair of speakers: they are simply producing an error-free glass master that meets the technical specs required to cut commercial CDs.

Increasingly, modern "mastering engineers" are often not even providing a master pressing copy: they're just running the mix through some eq, dynamics, exciters, delays, whatever, and then giving you back a modified digital file that the pressing factory is expected to reproduce exactly. It's more like "post-mixing". There is no longer any technical element to it, you're just hiring somebody to second-guess your mix as best they can, without giving them the source tracks.

Now, this or that "mastering engineer" might or might not have better or more expensive gear than you do, or a better room, or whatever, but in a purely theoretical sense, there is no technical reason why you couldn't do everything they do if you bought all the same stuff. You're paying them for their judgement, tools, experience, and skill at making it "sound better".

In short, modern "mastering" (as it is commonly thought of) is really just re-eqing, re-compressing, re-reverbing, etc of your existing "finished mix", usually with some help on the fades, track levels, and tuck-and-tail. But increasingly there are "mastering engineers" who work one song at at a time, so even those last criteria of album flow and sonic cohesion are absent.

This does not mean that modern "mastering" is worthless: on the contrary, it is often by far the cheapest and easiest way to get a second opinion and extra help from a set of expert ears with expert gear. But it has also become one of the easiest ways for someone with a computer to make money from people who don't know what they are buying. You could run a typical home recording through maxxbass, L2, and BBE presets and send it back to the client without even listening to it, and half of them would rave about your magic touch and recommend you to all their friends. (And maybe they'd be right... sometimes it seems like that is what recording is coming to: who knows...?)

As the process of reproduction becomes more and more automated, the role of mastering engineer is an increasingly consultative one. The professional mastering engineer is an expert set of ears who gets paid first to tell you the obvious things that are wrong with your tracks, and secondly to put a final coat of spit and polish on them. And that's still a valuable role: even very skilled mechanics and carpenters still hire specialty subcontractors to do the finish work.

Quote:
Originally Posted by MCV
Hi Yep, et al,

WOW. Finally made it to the last page. I've been reading this thread for more than a week now. I haven't really read everything from everyone, just all of YEP's posts and some of the rest, the longest ones mostly.

It's been a pleasure all along. This thread is without a doubt one of the most enjoyable readings I've found on audio on the web, along with maybe the mixerman diaries (I thought it was so good I bought the book in the end, highly recommendable IMHO). I endorse the opinion YEP should be writing a book. I know I'll buy it.

I'd like to contribute a couple of thoughts of my own on the subject of the thread.

1. Many a decision I do when tracking, editing and mixing is one of very short-term gratification versus long-term value. I'll try to illustrate.

A vocal will immediately sound better auto-tuned, will sit better in the mix, will most probably (crying shame?) be preferred by the singer; but it will make the vocal less remarkable, less interesting.

Drums doctored with beat detective might sound better than a real performance or not, but will for sure sound safer, as a performance must be judged subjectively, and unless you happen to be recording a stellar drummer, you may not be 100% sure if a little "deviation from the grid" is actually a cool thing or not - or more to the point, the customers (again, crying shame?) may be unsure.

A louder master will no doubt sound "better" than a softer one in a/b comparisons if not matching levels, but after a while it will sound tiresome, and the lack of dynamic range will make it sound dull in comparison with older, more punchy and exciting recordings.

More or less the same for a more compressed sound, over-edited guitars and bass, more lows, more highs, scooped mids, all the staples of contemporary productions. They kinda sound good to people on the spot, but six months down the road and the record will have lost all interest.

Wich all goes to say: maybe just 'trust your ears' and 'make it sound good' is not enough. In my case, what I try to do is to explain all this to the people I'm recording, as the sad truth is now that the singer wants auto-tune, the drummers are asking for beat detective, the guitar players rely on me editing as they can't keep tempo for their lives (don't even get me started on keyboardists), and the whole band cries for the masters to be as hot as possible (they all seem very filosophical about it even, they say they read of the loudness war in this or that forum and think it is a bad thing, but the music business being so competitive they just don't want to sound softer than the band next door, etc).

But it is hard to blame anyone for wanting to sound contemporary and "pro" whatever that is. I sympathize really. OTOH, I should try to be more coherent with what I really think sounds good. I think most of us should.

2. I think a lot of the problems that home-studio recordists face have to do with the fact that most people have never recorded anyone apart from themselves and maybe their band. So you ask them to judge whether things sound good or not, but they really can't tell because most people is used to listening mostly to themselves, and most people sound like shit.

I still remember the first time I recorded a good vocalist. It seemed to me like a miracle: all you had to do was place a microphone (any microphone) in front of her, and it sounded good. Before that I had only recorded myself and some other lousy singers, so I didn't know any better - I really thought it was about compressors, eqs, reverbs, preamps, you-name-it. Same with any instrument. Some people just sound good, but most people don't, and it obviously follows that most recordings have to suck ass no matter what (in their raw, unedited tracks).

Enter digital non-destructive editing and auto-tune or melodyne: the tools that let Joe Blow get his artistic vision across even thought he migth sound like shit by himself. I think this is wonderful, as now the "artist" is no longer constrained by its ability as a performer, hence the possibility to give birth to more and more meaningful artistic visions, sounds, songs, etc. If only Joe Blow would HAVE an artistic vision, and if he could concentrate on developing it instead of trying to sound like the last album by whomever, or wasting time and money in preamps and compressors and forums...

What I'm trying to say is that I'm convinced there's a lot of people out there that already have tne knowledge, the means, and everything to put out great music, but somehow we're missing some kind of deeper artistic direction. I'm talking about recordists like me, I've been doing this for years now and I'm getting the sound I want, everything sounds rather good, better everyday in fact, and I'm not complaining about myself - just in general, I've this feeling that so many competent but thoroughly forgettable albums are being made every month.

3. You've talked about studio setup, compression, vocals, drums, guitar, bass, etc. You haven't talked much about keyboards. Synths specially, if you try to sit them into a mix with mostly acoustical instruments, can be a PITA for most people (me included, if it doesn't work I tipically reamp through a guitar amp, real or virtual, as eq, compression, distortion and reverb just don't seem to work the same way, just like you said some 30 pages ago, something in the lines of "a dehidrated meal plus water is not the same as the original meal", when talking about reverb and real spaces).

Again, just wanted to say thank you for this long thread. It is deeply satisfying and reassuring to read someone I can agree with most of the time.

Cheers from Spain. Sorry 4 long post.""
=====

This is a fantastic post all around. I agree with everything you've said. And more on synths is in order.

Synths and keyboards (as distinct from piano) are harder to talk about because they are so completely wide-open.

For starters, when using a keyboard to try and replicate other instruments via samples, it's hardly worth talking about: just do the best you can. There are just way too many variables, starting with the quality of the samples, but mostly centered on the problems of trying to "play" saxophone or whatever on a keyboard. Saxophones don't play the same way keyboards do, and a huge part of their sound comes from performance gestures that have nothing to do with aftertouch or volume. So just do the best you can, and don't dismiss the idea of using deliberately "fake" sounds that achieve the effect you want instead of fruitlessly pursuing "realism": a good, well-selected electronic sound often sounds a lot less cheesy than a just-short attempt to create a "realistic" sax or guitar sound or whatever.

So setting aside piano, organ, and "fake instrument" keys, we are broadly left with sort of two different roles that keyboards tend to play: I'm going to call them "band keys" and "production keys". "Band keys" means keyboard parts that are essential parts of the song/performance, just like the lead guitar or whatever. Stuff like the Killers or industrial music or Enya, plus most modern dance music and a lot of top-40-type stuff. Wherever the keys are carrying part of all of the song. "Production keys" are the kind of stuff often played by producers or hired musicians to "sweeten" or fill out rock bands, acoustic songs, cinematic soundtracks, etc. The song could be performed live without these keyboard parts and hardly anyone would notice anything "missing".

Of the two, "production keys" are generally much easier simply because they tend to be layered on top of the "finished" song, and are only used when they make an obvious improvement. They might be anything from airy pads to chimey arpeggios to blippy percussive intervals to add motion to chugging guitar chords, but if they don't make an obvious improvement, or obviously fit, then they're generally not used to begin with.

When it comes to "band keys", MCV brings up a really good point above. I think there is a lot to be said for recording synths the way you would record an electric guitar. Something often tends to get lost when you track the direct output of the synth part that is used to being performed live, just as DI guitar almost never quite feels right. Moreover, guitar amps can work wonders for synthesizers. So if you're working with a band where the synthesizer is a core part of the band's live sound, try recording it through a speaker, and especially try recording it through a guitar or bass amp, since keyboard amps tend to basically be little more than mediocre PAs.

Secondly, contemporary synths tend to increasingly favor "vintage", "analog", or even "dirty" sounds to a great degree. It seems that synthesizer sounds are increasingly cousins to overdriven electric guitar, with squealy, overloaded, "fat", chirpy sounds, as distinct from the spacey, futuristic efforts of yore. And the world of electric guitar processing and amplification has decades of bizzare and wonderful alchemy dedicated to the art of signal transmission that has zero to do with fidelity and everything to do with awesomeness. There is a whole world of analogy, unnatural, electronic-sounding, soviet-era technology dedicated to guitar signal that is, if nothing else, vastly less expensive than comparable synth modules.

The guitar player's bag of tricks often includes a boatload of not just distortion effects, but reverbs, delays, ring modulators, flange/phase/chorus effects, analog eqs and tone controls, wahs, and so on that might not pass muster on even a cheap department-store stereo for [/]fidelity[/i], but that score big points for awesomeness. And these effects are often available for $20~50 apiece, unlike synth processors, which tend to start in the hundreds. And guitar plugin bundles can be got for little or nothing. Something like FreeAmp should be installed on every keyboard player's computer.

The synth artist has both the blessing and the curse of nearly infinite control over the sound from the start. As we have seen, what sounds good by itself often does not fit very well once bass, drums, guitars, and vocals are added into the mix. And the denser, fuller, and more complex the sounds are, the worse the problem gets: what ends up poking through the mix is whatever frequencies nobody else wanted.

The best way to do sound-sculpting and performance-tweaking for ANY instrument is through regular, disciplined live rehearsals in a decent room with a good sound-system. A band that is playing together regularly, and actually hearing and trusting what they sound like is a band that will consciously and unconsciously make those thousand little tweaks and adjustments that move the performance from "competent" to "kick-ass". The band that does most of their practicing solo, and who sporadically rehearse in a noisy room with an underpowered PA where nobody can't hear each other often sounds like a trainwreck when they all get together.

The problems of trying to fit fat synthesizers into a mix are much the same as those with electric guitar: when you have a full-spectrum, saturated sound, it tends to either drown everything else out, or else recede into a powerless background hiss and fizz. It's very hard to find a middle ground.

One of the easiest and most effective ways to deal with this problem is with basic arrangement. If you listen carefully to a lot of top-40 music, or popular dance music, or commercially-successful rock bands, you will find that very often the fat synths and guitars are playing pretty sparse parts, especially when the singer is singing. Think of C+C Music Factory's "Everybody Dance Now" or AC/DC's "Back in Black": there is quite a bit of silence in those riffs. So instead of a giant wall of white-noisy hash, we experience occasional stabs of saturated power. Alternately, you could consider something like the maddeningly catchy "Can't get you out of my head" by Kylie Minogue or Beyonce's "Crazy in Love": both feature pretty subdued, sparse, and punchy/open instrumentation behind the vocals. The latter obviously has the big Chi-Lites horn sample, which provides a big, dramatic contrast to the mostly drum-based verse sections.

An important thing to note is that in all of the above songs, the primary instrumental parts would be extraordinarily unsatisfying things to play as a solo instrumental piece. No sane person would ever sit down at a keyboard or guitar and simply play one of the parts from those songs all the way through as a solo piece: you'd go mad before you made it to the third verse, and it would be unlistenable. The effect depends on the whole ensemble. Incidentally, this is also true of quite a bit of orchestral and world music: if you were to isolate the third Cello part from a symphony piece, or a single Djembe from an Afropop ensemble, it would often sound like nothing at all, some leftover bit of distracted noodling or whatever.

I think this "curse" is a major reason why synth artists are uncommon in indie and "band"-type music: it is so very easy for the synth artist to create and play complete works, and it is very difficult to then fit those into a band scenario. Something to think about. More later.

Quote:
Originally Posted by Brandon7s
...In order to output a signal from the bass to both at once, do you use just a simple 1/4" Y-splitter?""
=====

People will argue a lot about the "best" way, but a simple "Y" cable has certainly been used by plenty of great players on plenty of great records.

Guitar transduction is a very crude, primitive, and imprecise technology. It's almost comical to hear people arguing about precision this and that when it comes to guitar signal. If you want an "accurate" capture of how your bass actually SOUNDS, stick a reference mic in front of your bass and record it through a good preamp (it will sound awful).

There is no such thing as a "high quality" signal path for electric guitar, not when the pickup starts out with crude magnetic transduction. All that matters is whether it sounds they way you want it to sound. If a Y cable gets you there, it works. OTOH, there are also plenty of other ways to do it, and you might like them better.

Quote:
Originally Posted by five_nines
Hi Yep,

long time listener, first time caller.

Thank you for taking the time to write so eloquently on these topics. Your insight is amazing and you have a gift for teaching.

I have a couple questions that I am hoping you can shed some light on….

1- Headphone mixing. I understand that it introduces an entirely different set of problems. However, I would like to understand that in a more practical sense. For a variety of reasons, I have a certain amount of time to mix in a room, and the rest of the time, I am constrained to headphones. What would be the best way to split up my time? Are there certain steps in the mixing process that are preferable/acceptable to mix on cans, while saving others specifically for the monitors? And in those cases where headphone mixing ends up being done, are there certain pitfalls to watch out for? There has been some talk on this, but any elaboration would be appreciated.

2- Automation. I have recently been starting to realize just how extremely vital automation is to having a great mix. Why is it so rarely discussed compared to compressor settings or which Neve clone sounds best? (caveat: I do read those with at least some fervor) Your breakdown of mixing on the thread (BDV coming first, temporary reverb, etc) was illuminating. Any chance you can layout a similar description of your automation process? Do you automate individual drums (or any other instrument), as well as the various sub-mixes, stems, groups, entire mix? Do you automate at whisper quiet levels as well?

The next one is sort of OT, but just something I was wondering (so even a link with an explanation would be great)...

3- The "Q" in EQ. Is there a standard for the numbers used that translates musically? The Waves plugins go from 7-100, while the URS plugs go from .25-3.00. I know to use my ears, but I would still like to fundamentally understand what's happening

Thanks again Yep, and Smurf, and everyone else who has contributed to this killer thread, and please keep up the good work.""
=====

Thanks for the kind words.

Topic 1 has been discussed endlessly. Just get the best results you can with what you have to work with. Think of mixing on headphones like mixing on bad speakers: try it if you feel like it, but verify on real monitors. If real monitors are unavailable, then just listen on whatever you got as often as you can. It's like asking for tips on how to play an out-of-tune guitar-- there is just no way to say what will work.

Automation... in reaper and most other DAWs you can pretty easily set "record automation" options. I would just suggest setting that to record while you are mixing, and then mix in real-time, as though you were playing an instrument. Then go back and tweak to taste. Not sure if that answers your question.

Q settings: on a typical eq, a Q of 1.2 roughly corresponds to one octave bandwidth of audible change, IIRC. This is a pretty good starting point for most EQ work. If the EQ uses an octave system (such as URS), you can use that instead. Otherwise, the narrower you go, the more ringy and "fake" the EQ starts to sound. The broader you go, the smoother and more natural it starts to sound, but the less control you have.

I think a good rule of thumb when starting with eq is to look for the mildest settings you can find that achieve the result you're after, and then cut them in half. I.e., if a 6dB cut with a Q of 1.4 seems to zap the offending frequency, then try pulling the cut up to -3dB instead of -6dB, and try setting the Q to .7 instead of 1.4. Then stop playback, rest your ears a bit, and then compare the un-eq'ed part with the "full" EQ and the "halfway" EQ-- I bet a lot of the time you find the "halfway" EQ sounds best (you might even try halving it again). In time, you'll get a better for when you're over-compensating.

In keeping with the above, the deeper the gain setting, the broader (lower) the Q should be, as a general rule for "natural" sound. A steep, narrow, high-gain EQ gives that sort of filtered dance/techno/telephone sound: neither good nor bad in and of itself, but very unnatural, and very easily overused. Also easy to get sucked into when your ears get burnt out: you get to that stage where everything sounds old and drab, just because you've been pumping the same material into your ears for 8 hours, and you start doing stuff just to change it for the sake of hearing something different. Beware. The next day you'll be like: WTF did I turn up the lower mids on the hi-hat 12dB for?

The other good rule of thumb for EQ when mixing is: try EQing a DIFFERENT INSTRUMENT from the one you want to modify. If you can't hear the vocals clearly, instead of cranking up the upper mids, try turning down the upper mids of other instruments (esp guitars and hashy drums/cymbals).

If you're fighting the EQ, the problem is often that you're trying to fix the wrong thing with the wrong tool. To a man with a hammer, every problem looks like a nail.

Quote:
Originally Posted by Arbiter
...I'm curious what everyone has to say about master bus effects (if it's not too early to get to this). Do you use anything at all or just edit on track/bus level? ... Or do you recommend rendering your mix to a stereo file beforehand and treating this as a separate stage? .. Or is it simply best to leave these off and allow a (pre)mastering engineer take care of anything above track/bus level?

-Michael""
=====

If you are giving the mix to a mastering engineer, always supply them with both an uneffected mix *and* whatever you were hearing while mixing.

As for which approach to take, some schools of thought:

- Put some compression or other effects on the mater bus while mixing just to "hear" what it will sound like after mastering, broadcast compression, playback through under-powered systems, etc (not necessarily because you intend to keep the compression). I'm personally pretty skeptical of this approach, and I think it's often a form overthinking/outsmarting yourself, but a lot of credible people like to work this way.

- Clean and pristine: in theory, anything you can do at the master out you could probably do better, with more control and more precision using busses and per-track effects. Makes sense in theory, but IMO there is nothing wrong with doing what feels right, even if logic says that it *should be* wrong.

- Whatever works: if master bus compression/eq/reverb sounds better than what you can get from individual tracks and busses, then go for it. Just be careful with this approach, and use the same level-matched listening approaches discussed above. Slapping more effects makes the mix louder, and if all you're doing is adding level instead of sound quality, then it becomes self-defeating, and often actually leads to significantly WORSE sound quality as you keep pushing more "louder" buttons. And definitely give your mastering engineer both versions: she may achieve whatever you were going for better than you did, or she may advise you to make some changes to the mix.

My own suggestion would be to approach anything you do to the master out cautiously and conservatively. In theory, if the idea is to present the most flattering and balanced soundscape of the various sonic elements in the song, you should be able to better achieve your goals with per-instrument processing or adjustable effects bus sends.

E.g., if a little reverb over the whole mix sounds good, then chances are, that same reverb would sound even better in a reverb bus with the various instruments set to appropriate levels that reflect the amount of distance and space you want each of them to have: it is unlikely that kick drum or bass guitar benefits as much from reverb as the cymbals or background vocals do, for instance, even if the bass does benefit from a little bit of it. Similarly, a compressor over the main out might help to "lock" the timing and rhythmic feel of the bass, drums, and guitars, but it will also probably start pumping and sucking the vocals, hi-hats, piano decay, acoustic guitars, etc. And unless you *want* that throbbing, sucking, artificial club-mix sound, it's probably more effective to "lock" the rhythm section with a separate compression bus than to pump the whole mix with a compressor that sucks everything down on the bass hits.

There are certain kinds of "special effects" that can be done on the master bus, e.g., "telephoning" the whole mix by cutting the lows and highs and compressing the whole thing for a breakdown, so it sounds "small" and "lo-fi", and then opening it back up so it sounds huge again, or pulling the master volume down to silence and then coming back in with a backwards reverb for that "rushing forward" accent and so on. And sometimes a little bit of distortion or lo-fi tape or record noise can be applied to achieve a "fake" or "recorded" or "old" sound, but this should be done judiciously: if the whole song has it, then it will just sound like a bad recording. And a lot of the above sounds very dated and corny if not done very carefully-- the late 90s and early 2000s saw a lot of that kind of stuff, and it hasn't aged all that well.

It's kind of funny how in the days when it was a struggle just to get recordings that sounded clear and lifelike, people put massive effort into the basics of clarity and accuracy. And now that it is very easy and cheap to make fool-the-ear accurate recordings, we spend a lot of thought, effort, and money trying to make recordings that sound distorted and "recorded" in those "just-so" ways to recreate those "vintage" records that were basically trying to reproduce the way the band sounded in the room.

I would encourage beginners to start the way the old-timers did, and to try to create a recording that sounds like the PA feed from the ideal performance in the ideal concert hall. Focus your recording efforts on technical perfection, clarity, and accuracy. And only then use the tools to polish and "improve" reality where possible. Partly because I think, in the long run, that approach produces the best and most lasting recordings, but even more because it focuses attention where it matters: on the songwriting, arrangement, and performance.

Wax and polish can make a cherry car shine and gleam and sparkle, but it can't disguise a dented rust-bucket. I think too often we waste time looking for better brands of wax and dent-fillers when we'd get better results quicker just by sanding down to bare metal and re-finishing the car from the ground up.

Quote:
Originally Posted by Gizzmo0815
THANKS!! The thread is much cleaner now.

Yep,

When, why and how would one use expansion? I know the basics of how an expander works, but I can't say I've ever come across a time when it was usefull.""
=====

Oh, wow, great question.

The obvious use is to reduce background noise. It's like the opposite of a compressor. If you imagine the little gremlin from way back who turns down the volume on a compressor, in this case it's his evil twin, who also turns down volume, except this time he does it only when the signal drops BELOW threshold. And a noise gate (or just "gate") is just a "hard" expander, like a limiter is to a compressor-- below threshold, the signal is set to "off". The controls are otherwise exactly the same, except they work kind of opposite:

- The "attack" delays the onset of the volume decrease. So once the volume drops below the threshold setting, the gremlin waits a little bit before turning the volume down, producing longer "tails" at the end of notes. This is helpful to avoid a "gated" or "cutoff" sound where the "note" crosses below the "noise" threshold, so that the expander kind of waits for the note to drop below the noise level, then brings down both the noise and note alike.

- The "release" means the gremlin holds the volume down for a little while after it has increased above threshold, "softening" the transients. Unless you know something I don't (which is entirely possible), release is usually set pretty short with an expander/gate, just enough to prevent "false starts" or little swells of noise at finger squeaks, etc.

You can use an expander/gate early in the signal chain to reduce hum and feedback with an electric guitar or similar, or to reduce bleed-through on drum mics, or room noise on a vocal track, etc.

In fact I would recommend trying them as almost a default effect: you can always disable it if it's not improving things, but cleaner "silence" is almost always a big improvement, especially when you start piling on lots of tracks, all including their own noise. Distorted guitars in particular frequently benefit from simply putting a gate early in the signal chain and just leaving it there.

A little expansion can make a big improvement when using cheap preamps, noisy electrical, home studios with background noise, etc. And you don't have to hard-gate everything to get the benefit: focus on setting the expander/gate so that it doesn't interfere with the "good" parts of the sound, instead of trying to kill all the noise or bleed-through. A 3 or 6db noise reduction times 20 tracks is a huge improvement, without making everything sound fake and gated.

You can use side-chains or an expansion bus to achieve some interesting effects.

- Feeding the snare drum into the side-chain of a noise gate that is set to gate a track of white noise allows a little "explosion" of white noise to cut through on every snare hit. This can add a bit of "explosiveness" to a dull snare, and tweaking it so the white noise only comes through on the peaks or sharp accents can help to focus the backbeat on a busy snare pattern.

- An expander/gate can help to clean up and "lock" layered instruments such as multitracked guitars and backing vocals that all have slightly different start and stop times. You can either send them all to a gated/expanded "bus", or take the "best" performance, and feed it to the side-chain of an expander that will trim the tails and transients to fit "behind" the "good" one. (one case where you might want a longer release).

- "gating" reverbs or other ambient/delay effects this way can force them to sit behind the main instrument, to provide the richness and density without "echoey" tails or transients.

- If you're into electronic or "fake" sounds, there are all kinds of things you can do, such as triggering a gate on a synth bass with kick drum hits to make it sound like the kick drum is playing "notes", or triggering a gate with a delay feed to get a sort of "stuttering" effect on vocal or instrument tails.

____

Quote:
Originally Posted by IIRs
This isn't really true. A limiter is just a compressor with a higher ratio...""
=====

I was speaking figuratively, in the sense that "gate" indicates a "hard" expander (e.g. one with infinite ratio), just as "limiter" generally refers to a "hard" compressor, or at least one intended to be used as such.

I.e., my intent was to convey that, just as there is some fungibility and blurriness between the terms "compressor and "limiter" and their respective uses, so there is with "expander" and "gate": one can be useful where the other is indicated, and discussions of one can also be applicable to discussions of the other. So when someone talks about "gating" this or that, an expander can often be used to equal or better effect. In fact quite a lot of people will talk about "gating" this or that when they are really using some non-infinite amount of expansion to achieve the effect. But thanks for the technical clarification.

____

Quote:
Originally Posted by TB!
Quote:
Originally Posted by yep

everybody knows what a great record sounds like...
I am not so sure about this...""
=====

I guess maybe I could re-state that as "everybody can name something that they consider to be a great-sounding record", but that conveys an implication of purely subjective relativism that I don't necessarily agree with.

I think one of the hardest things for beginners to untangle is the difference between a great recording, and great substance. IOW, a lot of my favorite things to listen to are rather poor or mediocre recordings of really good musical performances, and a handful of my favorite records are very technically good *recordings* or productions of material that otherwise might not be that interesting.

By way of analogy I might suggest "Transformers 2", which is an incredibly dumb movie, most of which revolves around an achingly stupid "off to college" teen romance that has nothing whatsoever to do with giant fighting robots, in fact it has no reason to exist at all as a piece of cinema. However, the passages that *do* involve giant fighting robots are visually quite impressive in terms of cinematic spectacle. In a sense, from a technical perspective, the robot scenes at least might be considered a bad story well-told. There is merit there for film students, etc, even if it is a pretty bad movie overall (even if you disagree with my review, I suspect you can see how the principle still holds).

In older, less-technical artistic fields such as painting or poetry, there is sometimes less of a distinction between substance and technique. In modern media, there can often be a tremendous disconnect between the two: imagine The Empire Strikes Back enacted by a local theater troupe wearing pajama costumes and using cap guns and painted broomsticks, or a coffehouse acoustic singer/guitarist playing Daft Punk covers.

Non-musicians and other people who are completely disconnected from the world of modern music-making often make zero distinction between the technical merits of a produced record and the underlying musical substance, sometimes much to the chagrin of musicians. Musicians, in turn, like novelists or theater actors reviewing hollywood films, often ignore the technical production aspects and hear only (or mostly) the underlying musical substance. Which is fine, from an artistic-merit point of view, but creates some problems if you want to sell records to anyone other than fellow musicians.

For starters, a technically "good" record in any genre has very little to do with arbitrary rules regarding the amount of reverb or anything else, any more than a good movie has anything to do with the number of helicopter shots or fighting robots or slow fades.

A good recording (as distinct from a recording of good music) requires AT LEAST that the basics are covered: the instruments (including drums and ESPECIALLY vocals) are all in-tune and in time; the recording is low-noise; any "distortion" is of the "good" (i.e., desired) variety, and not of the bad (i.e. undesired) variety; the instruments are appropriately balanced and the stuff that is meant to be clearly audible is so; the overall dynamic and frequency balance is as the artist intended it to be or better; the reverb, eq, and overall presentation is flattering and appropriate to the material, etc.

In other words, it doesn't look like painted broomsticks, pajamas, and cap guns on blurry home-video, so to speak.

If a modern record production were likened to a hollywood film, the record producer would be the movie's "director", and the audio engineer would be the cinematographer, special-effects supervisor, camera operator, costume-designer and makeup artist all rolled into one. The artist would be akin to the actors and screenwriter.

If the special-effects guy showed up and suggested painted broomsticks as lightsabers, or if the cinematographer or camera operator couldn't focus the cameras, the movie director would fire him. And if the director failed to, the film studio would fire the director.

In home recording, the screenwriter is often trying to act all the parts, direct the film, operate the cameras, and do the makeup, lighting, and special-effects herself. Which is an awful lot to ask of even a very talented screenwriter. And unfortunately for her, a too-rigid devotion to the art of screenwriting can often be a hindrance to her awareness of and respect for the importance of all those other technical jobs.

One of the things I keep coming back to in this thread is the value of focusing on the performance and the sound "in the room": the stuff that happens before you even turn on the computer or plug in a single mic. There is a threefold reason for this. The first part is that a brilliant live performance is vastly easier to record, mix, and produce: effects sound better and are easier to apply, mixes tend to write themselves, etc. The second is that the more the BAND IN THE ROOM is producing exactly the sound they want to hear from the record, the less "production" is required in the first place: just throw up the mics and hit record--you don't NEED as many effects and fancy techniques. The third is that focusing on the sound of the BAND IN THE ROOM forces upstream improvements from the musicians, instead of highly-technical downstream corrections and edits from the technical crew.

Young filmmakers typically have to produce engaging and entertaining movies on tight budgets, without the benefit of helicopter shots, exploding buildings, A-list actors, CGI Robot fights, etc, before a film studio is going to hand them a multi-million-dollar budget. They first have to prove a certain degree of artistic and technical competence in terms of visual storytelling, pacing, composition, aesthetic sensibility, etc.

Low-budget home recordists, on the other hand, start out with a set of tools and technical capabilities that is very similar to what the megastars are using. This can be a curse as much as a blessing. At the very beginning of this thread, I warned against the dangers of staying up all night A/Bing plugins and knob settings and so on, and of the potential to do more harm than good.

One of the benefits of a simplified process is that it forces the recordist to focus on the technical elements of good audio quality right from the beginning: everything in tune, tight bass, focused sounds, appropriate dynamics and frequency balance, etc. This is stuff that, ideally, the band should be achieving before a single mic is set up.

The best records are well-made recordings of great music. The worst records are badly-made recordings of bad music. Sorting out those two axes can be somewhat subjective, but I don't think it's entirely so.

Quote:Originally Posted by infinitenexus
...That being said, I would like your help with something...

Okay, these are some of my least favorite kinds of questions, but the kind of stuff I know everyone wants answered

For starters, forget about their gear or your gear, or what brand this is or what you're "supposed to" have. Unless you're looking to buy new stuff, that's all completely irrelevant.

Begin at the beginning. Setup your guitar with fresh strings and correct intonation and good action. This matters a LOT-- iffy intonation will KILL YOU with extreme hi-gain guitar, since it will shove all the dissonant murk in the harmonics right out front. Same with bad setup generally: clackety strings, fret buzz, etc will all get gained up into an annoying, sloppy sound with all the wrong kinds of distortion. If you don't know how to set up a guitar, learn. Do some googling or buy a book.

Now plug in your guitar to something clean, either your amp on clean or even just into your mixer and monitors or whatever. Just something so you can hear what the guitar is outputting unaffected. Now play some stuff. Is the sound clear, crisp, and articulate, or is all wooly and brittle and gross? (guess which one you want before you start layering a boatload of gain on top of the sound...). Mess around with pickup heights (this matters a lot and is tragically overlooked), knob settings, pickup position, different picks...

In this clean sound, you want a full-bodied but transparent low end, not tubby, wooly, or honky: that stuff is gonna turn nasty when you gain it up. The lows should sound like actual notes, not just a vague "Mwah, mwah, mwah". You want smooth and ringing highs, not brittle or clipped-sounding: that brittleness is going to turn into fizzy, sissy hash if you put gain on it.

The point here is to get your pickup heights and setup and so on situated so that you're outputting a good signal to begin with. Mess with *everything* that is adjustable to see what your guitar is really capable of. Most reasonable guitars will be able to get something decent, although cheaper ones with worse pickups might have a narrower "sweet spot" (i.e. you might have to stick with one pickup and knob setting, and you might have to crank down the pickup height and/or volume knob to the point where you're defeating some of the "high output", whatever that means). You may also uncover some issues with playing technique that need to be corrected with practice (sloppy fretting, string buzzy, excessively clicky picking, uneven palm-muting, etc-- you can't fix that stuff with effects). But that's the first step: you can't use an amp or a microphone or anything else to fix a guitar that's outputting bad sound to begin with. If your guitar has bad intonation or clackety, buzzy action or pickups set too close to the strings or whatever, then you could plug it into cannibal corpse's stage rig and still sound like ass.

Okay, so assuming you've got a decent clean, clear, transparent, noise-free, good-quality signal coming from the guitar, time to plug it into the amp. For starters, take nothing for granted. Do not assume *anything* about settings, or about the kind of sound you're going to get. We're going to see what we can get out of this amp, not try to force it to be something it's not.

Now, before I start, above I said you should be able to get a decent sound out of any reasonable guitar. The same is not necessarily true of amplifiers, I am sorry to report. I think a lot of people are out there in the world with the wrong amp for their sound (or just a bad one). But hope springs eternal, so let's put on a good face and give it a go. Worse comes to worst, we can always resort to plugin emulators or a pod or whatever (there are some good ones out there)...

So sussing out the amp is a bit of a chore, and the more knobs, switches, and effects involved, the more gruelling it is. I don't know what you use or what you need or how many pedals you like to push while playing, but the closer you can get to having ONE SOUND (or maybe one clean and one dirt sound), using the smallest number of options, the better. My favorite approach is to get one medium-gain sound, and then use the guitar's volume knob and/or pickup selector to adjust the amount of saturation (the "volume" knob on the guitar effectively becomes a kind of "distortion controller" when the amp gain is cranked, not so much affecting the perceived output volume as the amount of saturation). But maybe that doesn't work for you. Whatever.

Set all the amp knobs to halfway or the closest thing you can find to "neutral", and turn the amp up to a comfortable playing volume. That means louder than the sound of your pick on strings, but below (or maybe just occasionally touching on) the threshold of pain. Perhaps counter-intuitively, on all-tube amps and especially bass amps, I often like to *start* with the master volume turned all the way up, and everything else turned all the way down. That way I'm getting the full dynamics of the power-amp stage to start with. YMMV.

Now you need to decide on a riff to "tune" the amp sound to. Pick your most signature riff, the part that is most representative of your typical playing style (this might not be the coolest riff you have, in fact it's usually not). For this exercise, that's the ONLY riff you're gonna play. We can repeat again later with other favorites, but don't muddy the waters.

Now, you mission is to find the best possible sound you can get out of this amp *for that riff*. Your method will be to adjust every possible knob, switch, and combination of things, all the while making an effort to keep the overall output volume the same-- e.g., if you start turning up the eq or gain knobs, you should be turning down the master volume. DO NOT get sucked into the trap of just turning every knob *up* one at a time, confusing "louder" with "better".

This is likely to be an incredibly frustrating experience, for a lot of reasons. That's okay. You're going to do this multiple times over multiple days. After an hour your ears will be completely shot and saturated and you'll probably end up with guitar set to the bridge pickup and all the treble way up. When you come back the next day, that setting will sound shrill and hissy and brittle and you'll wonder how you ever thought it was good. So try again.

Try the most counter-intuitive things you can think of: set the guitar to neck pickup, tone knob all the way down, then turn up the treble and turn down the bass on the amp. Turn down every knob on the amp, and try to get the best sound you can with just the "presence" knob, if the amp has one. See how little gain you can possibly get away with and still have a good sound.

Half the point of this is to simply *learn your amp*, what it can and cannot do, what it's good at and what it's bad at. Focus on different parts of the sound as you change different things. How hard can you push the gain before the lows start to sound woolly and muffled? how does each knob affect the amount of "punch" (each of them probably has a "sweet spot" or a range where increasing the gain starts to flatten the dynamic variation). How does turning up or down the treble at different pickup and tone settings affect the amount of fizziness? Where does "fire" and "gravel" start to turn into "fuzz", and is it good?

The "quest" is not to find a particular sound, just to keep tilting the amp in a different light until you find the most awesome sound it can produce for that riff. If you do this multiple times with breaks in between over the course of multiple days, you will start to find certain settings that you keep coming back to, and others that you keep disliking.

Hopefully, you are overwhelmed with awesomeness at lots of possible settings:

that's great, because it means that once you bring in bass, drums, and vocals, you'll have a lot of versatility to adjust your sound to suit the overall band. In the meantime, you can just pick one or do whatever you're in the mood for on any given day.

Unfortunately, you may also find that you just can't get a sound that you're happy with for your "signature riff." Time for a new amp (or time to check out some plugin effects). The good news is that all this systematic practice will have made you a gain-staging master, and you'll find it becomes much easier to tell what you're getting or not getting from an amp very quickly.

PS-- please do not waste any time on any of the above if you have noise problems. If your guitar hums, hisses, crackles, buzzes, then *get that fixed* or you're just wasting time. Noise is a separate topic, and all guitars will have a *little* bit of it, but if your "silence" is a buzzing, hissing mess of ugly then it's like trying to cook with rotten food.

As for some of your other thoughts, next post...

Quote:I know what a "scooped" guitar tone sounds like, and to me that guitar tone sounds scooped.(etc)...

The Cannibal Corpse sound probably *is* "scooped". But they've got different guitars, different amps, different *players*, probably different effects pedals, etc. You can't just put the same "recipe" on top of different ingredients and expect to get the same thing. If I'm using pork tenderloin and apples, and you're using tomatoes and fish, you can't just say "oh I know what yep did so I'll just do the same with this different stuff and it will come out just as good".

You have to start from the ground up and make it sound the best you can. That's what I hate about recording and audio "recipes"-- they make it sound so simple when in reality, nobody has the same ingredients, nobody even has the same measuring cups. The "recipe" says to take a filet mignon and grill it over charcoal for 4 minutes a side, but the people reading it are taking chicken breast and microwaving it for the same amount of time and then can't understand why it doesn't come out the same. And they get this idea of 4minutes per side stuck in their head.

When so-and-so famous engineer/producer/rock star says "I always cut this frequency by this many dB on this instrument", well, they're using a certain mic setup, in a certain room, on a certain instrument, and that's probably a good approach for them. If you're using a different instrument, in a different room, with a different mic, in a different position, through a different preamp, and playing in a different key, with a different player, who has a different voice, then their "recipe" is meaningless.

Look for *principles*, not prescriptions.

Quote:Originally Posted by infinitenexus
...I just don't quite understand how simply multitracking a rhythm guitar track can seem to make up for less distortion and more mids. I know that it does, I'm just not 100% sure how....""
=====

I don't know where you're getting this from. How do you know that? (And what does that even mean?)

Moreover, why do you care? If your guitar tone is already "spot-on", what more do you need to worry about? Just record the guitar tone and relax with a lemonade while all those other suckers with sub-par tone try to compensate with multi-tracking and all the rest of it...

Quote:Furthermore, I'm wondering if there are any notable benefits to using 4+ rhythm tracks as opposed to just 2. Seems like multitracking can only "build up" the sound so much, then it just gets cluttery. ...""
=====

Um, what are the benefits of using 2 instead of one? I would suggest that if one guitar track delivers the sound you want, stop. If it doesn't, re-visit the guitar tone.

Obviously if you have two guitar *parts*, then you'll need two tracks, unless you have a very remarkable guitar player. But there is no reason ipso-facto that a guitar part or anything improves by tracking it twice. In fact it often degrades.

Half the time people "keeping" the doubled track are just trying to "hype up" a bland, lame, or uninspiring part, performance, or sound. If you stack up enough random variations from multiple performances, it starts to sound like there is more going on than there really is. It's like the "fast food" approach to music production: if you add enough sugar, salt, and saturated fat to a piece of cardboard, eventually it starts to taste like food.

And there is a place for that. If you're an engineer trying to create an album in ten days with a band you've never heard before, and if their material is flat and repetitive-sounding, then you do what you can to spice it up quickly: multi-tracking, distortion, delays, "telephone" effects, arbitrary percussion tracks or random breakdowns, etc.

And sometimes it's not even that pejorative-- maybe you have a very talented singer, good songs, and a merely adequate band who doesn't know how to do much more than alternate one power-chord riff for the verse, and one for the chorus. Unless you are going to start giving out music and composition lessons, and they are able and willing to pay studio time for them, you have to record the material that comes in the door, and it might actually be good material that is just a bit "green" with regards to arrangement, performance dynamics, etc. So you do what you can make the backing tracks sound bigger and thicker and livelier than just looping the same riff for 32 bars over and over. Or maybe that's part of the "sound" everyone is going for: a fake, "produced" modern car-commercial sound. That's okay.

Quote:Of course, after some thought and observation I do realize now that much of the "thickness" we hear in Cannibal Corpse's guitar tone is, in fact, the bass guitar.""
=====

I think I understand what you're talking about, but I'm not really sure what you're getting at with this. Yes, there is a reason why even metal bands typically have a bass player, even though the bass is often just doubling the guitar riff: bass is usually better at lows than guitar is, no matter how much you de-tune the guitar. Even with piano (which extends lower than string bass), it is useful to have an instrument specifically focused on the low-end foundation, and playing with appropriate dynamics, tone, and note duration for that part of the sonic spectrum.

Quote:Although I'm very happy with my current guitar tone, I'm going to try your technique of turning everything down and then just using one knob at a time. I have high hopes for that technique.""
=====

Again, I'm not sure how this relates to your question,or if it does. There are other techniques than "mine", but mine would be something like: get the best and most appropriate sound you can with what you have to work with. And that might be one track or 20 tracks, it might be midrangey or scooped, it might be low-gain or high-gain...

The methodology doesn't really matter. If your guitar sounds the way you want it to, don't go killing all your inspiration with paralysis by analysis.

I think what might be getting mixed up is some earlier advice, intended for people who are unhappy with their recorded or live guitar sounds, to try backing off the gain and focusing on getting a better midrange (i.e. "guitar") tone, with the very specific idea that a lot of guitar players (and a lot of musicians, generally) are trying to do too much with one instrument. For example, the guitar player who wants his guitar sound to have as much depth, impact, sheen, sizzle, and fullness as his favorite records might be tempted to crank up the lows and highs and gain settings in an effort to try and make his guitar compete with finished records which include drums, cymbals, and bass... That approach often leads to a lackluster guitar sound that also drowns out and strangles the drummer and bass player, or that has to be turned down in the mix to the point where it just sounds like atonal fizz.

As I said four years ago, in the very first sentence of the very first post in this thread, if the title does not apply, feel free to ignore. It has never been my intention to tell anybody what to do or how to do it, only to help those who are having trouble figuring out why their recordings don't sound as good as they think they should. If what you're doing is working for you, my advice would be to ignore advice from me or from anyone else. There are plenty of people out there making better records than I've ever made, and far be it from me to argue with anybody who has got the sound they're after.

Cheers.

Quote:Originally Posted by infinitenexus
...if I had a two part harmony, would there be much difference in just doing 2 tracks panned some degree of left and right, versus 4 tracks, panned (just for example) 100R 70R 70L 100L?...""
=====

If a track/instrument really sounds good on its own, it is almost never improved by doubling it. Doubling is generally used to liven up or thicken up boring or weak sounds. The effect is usually that the doubled track has a more vague tonality and less definition, and more internal sonic "motion" and harmonic variation.

I.e., you would never try to "double" brilliant vocal by Pavarotti, since that would only muddy and smear the perfect vocal performance. But you might want to double or even quadruple a singer with a weak voice and mediocre intonation, to make the voice sound "bigger" and to make the pitch of the notes more vague, and to re-focus the listener on all the cool and varied throaty stuff that comes from four tracks, instead of the weak, tuneless vocal from one track.

It's purely a judgement call.

Quote:And for a second question: I use EZDrummer, and I've got it EQed so it sounds really nice. So that sound isn't going to be changing. Once I get my guitar tone 100% perfect, that won't be changing either, same with my bass and singing style and tuning. Since they won't be changing, my mindset was that I could essentially record one song, get my basic EQ, compression, etc as good as possible for that song and then use it as a template for the rest of that album. I would redo it for each new album/project, but within each album/project it seems to me that it could stay the same. What are your thoughts on this?""
=====

Setting aside whether EZ Drummer or your guitar tone are perfect, are any of your songs in a different key? Because the kick drum that perfectly complements a song in dropped-D tuning might sound flat and tubby and wrong with the same song in standard "E". Moreover, your guitar track EQ that finds the perfect low-end "thunk" for an open-E pedal tone might instead bring out tubby wub-wubs if you drop the tuning.

Similarly, are any of your songs a different tempo or time-signature? A reverb that is perfectly tuned to fill out a quarter-note part at 120bpm might just smear over and wash out a sixteenth-note part at the same tempo. A compressor set with a slow release time that perfectly pumps out impactful, pounding lows on a slow, heavy song might just smoosh over everything on a faster track, and turn it all into muffled random warbling.

So again, you can do whatever you want, but only your hearing can really decide whether it's right.

Quote:Originally Posted by infinitenexus
...All songs on each project of mine are in the same tuning. My main band is always in C#, and I have a death metal side project in B, and a kinda bluesy hard rock project in standard E tuning, so the tuning wouldn't change. Thanks again for your tips and insight, it's really taught me a lot.""
=====

"tuning" is not the same thing as "key".

I don't want to belabor the point, but there was a notorious internet thread often referenced by even non-audio-types as "Ha! Ha! I don't own a tube amp!", where a guy named JP22 started a thread where he described in detail how he wanted to record a guitar sound, with multiple mics and multiple compressors and all kinds of weird stuff, and then wanted the forum-goers to tell him how to make that setup sound good.

Knowledgeable people began asking JP22 why he was using this strange setup, and what he was trying to achieve with it, and JP22 would respond by insulting or attacking anyone who questioned his methods: he just wanted them to tell him how to make it sound good, not to question whether he was doing it right. The hilarity and absurdity of his position became an internet meme even among people who have nothing to do with recording.

I guess my point is that reality is what happens while you're busy making plans. Or, as Mike Tyson put it, "everybody has a plan until they get punched in the face."

There is only so much typing and debating that we can do constructively about pan position, layering tracks, templates, etc. In half an hour you could try a dozen mic positions and effects. In another half an hour you could try stacking four or six tracks versus two or one.

You can't think through this stuff, you have to do it. Preparation and understanding the principles and fundamentals can help you to make better decisions more efficiently, but you can't get great sound by scratching your chin and posting on the internet. Putting one foot in front of the other is usually a much quicker way to get where you're going than thinking about how to get there is.

There are a million books and threads all over the internet that will tell you how to get a great guitar sound: use this or that brand of strings, put this mic in such-and such a place, use a Pod set to "British Funk", use an all-tube preamp and germanium transistors, use a $100 guitar cable, use an illegal pick made from endangered tortoise shell, whatever... Everybody has a plan until they get punched in the face.

The internet is a great place for arguing and thinking about stuff. And one of the things I've found, is that the more you argue and think about something, the more you come to whatever conclusion you wanted to reach in the first place. But good sound and bad sound ultimately kind of trump all the arguments. Not many listeners are going to look up your forum posts to see whether you have proved logically that the record sounds good, if they think it sounds bad.

Quote:Originally Posted by infinitenexus
I don't mess with reverb much, mainly because I don't know as much as I want to yet. I know that a bit of reverb helps my vocals sound, well, better, and a little reverb seems to help my guitar solos "sing" nicely.""
=====

Let's skip reverb for now, since it's a topic unto itself. One thing to be aware of though, is that not all reverbs are created equal, and more money and/or processing power does not necessarily make for a better reverb. Try lots of reverbs, especially ones that claim to sound like "vintage" or "spring" or "plate" reverbs, and also impulse reverbs that model real-world spaces. Reverb is perhaps the least obvious, although the most audible of all effects.

Quote:I'm just worried that if I tackle each song individually, I'll end up with 10 tracks on the same album, all with 10 different guitar tones, and I don't want that.""
=====

Forget about this. Seriously.

Make every song sound the best you can make it sound, whether live or in the studio. You can tune your guitar to dropped-D and the dial up the heaviest, most awesome dropped-D, palm-muted power chord possible, and then try playing a lead up at the 12th fret and find that it sounds rubber-bandy and flabby and weak. Maybe you need two separate guitar sounds, one for rhythm and one for lead. Maybe a stompbox "boost" pedal can take care of this. Or maybe your awesome rhythm sound isn't really as awesome as you think it is. Maybe focusing on different registers and different keys will lead you to discover beefier, thunkier kinds of heaviness than you ever thought possible.

If you're a guitar player, your job is to make awesome guitar sounds. The engineer's job is to capture those. And the engineer's job is frankly a lot easier than the guitar player's job is, in that respect. The guitar should sound awesome first.

Quote:Originally Posted by DuraMorte
I sincerely hope you explore reverb within the context of this thread. It seems to be conversely one of the easiest effects to learn, but one of the hardest ones to truly master.""
=====

I think that reverb has been covered pretty extensively earlier in the thread. You can take a look at page 3:

http://forums.cockos.com/showthread.php?p=264886

CTRL+F "reverb" to find quite a lot of thoughts on the topic generally. Specific questions are easier to answer than just rambling over and over again on the same topics, but the short version is that reverb should usually be like movie makeup: you can use a ton of it, but if the audience is aware of it, you're usually doing it wrong.

It's (usually) supposed to sound natural and "real". In real life, reverb is everywhere, and even blindfolded, you can immediately sense what kind of space you're in just by the quality of silence. But you never really "hear" reverb as a specific sound unless you're in a very unusual environment like a parking garage or a concrete stairwell or something.

Reverb should usually be a subliminal effect, a sense of space and place and distance. Sometimes it's cool to get that echo-y, tunnel-y "wash", or the delirious and unnatural sound of psychedelic garage rock, but usually reverb should not be audible as "reverb", it should just sound like the same sound, except more "grounded" in a real space and place, with a realer sense of depth and moving air.

You can use high- and low-frequency damping or cutoffs to better fit the reverb "behind" the instrument and control splashies and tubby rumble, and you can use pre-delays to give that sense of space without pushing the sound too far back...

Pre-delay on a reverb is basically a control for how "close" the sound is to you: the longer the delay between the initial sound and the onset of reverb, the more it sounds like you are physically close to the instrument, because you're hearing the "dry" sound first, followed by the sound of the reverberating room. If you sit right in the front row, you'll hear the direct sound of the singer's voice singing at you, followed by the reverberation of the room behind you. If you're sitting in the back row, you'll hear the sound as a big wash of the source and the room reflections arriving at your ears at more or less the same time.

Decay and density should be "tuned" to suit the note durations and tempo of the performance. A sensitive musician in a big stone cathedral will intuitively tend to allow more "space" between notes to let the sound "bloom" in the space. This might manifest as playing at a slower tempo, or playing shorter note durations, or playing with a more aggressive dynamic attack to preserve articulation and clarity through the "wash" of sound. Similarly, the same musician playing an outdoor concert or in a "dead" curtained nightclub will tend to rush the tempo a bit, or to play with longer note durations and a fuller, fatter, less dynamic tone in order to fill out the dead silence between notes.

The engineer applying reverb after the fact is doing almost the opposite, but to achieve the same effect. A dry, spiky, choppy sound is often improved by a big, lush reverb that might drown out and wash over a richer, mellower, more sustained sound, and vice-versa. You can often sort of "tune" the decay time and reverb density to the tempo and note duration, such that the reverb tends to fade out at each beat.

Quote:Originally Posted by DuraMorte
...I guess I was doing it right on accident. Awesome.

...to give some breath and space to a dryly-recorded metal drumkit, in a mix with lean, bright guitars and a grungy, low-mid heavy bass, would it be safe to start off with a long pre-delay, damping in the upper and extreme lower registers, short to medium decay time, and a medium room size?...

Is that somewhat accurate? Or am I way off base?""
=====

Okay, everything from "I guess I was doing it right" to describing which settings you should use is completely the opposite of the stuff I can help with.

I really don't believe in recipes or "rules" that allow you to verify that you're "doing it right" by describing it on the internet. I can no more answer this kind of stuff than I can tell you which notes to play after you've played G and then E... What matters is whether it sounds good, not whether you can barely hear the reverb or whatever.

The point about reverb being subtle subliminal doesn't mean that tiny amounts of reverb sound best... a ton of reverb can sound pretty subliminal and inaudible if it's a good reverb and used appropriately, just as movie makeup artists might cake on a pound of makeup to make the actors look "natural" and "un-made-up" under the lights and cameras.

Maybe you have been doing everything right by accident, or maybe your instinct is correct that you've been under-doing it.

The purpose of most "non-special-effect" reverb is not usually to make the instruments sound splashy, tunnely and metallic, it's to create a more natural, realistic, and immersive sense of space, place, depth, and size for otherwise dry, close-miked, disconnected-sounding instrument tracks.

You know how when you take a snapshot of some people at a party, they sometimes come out looking sort of greenish or yellowy or ghastly gray-pale, because of the lighting? That's why actors get makeup, even though they're not supposed to look "made-up". The point is to make them look normal and natural under all the lights and cameras.

Same thing can happen when we record 40 tracks of close-miked instruments and samples all recorded with different mics shoved right up in front of them, at different gain levels... you play it back and everything sounds sort of disconnected and floating and random. "Reverb" is one of the kinds of makeup that help to fix or improve the sound, so that it looks less like a weird photograph and more like the actual people we were trying to take a picture of. Whether it's an actual reverb box, or a set of room or overhead mics, or the sound sent back out through a PA and re-recorded with room mics, or some arrangement of short delays, or putting a speaker in a bathtub and recording it from the hallway, or a plugin with one knob or 20 knobs, what matters is not which settings you adjust in which order, what matters is how it sounds coming out of the monitors.

You can get an A+ in "theory of movie makeup" class (if there is such a thing), but if the actors all come out looking grayish-green and sick, then you're not "doing it right" no matter how many times you double-checked the recipe against all the books and internet posts. On the flipside, unless the point is to win internet arguments, as opposed to making good-sounding records, it doesn't really matter what brand or type of reverb you use or how many knobs it has or how you adjust them.

What matters is whether it sounds the way you want it to sound. And there is so much going on before we even get to the reverb that there is no way for me to tell you whether "put X amount of this setting on drums for good sound." What does "drums" mean? How were the drums recorded? Are there overhead or room mics? What kind of drums were they? Who was playing them?

We can, to some degree, make "recipes" for stuff like guitar amp settings, or synthesizer patches, because there are relatively few variables: if you take the same synthesizer and dial in the same settings, and play middle C, it's generally going to produce a very similar output signal to what I get when I do the same a hundred miles away.

But as soon as you get into actual acoustic instruments and voices producing variable and non-electronic sounds that have been sent out into the open air of a real-world room, and then recorded with a microphone, then half the equation is unknown.

I think this is often a big part of why musicians have a hard time switching to engineering... they know that if they play a Les Paul through a Marshall JCM half-stack with the pickup switch and volume and tone knob set to such-and-such, and all the amp knob set just so, then it's going to sound pretty much the same as last time, unless the tubes are going bad or something. But when you try to take the same approach to recordings of real sounds in open air, this approach breaks down.

You can't just "plug vocalist into channel A and set the gain knob to halfway and the presence knob up three quarters and the bass knob at three with the bright switch engaged" because we don't know anything at all about the vocalist, or the mic, or the distance from the vocalist, or the room they're singing in, etc.

In your own work, in your own world, in your own room, with your own gear, and your own voice, you might very well find some very useful "recipes" similar to basic amp settings. But your voice was not made in a factory somewhere, and your room is not a guitar amp sold in a catalog, and your mic and placement are not fixed-position pickups that only pick up the direct magnetic vibrations of your vocal chords. Trying to use someone else's "recipe" is like trying to get a Danelectro guitar played through a Roland Jazz Chorus amp to sound like Death Metal by copying the knob settings. Now, you might be able to get somewhere by focusing on the SOUND, and then manipulating the gear that you have, to try and push it in that sonic direction, but it's never going to be a "recipe" that you can just follow, because you're not cooking with the same ingredients.

Quote:Originally Posted by sm7x7
...Now, I do my sequencing in Reason, render separate tracks to WAVs, and do the mixing in Reaper. Now, if you please share anything specific to my setup (regarding some new things about EQing and so on)...

Nothing to it. What you're doing is no different from someone recording a room full of actual instruments. In this case, though, Reason is your band.

That said, working with electronic instruments is often slightly different and somewhat easier than working with "real" instruments, because you can pretty easily modify the actual sounds as you go. Moreover, since the musician and "engineer" are both hearing the exact same thing as you go, presumably through the same monitors and at the same volume, a lot of the hard work is often done by the time you hit play in Reason.

IOW, you are spared the difficulty of getting a big, booming, subsonic room-filling kick drum sound to fit inside some little 6" monitors at 80dB, because the sounds you have already created/selected were selected to sound good "in the mix" so to speak.

All the same considerations apply, but you may have deliberately or just intuitively done a lot of prep work by the time you're ready to mix.

One thing that might be worth thinking about is that mixing sequenced music is often as much about arrangement as it is about sonics. Music constructed in a sequencer can often be prone to sounding more "static" and unchanging than music hashed out in real-time rehearsals. There is a lot you can do with shifting delays, breakdowns, special effects, eq sweeps, etc to try and add some dynamism and sonic movement. It's kind of hard to tell what we're talking about with tracks that come from reason, since that could mean anything from a handful of loops repeated over and over to a massive "played" orchestration that has as much variance and "humanness" as a lot of actual bands.

If you're doing loop-based dance, hip-hop, or club music, then you're probably already familiar with a lot of the "tricks" of electronic music. However you may find some new challenges trying to mix in "real" vocals, especially related to having the vocals seem disconnected or "floating on top of" the backing tracks (FWIW a lot of "real" bands also have this problem, especially if they are unaccustomed to rehearsing as a full band with adequate monitoring for everyone to hear the whole mix).

What happens is typically that the self-producer manufactures a great instrumental track, and then tries to put the vocal on top of it, almost as an afterthought. I hope people who have been following this thread can start to see the problem before I've even said it...

The vocal is the most important track in the whole thing, and none of the other musicians have really even "heard it" until after their tracks have been laid down. And as anyone who's been doing this for any length of time at all knows, what works in theory doesn't always come together right when you actually put all those ingredients together in the same pot.

This starts to get a bit outside the scope of this thread, and more into the "Producing Yourself" spinoff, but one thing I would strongly recommend is recording a vocal scratch track very, very early in the process of creating any song that involves a lead vocal part. Otherwise it becomes really easy to end up with an instrumental bed that creates a ton of problems for the vocal. Common symptoms include generally being too busy and dense to fit the vocal in, stepping all over the vocal range, having instruments build to complex crescendos while the singer is trying to sing something important, having transitions that cut short the vocal line (the old "singer gasping for breath and rushing syllables because there aren't enough beats in the transitions" syndrome), and generally having the "band" sound like they hate the singer and refuse to pay any attention to what she's doing.

Quote:Originally Posted by sm7x7
Wow, YEP, you're like a human x-ray: you read my mind with all my previous sequencing experience at once! All described above is exactly what I always complained for and not knowing the solution for it. Now I'm ready to start with the DBV method, and I'm sure it will make a difference.

One more question:
Since all sounds are already prepared somehow sonic-wise, what's your advise for EQing them (HPF/LPF)? Same as for regular instrument tracking or some specifics ? I know, I know, beholder and his ears, but sometimes you watch and don't see, or hear and don't listen (or vice versa?) Also, I'd appreciate some panning hints (I already bought and read Nickolas's ReaMix book and have some ideas), because, listening to the AUDIO'S Audiophile Reference CDs, I'm amazed of the wonderful space positioning of the instruments and vocal part. Thanks again for sharing your knowledge with all of us noobs (40+ years of my music experience doesn't tell I know anything good enough, and older I become more I know that I know nothing ;-))

P.S.: Excuse my "rusty" English - I'm a native Ukrainian living out of that country for a couple of decades... 8-\""
=====

Your english is a lot better than some native speakers!

RE: EQ, my opinions aren't much different than elsewhere in the thread. Using samples and synthesized sounds (hopefully) gives you the benefit of not having to worry so much about rumble, hiss, hum, etc, but I'm still a big fan of at least experimenting with high- and low-pass filters, just to see how much you can get away with.

Very often an awesome bass sound and an awesome kick drum sound are doing a lot of the same things, and you can often improve both by, for example, cutting out the extreme lows of the bass to let the kick have a cleaner, more focused punch and impact, and also by cleaning up some of the lower mids of the kick to let the bass really fill up the body of the sound without getting muddied up by the atonal "boom" of the kick drum.

Same thing on the high end: a lush, airy pad or sparkling cymbal might sound great on its own, but put the two together with some lead and backing vocals and the high end might start to turn into a washed-out pile of essy white noise, especially if you start to bring in reverbs or delays. High-frequency "Air" starts to seem a lot less "airy" when it's full of stuff. You need some space up there for the sparkles to sparkle and the shimmers to shimmer and the breathing to sound breathy and so on.

RE: panning, I change my mind all the time on this topic. Some things to think about as follows... Different schools of thought:

- School 1-- Pan it like a "real band" onstage: drums and vocals center, bass mostly or totally centered, leads either stage left or stage right (e.g., "organ" to the left, "guitar" to the right, even if you're not actually using "organs" and "guitars"), backing instruments, secondary percussion, pads, backup singers spread in-between (e.g., a congo/tambourine player slightly right, a cluster of "soul girls" slightly left, etc). This often gets a very "natural", spacious,and immersive sound, especially for listeners with good playback systems.

- School 2-- "Big Mono": similar to above except looking for symmetry instead of difference, especially with modern layered, double-tracked stuff. So hear you maybe have the "guitar" double-tracked with one part hard left and the other hard right, the "organ" sent through a wide-panned stereo leslie sound or stereo delay, the "soul girls" panned across the whole stereo spread, the drums and percussion panned across the whole stereo spread, complete with tom rolls that go all the way from left to right, etc. This can produce a very "big", "modern" although somewhat artificial and phase-smeared sound that often works especially well on headphones and mediocre car stereos.

- School 3-- "Three Cardinal Points" or "stupid stereo": pan everything either dead-center, hard left, or hard right, like those awesome old records from the 60s where all the drums were in one speaker, all the guitar and reverb was in the other, and so on. The cool thing about this approach is that it avoids the phase-smeared "no man's land" of "in-between" pan positions and is extremely forgiving to poor speaker placement or playback rooms. It also allows for a great deal of clarity, headroom, and sense of individual parts, since you can, for example, pan the hi-hat to one speaker and the tambourine to the other, pan the pads to one side and the "soul girls" to the other, etc. You can basically create three completely separate mixes that each play from one of three locations: Left, Right, or Center. And you can use those distinct locations to localize each instrument and maximize clarity and headroom for each frequency range and instrument type.

- School 4-- "moving stereo": this is basically the practice of moving the different tracks around in the stereo field, which can mean anything from gimmicky (but possibly cool) "whooshing" effects or instruments "bouncing" from left to right, to very subtle and gradual sliding to different positions. Whether obvious or subtle, this can help direct the listener's attention to sonic subtleties by shifting the phase and frequency relationships of the various instruments, drawing attention to this or that part of the sound depending on what's being masked or revealed by the content in each speaker channel at any given moment. On the flipside, it can also become a somewhat goofy or focus-distracting diversion for both the mixer and the listener.

Another thought is to give some serious consideration to mixing in mono, and then start to pan stuff out when you need more "space". Frankly, mono is vastly under-rated, and quite a lot of modern approaches to pan are basically just the product of people feeling like they have to do something with these two speakers. It's not always necessary or even desirable to force yourself to find the "perfect place" for every track-- club PAs and live sound-systems are nearly always mono, and an awful lot of modern playback systems might as well be... when someone is listening to a little 2.1 computer system or table radio from 3 meters away, those two speakers a foot apart are not really doing much in terms of delivering "stereo sound" to the listener. Moreover, it's got to sound good in mono anyway.

Last but not least, it may be stating the obvious, but there are sound technical reasons to put the vocal, snare, and low-frequency instruments (bass and kick drum) dead center, or close to. Low frequencies devour a lot of headroom and require a lot of energy to reproduce, and both your record levels and your listener's speakers will thank you for allowing them to make use of both channels. Moreover, low-frequencies are much less "localized" (i.e., it's hard to tell where they are coming from, so there is less benefit to "panning" them).

Lead vocals are obviously important, and should be audible even to people standing on one side of the room, or listening to a system with a blown-out speaker or a stupid setup where one speaker is in one room and the the other is in another. Both the driver and passenger in the car typically want to hear the vocal clearly.

Similarly, the snare drum (or whatever is hitting the backbeat accents) is not only extremely important to almost any conventional mix, it's also usually the loudest "peak". Having that instantaneous "crack" or "snap" poke out above the average music level is a huge part of what gets heads bopping, fingers snapping, hands clapping, girls dancing, and all that good stuff. In almost any normal mix, the very loudest transient "peaks" (the ones that first start to clip when you turn up the level) are on snare drum accents. So it makes sense to give the snare access to both channels of dynamic range and headroom, in order to get the biggest, loudest relative "pop" compared to the rest of the music.

So, while there are no "rules", it usually makes sense to keep the vocals, bass, kick and snare panned center, or close to.

Quote:Originally Posted by brainwreck
here's one of the biggest problems i have with recording myself. i can record/mix with either decent balance in mind or feel, but almost never both, which pisses me off to no end. 99% of the time, if i just record without thinking about having a good mix, the feel is there. but as soon as i begin tweaking the balance of things, i lose it. most times i don't even notice it sneaking off undetected until i think i have a good balance going on. then i'm like, what the hell happened? that exciting piece of music with a way too loud kick just turned into boring crap with a kick that's sitting fine. i know for sure that the less i mess with things after the fact, the less chance of killing the vibe, and i try to do as little after tweaking/ processing as i can. i'm tired of getting chumped by my own self. help yep, lol.

Wow, fantastic post. You are certainly not alone, and this is exactly why major labels often spend months in the studio and huge amounts of money with a team of professionals to try and make a good record of 10 or so 3-minute, four-chord songs by a five-piece act. I mean, in theory, if the person doing the "recording" knew what she was doing, shouldn't it take about 30 minutes to record that "set" of 10 songs?

Or even if took two days to set up all the mics correctly, and the band had to re-play each song 15 times to get a good take, then 15 x 3 minutes= 450 minutes, or about one 10-hour day per song, even allowing for breaks? So shouldn't that be about two weeks to record an album, even allowing for some things to go wrong?

What's great about your post is that you have hit upon the very nut of the problem. The band gets together and, over many rehearsals, works out some material that achieves something really good-- the hairs on the back of the neck are standing up, you're all in the zone, you can't help but bop your head, the music is taking over your hips and feet, the audience is feeling it and into it... so you set up a microphone (or a dozen microphones, or 4 dozen mics), adjust all the levels so nothing is clipping, and hit record...

Then you play it back. And it sounds pretty cool, it still has that hip-wiggling, hairs-standing-up effect, but only if you play it back at really high volume. Moreover, the kick drum sounds too loud and muffled, the vocals are indistinct and a bit tuneless in places, the cymbals are a hashy mess, etc...

The thing is, you KNOW it was good. You were FEELING it, and the audience was too. And in a sense, it still IS good. The problem is that it tastes like a great restaurant or homemade meal that was wrapped up, stuck in a freezer for two weeks, and then reheated in a microwave. It tastes dried out: the onions are too strong, the meat tastes like nothing, the tomatoes have turned bitter and bland, the potatoes have turned starchy and dry... it still reminds of the great meal you had, but it's just a reminder, not the real thing. What happened?

In the food world, it's a lot of fairly quantifiable chemical changes that occurred, along with a few harder-to-quantify changes: If you take a steak, and dehydrate it down to beef jerky, it's never going to taste like a steak again, even if you soak it in water. The food industry spends billions on trying to find ways to make frozen, jarred, canned, and powdered foods taste as good as the real thing. The US military in particular has massive teams of chemists and chefs dedicated to trying to come up with packaged, non-perishable foodstuffs that are both palatable and nutritious, and any veteran can testify that they still have a lot of work do.

In the audio world, we have some similar challenges, but the fundamental problem is that same: how do you take (for example) fresh tomato sauce made tonight, and put it in a jar, so that somebody 1,000 miles away and six months from now can enjoy the same experience?

We have now moved from the creative and intuitive business of music-making (or cooking), into the technical and analytical business of audio-engineering (or prepared-foods manufacturing). In either case, we need to START from the same creative place, but the process of preserving and packaging it might be just as complicated and time-consuming.

Fortunately for audio, it does not degrade nor dehydrate nor undergo chemical changes over time the way food does, so the technical challenges are a lot simpler. But the effect is often very similar: a brilliant, ecstatic, emotionally overwhelming concert from the night before can often sound like a muffled and uneven mess the day after, as though it had been put in plastic wrap and left in the freezer for a month.

That essential question of "feel" vs "sound quality" is at the heart of modern big-budget approaches to record-making.

Simple decibel level and environment is the first thing: Last night, in a packed and sweaty bar, the band was playing at 110dB before a crowd of people who had nothing to do but drink and watch the band, and who had come out to be entertained. Everyone from the musicians to the bartenders had an investment in making it a good time, and the focus was entirely on the music.

In the cold light of tomorrow's hangover, sitting alone in front of a computer screen, listening at a non-roommate-bothering 60dB, everything sounds a lot less impressive, a lot sloppier, and generally looks a lot more like the girl/boy you woke up to after a one-night stand than the one you went to bed with the night before. It tastes like leftovers. Even if you didn't drink.

So a big part of it is starting from a clear-headed and sober approach to sound quality and sonic "presentation", independent of the quality of the music. The earliest movies were basically static-camera captures of stage plays, and they are kind of the worst of both worlds. Watching a play means actually being in the same room with the actors, watching it happen, with your eyes and ears free to focus on any aspect of anything. Moreover, it's all genuinely happening, it's real people right in front of you... This makes stage plays both more demanding in some ways and less demanding in others than movie-making.

When you watch a movie, for starters, you are seeing only one point of view. The camera decides what is in focus, not your eyes. You're not looking at real people and real events, but at a screen, so the threshold for suspension of disbelief is much higher. On the other hand, a skillful director has a much greater ability to manipulate your perception, provided that she is able to sustain that suspension of disbelief.

When we watch a play, we see stuff actually happening in front of our eyes. When we watch a movie, we see instead someone else's vision of what those events should look like to us, which increases the artistic potential of the specific filmmaker's vision, but decreases the immediacy and power of actually watching it happen.

A similar thing happens with audio recording. If the fights depicted in, say, "Raging Bull" had unfolded exactly the same way but had been merely captured by a static camera in seat 3B, the film audience would not have had nearly the same effect, although the live ticket-holder sitting in seat 3B actually watching, hearing, and feeling the blood, bones crunching, spit and sweat, and leather gloves swiping skin would have probably had a more intense experience than either film or stage could ever convey.

This is the job of the producer/engineer/recordist... not to simply set up a camera in seat 3B and record the fight, but to, as much as possible, recreate the sensation of actually being in that seat, which is a very different thing.

The person sitting in seat 3B had the realest and most visceral and intense experience of that fight of anyone other than the fighters themselves, but a simple camera and mic, no matter how good or high-quality, would ever have conveyed the violent intensity of the fight so well as Martin Scorsese's adept film-making did.

More to come...

Leftovers continued:

The second major part in all this that tends to get overlooked is plain old musicianship.

The Boston Symphony Orchestra pioneered a method for auditions that put the musicians behind a curtain, such that the judges could not see them, but could only hear what they were playing.

Now, by the time you are credibly auditioning for the BSO, it's a pretty safe bet that you are already one of the best in the world. You already have a good instrument and are a better-than-competent sight-reader. You are playing centuries-old pieces that millions of people have played before, and have had access to dozens if not hundreds of recordings of the material.

IOW, whether it's an audition for triangle or lead violin, the audition process is not [i]whether you can play it[/], since that's taken as a given, but how good it sounds when you do.

It is worthwhile to note that, with top-flight orchestral players, we are often talking about players who have been loaned multi-million-dollar instruments based on their talents; e.g. some museum or collector has loaned some piece of world history to a college student based on their ability to play. This is a categorically different world from Squier vs. Fender strats, or owning a '55 Les Paul or original Rhodes or Moog... we're talking about centuries-old priceless museum pieces being played by the most talented and best-trained people in the world, playing pre-written music that anyone can learn. And they are being judged based solely on how good the exact same piece of music sounds.

Just in case you're not entirely clear on this point, what it means is that the musician matters. It's not just what you play, but how you play it. Musicians sometimes get sensitive and defensive about this. But as an analogy, consider joke-telling, or acting: two different people can tell the exact same joke or deliver the exact same line, and one is often much better than the other.

This might sounds like some kind of snooty "you're not good enough to make good records" but it's not. Instead, it should be almost the opposite: an admonition to focus on what you're good at, and to let the "feel" and hairs on the back of your neck be your guide. You're not auditioning for the BSO, you're making your own music.

Way too many musicians waste way too much time and effort trying to be something they are not, trying to prove something to someone else, instead of simply doing the stuff that they can do well, and that they can do expressively and meaningfully. As Yoda put it:

"All his life has he looked away... to the future, to the horizon. Never his mind on where he was."

More to come...

RE: recording to a click, pros and cons... It is certainly possible to make really good records to a click, and it may in some cases be extremely difficult or impossible to make good records NOT tracked to a click. (As an aside, REAPER offers some outstanding tools for tempo-mapping to a "live" non-click performance, see here and elsewhere: http://www.cockos.com/wiki/index.php..._Freetime_Song)

Whether or not to use a click is usually dependent on a number of factors. I suggest rating each of the following on scale of 1-5:

- How "tight" and generally good the band is, and how close they are able to come to performing a "perfect" take live and without overdubs. Rate 1-5, where one is "band plays perfectly every time", 3 is "band can usually play the song through with only a few mistakes", and 5 is "band cannot play the song all the way through without losing tempo or making serious mistakes"

- How much the ultimate recording will depend on loops, samples, or midi sequences. Rate 1-5, where 1 is "whole band plays whole song live, only thing to overdub is backing vocals, etc", 3 is "band plays whole song live, occasional midi pads may be overdubbed, no loops", and 5 is "project makes heavy use of loops and/or sequenced parts such as electronic drums".

- How much use the band makes of deliberate and controlled tempo or time-signature changes for artistic effect (one is band frequently uses tempo/time changes, 5 is band never intentionally changes time or tempo mid-song)

If the project scores all "ones", go ahead and skip the click. If any score is a 4 or 5, use a click and don't ask questions. If the scores are mixed from 1-3, then using a click will probably make life a lot easier, but skipping the click *might* make for a better-sounding record.

Notice that none of the above scores are "band has a hard time playing to a click". Any musician who cannot play in time to a metronome is automatically a candidate for recording to a click, and should probably go home and practice with a metronome for a week or two before we start recording. Frankly, if they can't play to a click, then they cannot play in time, and they have been fooling themselves all along. Such musicians MUST be recorded to a click, since the reason it's hard for them is because they don't play in time to begin with. Which leads to....

A separate but almost equally difficult culprit is the good "emotional/expressive" the singer-songwriter who strums chords or tickles keys willy-nilly while singing atmospheric and sometimes semi-tuneless vocals. These timeless wonders can often be spotted by a tendency to sway their head around randomly as a time-keeping measure, while their feet remain motionless (people accustomed to playing in time have a tendency to tap their feet and/or to bop their head on the quarter-notes).

Note that these are NOT bad musicians, and may actually be quite good, and have extensive classical training. It's not that they cannot keep time, it's that they approach timing like a maestro conductor would. I suspect but cannot prove that certain very good and well-known musicians, possibly including Leonard Cohen, Nick Cave, Tori Amos, and Dan Bern either started out or still do work this way, to varying degrees.

This is a very difficult breed of musician to try and accompany or produce. Their timing and delivery is often more like a free-form poetry reading than a musical composition, with piano chords or guitar strums delivered as accents and punctuation to the sort of poetic and impressionistic timing of the vocal "melody", which may only barely even qualify as such.

Please note that the above musicians are absolutely CAPABLE of playing in rigid time, they just don't always write or perform that way. For example, I am certain that Tori Amos COULD play the song "Winter" in strict time, but it would sound a lot more sing-songy and nursery-rhymey. IOW, the problem is not that she couldn't play it to a click, the "problem" is that the song is meant to have a lurchy, time-draggy, dreamy/emotional feel (probably a big part of why there are no drums).

The hard part of trying to fit a "beat" or even a proper "melody" to such songs is that, when you get a full band, or even just a drummer, trying to play that kind of material, it starts to sound like the band is making mistakes. Are those "expressive" devices extra beats, or are they time-signature changes, or are they tempo changes? How do you "notate" such a performance? And if you can't really quantify it in that respect, then what the hell is anyone else supposed to play?

So long as it's just solo piano and vocals with some string swells (like "Winter"), we're in the clear. But try to add drums, a bassline, a guitar riff, or backing melodies to something like that and you have quite a row to hoe ahead of you. Moreover, punch-ins, edits, and replacements are going to be almost impossible.

This is the kind of thing where you really either need to record the solo performance straight through, and the solo performance has to be strong enough to stand on its own as a finished record, or you practically need a tyrannical and brilliant symphonic conductor to lead a highly-trained orchestra or ensemble who can follow "expressive" time and performance instructions.

Again, note that this is totally different from musicians who cannot play to a click. They may try to claim that their material is too expressive to fit to a grid, but never believe it. The test is that they CAN play it to a click, but that it sounds worse.

Very rarely, you may be lucky enough to work with a band or ensemble who can collectively play with this kind of "expressive" timing (more commonly, you will work with sloppy and messy-sounding musicians who claim that the click robs the music of soul and feeling when in fact they simply can't play in time-- more on that later).

So what to do if you are, or are trying to record one of these soulful genuises of poetic time?

Well, for starters, just record a solo piano/vocal or guitar/vocal take straight through, and get the best one you possibly can (hint: if the musician cannot both play and sing the song all the way through then they are almost certainly just unable to keep time, which is a completely separate problem-- true "expressive" timing is the province of very capable musicians).

Next, assuming the musician is genuinely "working the beat" as opposed to simply playing out of time, you will almost certainly be able to tap your foot or finger reliably to at least one beat in every measure. At the risk of stereotyping, with white/classical musicians, they will tend to always land the "one" more or less on the beat, sometimes the "three" (like either ONE-two-three-four-ONE-two-three-four or, less commonly, one-two-THREE-four-one-two-THREE-four). Black musicians or musicians raised on gospel/R&B will tend to keep either the two or the four steady, or more commonly both (like one-TWO-three-FOUR-one-TWO-three-FOUR, etc), or occasionally the "and" in-between the beats (like one-AND-two-and-three-and-four-AND-one-AND-two-and-three-and-four-AND). If you're dealing with indigenous African music, authentic Irish folk music, or traditional Eastern European music, then god help you, because they are often following insanely complex time-signatures based on the cadences of spoken phrases in languages you can't speak, and trying to fit that into a grid turns into things like trying to count out 27 64th notes per measure with accents all over the place.

But setting that aside, and assuming we're dealing with more or less conventional music in 4/4, 3/4, or something similar, a musician playing with "expressive" timing is still usually playing more or less consistent measure lengths, they're just not necessarily playing every beat with equal time durations.

So if you listen to a song like Tori Amos' "Winter", you can probably predict and "tap" at least the "one" on every measure. If you were producing this song for her, then you could probably sit at a piano and hit the root note of the relevant chord at every "one". Having done that, you might be able to try a second pass and hit all the ones and threes, or even most of the beats. If you were to record this, you have the beginnings of either a tempo map or a reference track for the backing musicians to work out parts to. You may still be fairly limited in terms of what you can do (the album track has only strings and backing vocals, for example).

Stereotyping once again, "black" (and R&B/gospel-style) musicians are often considerably more adept at working the beat and being expressive with timing within the constraints of a very "tight" backing band.

Otis Redding's "Dock of the Bay" is a great example of a vocal performance that is so sensitive and attuned to the timing that he is able to achieve quite a bit of expressive internal "stretching" of the beats, while always landing exactly where he should. The band is nearly metronomic in terms of "tightness", but Redding seems to find ways to fit extra beats into the measure, and to achieve a very naturalistic and conversational vocal delivery without ever having to rush or crimp the note duration. A lot of bar band singers do a very poor job of this song, failing to anticipate the ebb and flow of accent and passing notes. What makes Redding's performance special is not that he is ignoring the measure and beat divisions, but that he has such past-mastery of them that he is able to completely re-write them on the fly-- most of his syllables don't land on beat divisions, only the ones that count. And they land there not because he suddenly rushes the leading beat once he starts running out of measure, but because he phrased everything in advance to land casually on the right beat.

IMO, this is a better and more effective approach for most popular music than the Tori Amos approach of forcing all the music to ebb and flow with the singer's immediate emotional intent, but it also one that requires a very, very adept singer with a very nuanced and skilled technical sensitivity to both timing and to vocal delivery.

So what about those of us who have to admit that it's not really about "expressiveness" but more about having material that's just not really in time? More to come...

____

So what do you do if the problem is not the click, but you? Or stated differently, what happens when you have a good song, but the click seems to mess it all up?

Fortunately, there are often some very easy fixes.

The most common culprit here is usually lyrics that contain too many syllables for the melody or musical accompaniment. And the most common sub-culprit is writing a lyric that ends on a "one" or downbeat when the accompaniment or melody expects to start the next line there.

It's impossible for me to really illustrate this without including music notation, which I expect is pointless for most of those who suffer from the problem, not to mention a lot of work for me. But by way of example, we might look at the following couplet:

Baby you're the best
You're the sun and the moon and all the rest

Now, aside from the fact that the above is very bad poetry, as an informal poetic couplet it works just fine, which is to say, it rhymes. But as two lines in a 4/4 song, it presents a serious problem. The first line is okay, with three stressed syllables:

BAB-y YOU'RE the BEST

That's okay, in fact, it gives us room to bring in the first beats of the second line on the fourth beat, if we like:

BAB-y YOU'RE the BEST, YOU'RE the

But we still have one or two too many syllables to fit into the second line. Now matter how you distribute the accents, there is no non-weird way to say:

sun and the moon and all the rest

inside of four accented "beats". Tap your fingers four times, over and over, going: index, middle, ring, pinky, and counting "one, two, three, four" (this is a good habit to get into when analyzing music of any sort).

There is no non-weird way to say "sun and the moon and all the rest" inside of four taps. I can't make it sound natural in less than six.

Now, there are a number of ways to approach this. One of the worst ways, and the first resort of novice songwriters, is often to try and force it: e.g. sing the second line like "sunandthemoonandalltherest" and try and do it fast enough so that you get it out before the next lyric is due. But that sounds stupid.

For another, we could re-write the lyric to have more or fewer natural "beats" to better fit four beats per line. Maybe something like:

Baby, bay, you're the best
You're all that and all the rest

And now we're getting even dumber.

But assuming we want to keep the lyric, a better approach is often to insert more musical "beats" between the syllables to extend the vocal melody. So maybe something more like:

Ba-a-by-y [4 beats]
You're the best, (Pause)[4 beats]
The sun, and the moon, and [4 beats]
all the rest (pause)[4 beats]

I'm not sure how much sense the above is making, but the point is that by stretching the word "baby" to four beats instead of two, we were able to turn a two-and-a-half measure couplet neatly into a four-measure couplet.

Another approach would be to extend to the couplet halfway into the third measure, then have a two-beat rest, then have the next couplet start.

If you're really good at lyric-writing, you could do all kinds of slick and clever stuff with internal rhymes, but for most of us, it's kind of important to get every line to end on either the four or the one of a musical measure. And if it ends on the one, that usually means that it's the end of that lyrical passage, since it has "blocked" the "next line".

I.e., normal "internal" rhymes should usually end on the four, while "final" or "closing" rhymes can end on either the four, the following one, or the following two.

The same as above applies to musical and not just lyrical ideas.

If you listen to the final "fadeout" note or chord on practically any song that actually ends (as opposed to songs the just have an engineer fade out on a repeated chorus or some such), it never ends on the last beat of the last measure, instead it ends on the FIRST beat of the last measure. It's like:

1,2,3,4: BAHHhhhh....

One of the problems that one-measure-at-a-time songwriters get themselves into is writing ideas one couplet or measure at a time, that END. And like the above lyric, they tend to END on the next measure. Which is okay, except that their next idea starts on the one or downbeat of that same next measure, which means the next lyric or melody is starting before the last one is finished.

Do not delude yourself into thinking that you're writing in complex time-signatures: writing something that sounds good in 5/4 or 7/4 is a complex and sophisticated task that requires a fairly sophisticated mastery of the more basic time signatures.

What you are doing is writing basic 4/4 or 3/4 or 8/4 or whatever-type stuff that includes lines that extend into the next measure.

So your first task is to sit down and sing your lyric (and any relevant musical breaks), and just tap out 1,2,3,4,1,2,3,4,etc throughout the whole song. Just put your right or left hand on your thigh, and tap out forefinger, middle, ring, pinky, counting "one, two, three, four" as the metronome plays, and humming the core melody.

If this gives you trouble, I guarantee it's not because you have written a too-complicated song for conventional time-signatures, instead it's because you don't have a good grasp on the song you have written (if it IS a more complex time-signature, such as 5/4, just start including your thumb, or if it's 7/4, just start counting "thumb, pinky, index, middle, ring, pinky, etc) But I guarantee you haven't accidentally written a song in complex time. You've just written a song with sloppy time, that you can't play consistently.

To the point, Tori Amos, Dave Brubeck, and Otis Redding would have no trouble whatsoever playing or singing precisely to a click. It might not sound as good, but they'd be able to play it like a metronome if you asked them.

Playing to a click doesn't crimp Otis Redding's style, although it might crimp Tori Amos or Rachmaninoff. If it crimps yours, chances are it's because you're not playing in time to begin with.

Quote:Originally Posted by MCV
...part of what you said now kind of touches that subject: one of the main tools engineers use for conveying a recording that tries to mimick the emotion of a full blast live act is compression & limiting, so that you get that feeling of loudness at lower volumes. I'll try to find the time to elaborate in the other thread...""
=====

We're mixing topics a bit here, but being "anti-loudness-wars" does not mean being anti-compression.

In fact, I am very pro-compression. Compressors are about my favorite effect-- they can be the most subtle, musical, and creative tool available to the engineer, and they can not only solve a world of technical problems, but they can also sound really, really cool if you know what you're doing.

What I expressed opposition to in the other thread is the "loudness war" approach, which I have mostly tried to leave out this thread, because I hope it will be useful to people on either side of that somewhat nerdy debate.

There is a very big difference between using compression to make something sound "better", versus using compression to make something that already sounds good sound "louder". Mastering considerations are mostly outside the scope of what this thread was intended to be about. If this thread has helped anyone to make better-sounding recordings, then I've achieved what I hoped for, and the rest is out of my control.

My opinion is that the "loudness war" attempt to make "louder" records by pushing record levels closer to full-scale is self-defeating and counter-productive, plus I think it sounds bad.

However, I have no objection whatsoever to sculpting sounds in such a way that makes them "sound" louder (at any volume). In fact, I'm a big fan of that. If you're sitting on your sofa and your kid is popping bubble wrap next to you, and then someone sets off a firecracker half a mile away, both the bubble-wrap-popping and the firecracker might hit your ears at the same dB SPL, but the faraway firecracker will "sound" louder-- that's not a function of volume level, but of subjective sound texture. You can tell the difference between a loud sound from far away, and a soft sound (at the same volume) from nearby.

This is the critical fault with attempting to make "louder" records, as opposed to records that "sound" louder. Just pushing the record level ever closer to full-scale does not make popping bubble wrap sound more like firecracker. On the other hand, running that bubble-wrap popping through a high-pass filter, then a distortion effect, then a low-boost eq, then a muted delay, then a big, dry reverb might make it sound like a faraway cannon at any volume.

That's the difference between between making something "sound" louder versus trying to make it "be" louder. It's bad, sloppy, and dumb recording practice to try and control the listener's volume knob-- you can't do it. What you CAN control is the texture and the quality of the sound they hear, at any volume.

I can turn up a mumbly singer to the point of clipping, and they'll still sound like a mumbly singer, they'll just sound like a bad, clipped recording of a mumbly singer. Or I can take a mumbly singer, and (sometimes) find the articulation frequencies to boost, and the proximity-effect frequencies to cut, and the right amount of parallel saturation/compression and natural reverb to make that singer sound like they're howling and moaning.

That's totally different than trying to make a mumble sound like a strong singer by just turning up the record level until it's clipping, then limiting the output-- that only sounds like a singer mumbling into a too-loud microphone, which will be quickly turned down by the listener, until it sounds like a mumbly-but-clipped singer.

Quote:Originally Posted by MCV
Hi Yep.

You like compression, you like limiting, you don't like too much limiting.""
=====

That's not precisely what I said, and not necessarily something I agree with. I don't object to even very heavy compression or limiting.

Quote:What I'm saying is that some people ligitimately like their recordings to be mastered hotter than some other people - and they like it also by doing even-level comparisons.""
=====

Which is perfectly fine-- limit as much as you want, just don't use the makeup gain. That way you know you like the sound better, and not just the temporary volume increase.

If everyone followed that approach, there would be no debate, because 99% of the people who think they like "hot" masters (read "flattened" masters) would immediately realize that they don't, they only like the increased volume, which they could better get by turning up their speakers. And nobody would be arguing with the other 1%.

I personally disagree that it's about finding a "balance" between "loud" and "good-sounding", because the "loud" side of that see-saw is an illusion. So "balancing" the two just makes it less good in pursuit of an unattainable goal.

However you or anyone else are certainly free to disagree, and I don't really want to argue that topic any further in this thread unless there is some technical misunderstanding that I can help clarify. Moreover, I hope this is a useful thread to those who choose to try and make their records "louder" as well as those who just try to make them better. So on that note, I'll bow out of the opinions on the loudness race in this thread.

Cheers.

Quote:Originally Posted by thalweg
Hi Yep,

I understand the mechanics of compression based on your posts and use it often primarily to control peaks and smooth out the volumes. However I'd like to know more about the various approaches to compression to try to get more jump and texture into my work. I find that most of my stuff while nice and smooth and even sounding lacks the vibrancy and sense of thump if you know what I mean? I don't care about the loudness thing...I'm loud enough. Apologize if this has been addressed before.

Thanks""
=====

Much, much earlier in this thread I spoke at some length about the "gremlin" in the compressor, and how it works. It starts here:

http://forum.cockos.com/showthread.p...lin#post278807

I would encourage you to check there and to try and wrap your head around how a compressor really works, especially if you have gotten used to using one- or two-knob compressors that don't offer much detailed control.

Compression can be a very difficult tool to get started with for beginners, partly because it's difficult to hear at first, until you get a bit of practice, and also because every setting affects every other setting, sometimes in counter-intuitive ways. For example:

- I might tell you try lengthening your attack times to let more uncompressed transient through, but depending on where you've set the threshold and release, it could have almost the opposite effect, or it could simply negate the effect of the compressor entirely.

- I might tell you to shorten the release times to open up the sustain, but if the threshold is set very low, that might only end up boosting the hiss and noise between notes.

- I might tell you to ease off the ratio to get a less compressed, more dynamic sound, but that might negate the levelling consistency that prompted the compression in the first place, when something like attack/release adjustments would have restored more of the "bounce" and "vibrancy" without the disappearing/reappearing instrument effect.

And so on.

With compression, more so than with most effects, it's very hard to talk intelligently about settings in general terms. It used to be much easier to swap "recipes" in the analog days, because people were working on calibrated systems and you could generally assume that the VU meters were roughly bouncing around 0dBu average level.

But with digital recording, there is no way to tell what we're talking about. I could say that I usually start with the threshold around -12dB or whatever, meaning that I try to set the threshold right around the average level, but you might be working with a track that only has occasional PEAKS at -12. Which means that my settings will be meaningless for your track since the compressor won't even be activating.

Hopefully some of that older stuff is helpful. Please post back if there are specifics that I or others might be able to help with.

____

Quote:Originally Posted by Gizzmo0815
Yep I know you used some images and audio examples earlier in the thread, but I'm wondering if you could provide some examples of what you think is "good" compression vs. "bad" compression.""
=====

First, setting aside any "loudness wars" issues, IMO the primary benefits of compression are the ability to alter the dynamic "feel" of an instrument to better suit either the mix or the recording or sonic goals.

Listen to how the music changes in terms of how it breathes, pulses, and feels. Different compression settings can make the same track either feel hard and thumpy, smooth and spacious, pumpy and "Closed-in", scrunched up and telephoned, or smacky and slurpy.

Don't know what all those vivid but non-technical terms mean? That's okay, neither do I. And it doesn't matter, either. The point is for YOU to start listening for "slurp" or "fatness" or "pumping" or "smoothness" while you play with the compressor-- whatever YOU think those words should mean, or whatever words or concepts you want to look out for, good or bad.

Chances are, whatever characteristic you think your music could use more of, or less of, even if you cannot put a name to it, compression can often help you get there, because compression affects the very essence of the musician's "touch" and instrument response.

There are a million places to find compressor "recipes" in books and on the web. I don't have any to offer and I'm not very interested in them to begin with.

Instead, what I am suggesting is to forget about the technical details of this or that setting, and instead focus on the aesthetic goals you are trying to achieve. Understanding some of the theory, how the controls work and what they do, can help to make the experimentation process quicker and more productive, but what matters is not the knobs but the sound coming out of the speakers.

Quote:I think one of the most difficult things about the whole compression topic is that it's hard to hear (see) what effective compression looks and sounds like as opposed to compression that does damage to the audio. It's extremely hard to understand it without a solid point of reference for either extreme.""
=====

My suggestion is to stop trying to think through it, and instead, sit down with a compressor and a repetitive track of music, and try to create the most completely opposite-sounding versions of it you can with just a compressor. Start out by trying to make one version sound as big, flabby, and pumpy/sucky as you possibly can, and then make the other sound as tight, punchy, and focused as you can. Or whatever...once you get started, let the sounds you're getting guide you to extremes... you're not trying to make it sound "good", you're just trying to explore the limits of what you can achieve with just a compressor.

"Good" is way too hard to talk about, because anything I might say you should "do" with a compressor might already exist in one track or another. If the track needs more attack, then it's "good" to use the compressor to punch it up by tamping down on the body and sustain. But if the track already has too much attack, then it's "good" to use the compressor to control the attack so you can beef up the body of the sound.

Quote:...loudness war and the destruction that occurs to audio in this case. But what do you do about it? Do you release your album/song/project at levels that are simply lower than the others? This can have implications with the current methods that people used to listen to their music...

I have talked about this extensively elsewhere, and have semi-promised to lay off the subject in this thread, but my response to your question/statement is NO IT DOES NOT have any realistic/significant implications for how people will hear your music. The in-studio A/B comparisons that lead to "loudness war" compression and limiting have no application to the real world. It's a completely pointless race that you cannot win.

Radio stations, shopping-mall PAs, Club DJs, bar jukeboxes, TV broadcasts, and everything else that mixes your music with other content already level-matches everything, so you can't make your songs play back louder or softer, you can only make them flatter and more distorted. Someone out jogging with your song on their iPod has already purchased or pirated your song, so you cannot sell any more songs or win any more fans by blowing out their ears and trashing your sound quality. Moreover, the annoying songs when you're listening on shuffle are not the soft ones (turning up the volume when a good song comes on is half the fun of listening to music), but the loud ones that sound awful and inappropriate at any volume.

I cannot stop people from concocting hypothetical scorched-earth scenarios where being 6dB quieter than something else might lose a would-be fan or record buyer. If that outlying hypothetical is more important than the sound quality of your record, then do what you have to do. I won't argue the topic here, and I hope to be of some help whether you pursue that fan or all the others who actually care about good records.

Quote:Originally Posted by thalweg
Hopefully some of that older stuff is helpful. Please post back if there are specifics that I or others might be able to help with.

...How would you, using compression, radically drop the volume of a 5 second bass note to as low as possible for a 100ms duration somehwere in the middle of that 5 second note?...[/quote]""
=====

Okay, if you're going to go all mathy like that, then just use an automation envelope. You *might* achieve this precise result with the right compressor and the right settings, but this is getting a little nuts.

Quote:In other words, I want a significant initial attack and body of the note to come through uncompressed, then in the middle of the body of the note significantly reduce volume and then have the volume swell up again to the original signal....Does any of this make sense? Again its not for any musical purpose but to understand the extreme elements of compression. Sorry for the butchered articulation.""
=====

You keep changing what you're saying, which makes it hard to tell what you're talking about... First you said you wanted a "zero volume" period in the middle of the note, then you're saying "significantly reduced"... You're misunderstanding a lot of the controls, and I think you're thinking too hard. You can either take a deep breath, re-read, and really wrap your head around the controls and what they do, or you can just start messing around with the controls and see how they interact and sound. Either is a valid way to get started but this frantic meth-head approach of trying to make specific results happen by manipulating controls you don't really understand is a waste of time. And trying to work on five-second bass notes etc is pointless.

For starters, the "release" isn't going to do a damn thing until AFTER the signal drops below the threshold. So at your super-low threshold, the release is just going to wait until the note dies and THEN keep compressing the silence for some period of time.

Here's what's happening in your example: the note begins, and as soon as it crosses the -30 threshold (basically as soon as the note starts), the "gremlin" inside the compressor wakes up, and hits the snooze button and goes back to sleep for 100 ms, during which time the compressor is doing nothing (or almost nothing, just "rubbing it's eyes", so to speak). After 100ms, he gets out of bed, and basically starts riding the fader so that the level is constantly at -30 (infinite ratio does not mean he turns down the signal to minus infinity, it just means he keeps the signal infinitely close to the threshold of -30). Once the incoming signal drops BELOW -30 (the very tail end of the note), he starts watching the clock for quitting time, but for the next 400ms, he's still keeping the volume turned down. 400 ms after the note ends, he allows the silence to come back up to normal volume, and goes back to sleep.

Now, something thing that is important to distinguish here: not all compressors work exactly the same way. In fact the controls on a great many of the most highly-prized compressors are not really algorithmically precise. Things like the hardness of the knee, the "recovery curve" at the release, and especially the detection circuit or algorithm make a very big difference in how a compressor sounds. We're pretending that this is all perfectly linear for purposes of discussion, but it's usually not (probably never, but I like to hedge my bets).

For example, how the compressor decides what input level is matters a lot to the sound and response of the compressor. It should not be tracking individual samples or it will just fuck up the waveform. It's using some kind of "window" where it looks at the last 100ms or whatever to decide how much gain-reduction to apply. It may also be somewhat frequency-weighted (it is already, just based on the "window size"). And so on. If that doesn't make any sense to you, it doesn't have to right now, because unless you're designing your own compressor it doesn't matter.

Quote:Originally Posted by IIRs
...I explain the basics of compression in this video... don't worry that it features a FabFilter plug: the basic concepts apply to all compressors, and the graphical displays in Pro-C help to show what's really going on.""
=====

The above is a great video on this stuff (and a pretty slick compressor interface, too)

Quote:Originally Posted by MCV
Hi again IIRs,

I do understand what you say. I think it is irrelevant. And also, the way the attack of the compressor behaves will depend on the compressor. A "perfect" hard-knee compressor, as I understand it, should wait until the attack time has passed to do its thing. But maybe I'm wrong about that. Not that it matters really. I use compressors, I don't design them, and although I do know how a compressor works, in the day to day job I go by ear, as I believe most people do.

Cheers.""
=====

This debate is splitting hairs a little bit.

IIRS is technically correct that the attack curve (the delay set by the "attack" control between when the compressor first detects signal above threshold, and when it achieves full compression) is usually not a single, hard "step". In fact compression wouldn't work that way anyway-- unless we're talking about absolute and instantaneous brick-wall limiting (aka clipping) the detection element alone would necessarily introduce a certain amount of "gradualness" and time-delay.

That said, your understanding is correct in principle, although not necessarily an algorithmically precise description of what happens. The "attack" control does essentially allow a certain amount of sound to "poke through" before it gets compressed. How that attack curve works, when it kicks in, how steep it is, and how it relates to the detection circuit etc is a big part of what makes some compressors sound "punchy" or "smooth" or whatever.

Some compressors give you some degree of control over this, but usually it's a function of the designer, just like pressing the throttle on your car 30% of the way down does not necessarily translate precisely into a 30% increase in fuel and air in the combustion chamber. Your accelerator pedal is probably "tuned" to allow more precise speed control with wider throw at lower acceleration, and to become somewhat vaguer as you stomp all the way on the accelerator pedal.

These distinctions are not important for the end user. What does matter is whether the car handles well, accelerates predictably and intuitively, and so on. With compressors, what matters is whether the compressor is usable and suitable to the material. If your compressor muffles all the transients or makes them sound distorted and flat, or squishy and weird, then try another compressor. If you're test-driving a car and it lurches and halts without offering good control, test-drive some more cars.

A lot of compressor plugins these days tend to come with switches to select between "opto" or "VCA" response, or "vintage" buttons and so on, in an effort to approximate the different responses of various kinds of analog detection circuits. It's often hit-or-miss and kind of random whether they make any difference at all or sound better or worse on any given material, but the whole point behind those kinds of controls is to try and approximate the response curves of compressors that are thought to be "punchy" or "smooth" or "musical" or whatever.

For example, "optical" or "opto" detection circuits on analog compressors basically send the input signal through a light bulb or LED inside the compressor. A photocell opposite the lightbulb then "reads" the light level and compresses the signal based on the intensity of the light. Since light emitted does not precisely correspond to perceived loudness, and is somewhat nonlinear in relation to voltage, and since the light-sensor has its own imperfections, and since the lightbulb takes time to both "warm up" and "cool down", optical-based compressors tend to be somewhat "slower" and also nonlinear compared to VCA (or Voltage-Controlled Amplifier) compressors.

But people over-think this stuff. What matters is not whether the compressor has an optical circuit or a VCA or whatever (and certainly not whether it has a digital switch that pretends to be one or the other). What matters is how it sounds on the material you want to compress.

The first step is to learn how to use a compressor, and to learn to hear what compression sounds like... it's not nearly as obvious as effects like eq or distortion or reverb, but it can have a very big effect on how the listener "feels". Once you start to get a handle on the controls and how compression works, you will probably start to find that you like some better than others, just as a new piano player first has to figure out how to play the notes and chords, and then will start to find that some pianos sound more like what she means to sound like than others do.

Simple two-knob compressors such as blockfish or LA-2A can be helpful to beginners (and sometimes to old hands) in that they make all these decisions for you-- just decide how much compression you want, and turn up the knob to suit.

But sooner or later you will find a track that the two-knobber just mushes over by the time you get enough compression, or something like that, and then it will be time to get in there and start adjusting the attack and release curves.

In any case, it'snot all that useful nor important to know the ins and outs of every detail of the compression algorithm. You're not trying to precisely manipulate signal for scientific results, you're trying to make it sound good. A violin-player doesn't need to know the moisture content of catgut or the particle-density of wood or any of that, and it wouldn't matter if they did. Even with a doctorate in physical substances, they could tell a lot more about the quality of the violin by picking it up and playing it for 90 seconds than they could by running analytical tests on the material.

Quote:Originally Posted by carbon
I think only you can be the judge of that, if the amp sims fulfill their promise.
Just listen to people's recordings who use them.
In this thread you have even the FX chain to experiment with:
http://forum.cockos.com/showthread.php?t=66815

Is it good enough for you?
I think it is as good as it gets with sims.

And the best thing is to go to the studio and observe how people achieve their tone.

Your quest for tone seems to be largely theoretical, but the solutions are sometimes very specific - dependent on the song you've composed.""
=====

IMO, amp sims are usually not as good as a perfectly-tuned, perfectly-miked, genuine tube amp in a good room. But they are sometimes nearly as good, and often a lot better than the typical home-recordist is going to get on a typical home-recording budget.

Guitar sounds are one of my least favorite topics to discuss, for two reasons:

- First because they are both incredibly subjective and passionately-debated: one man's holy grail is another's fizzy trash.

- The second, and IMO more important reason, is that performance trumps everything else. A brilliant performance through a hissy, buzzy, flabby-sounding rig is worth 10X as much as a formulaic and rote performance through a primo rig. This holds true for all instruments, not just guitar, but becomes a more acute issue with electric guitar because the sound of the recording is much harder to divorce from the sound of the instrument.

To elaborate on the second: a brilliant cellist, or pianist, or vocalist (or even acoustic guitarist) can deliver a brilliant performance even if they are being recorded over a telephone line. The ultimate recording won't sound as good, but the "there" will still be there, so to speak: it will sound like a poor recording of great music, assuming the music was great to begin with.

But with electric guitar, the crude, soviet-era transduction system *IS* the sound of the instrument. If the player isn't "feeling" it, then the performance is going to reflect that. And guitar players can be weird about how they want things.

Electric guitar is not a midi input. Probably more so than any other instrument except vocals, ten different competent guitar players can play the exact same sheet music through the exact same rig, and sound totally different.

Some guitar players sound awesome through just about anything, even if they are playing some old rusted-string cheapo through a practice amp. Some other, equally-good (maybe better) guitar players have learned squeeze just the right nuance, expressiveness, and "voice" out of their particular rig, and can't get the right sound out of anything else, no matter how expensive. Those players might never deliver a decent performance through a (for them) awkward-responding rig, even if some other superstar uses the exact same setup to great effect.

When it comes to tracking (i.e. actually "recording") electric guitar, the most important thing is that the player is happy with the sound, and can get lost in it. It is very easy (and good practice) to split that sound with a Y-cable or dual-out DI box or un-embedded "in the box" amp emulation or similar, so that you can "re-amp" the sound later, if need be. The sound that the player loved when they were standing in front of 120dB half-stack might or might not be the ideal sound for the mix at 83dB SPL monitoring. What DOES matter is that the player had "his" sound (or "her" sound) while playing the part.

You can spend a million years reading internet posts arguing over whether Les Pauls or Strats or Plexis or Twins or JCMs or boutique amps are the best, or whether PODs are just as good, or better, or worse than the real amps or some other freeware emulator, and whatever. Some people will physically fight you for having the wrong opinion. It's a waste of time.

Different guitars and amplifiers DO sound different, and it's up to the player to care enough about her sound to find a rig that allows her to achieve what she wants to express.

Where it gets difficult is with mediocre players who can't/don't control the tone and timbre of their instrument sound, which is common among electric guitar players who are used to practicing "unplugged"-- that is, they practice the mechanics of the finger-movements without practicing the "sound". This is a particular risk with electric guitar (and bass) for the very simple reason that, unlike any other instrument, it is entirely possible to practice without hearing what your instrument really sounds like.

This can be a serious problem with "bedroom" guitar-players. Such players are probably well-advised to get used to emulators and playing with headphones and so on, because otherwise all their practice is like a singer who never sings, but merely hums softly in their head-- their "voice" is apt to show significant shortcomings once they actually open up.

Quote:Originally Posted by Marah Mag
A question....

Why (other than ideology) would you record guitars without taking a direct? Isn't that routine procedure at this point? Especially if recording with a DAW?""
=====

Well, to play devil's advocate to my own advice above, there *is* also merit in making decisions once and then printing them, if only to prevent mixdown from turning into a months-long agonizing over guitar amp settings...

Horses for courses, though. If you're playing and recording yourself, it's probably going to be hard to get a good handle on monitoring while also performing. You might find down the road that some aspect of your mic or setup didn't really get the quality you were hearing from the amp. So keeping a re-ampable DI is useful in that situation.

Quote:Reading that (at least between the lines), it seems to pretty much scream: Use some kind of sim!

Or am I missing something? I mean, lie about it in your liner notes, or in your interviews w the gearhead mags, but why make things more difficult than they need to be?

It makes me wonder.""
=====

That's not really what I meant to imply, but you're certainly free to read anything you like where nothing was written (i.e. "between the lines").

In a real studio I'd basically always use an amp, assuming a good and well-maintained one was available to record.

In a bedroom, a Marshall half-stack or whatever (even a great one) presents a lot of potential problems. First: can you even realistically set it loud enough to get the right sound for a 6-hour tracking session? Second: has the amp actually been maintained, are the tubes matched, working, and biased properly? Is the acoustical space suited to recording something like a guitar amp, or are you dealing with all kinds of room artifacts and rattling furniture and so on?

Most especially: whether playing live or re-amping, can you even hear what the sound is like through the monitors while the amp is working at volume? Or are you fling blind with placement etc? A big advantage to sims, especially in a one-room, one-person studio, is that what you hear is what you get.

On tip I've recently been trying when using amp sims is to eq the dry input before it hits the sim to remove ugliness. That is, load up your amp sim, mess around until it sounds pretty good, then insert an EQ in the chain BEFORE the sim, and with the sim still playing, try cutting extreme lows and highs, and notching peaky, gritty, fizzy or otherwise objectionable frequencies. It seems to help with stuff that digital processor doesn't seem to be handling quite right (you can also certainly EQ after the sim, as well, and/or fiddle with the tone controls).

And of course, alternatives abound for those who want to explore them: power soaks, iso cabinets, low-power "recording" amps, etc. But that way lies a lot of expensive experimentation that is a lot less convenient than something like PodFarm or even good freeware sims like Simulanalog. At that point I start to think about just booking a few hours at a studio to do it right in the first place.

Quote:
Originally Posted by reapercurious
now there are better freeware plugins to do almost everything than the best payware

plugins of 10 years ago, but i was so much better at mixing back then. now my mixes sound like ass. everything sounds so much better in general these days, but when i try to mix them together, it just sounds flat and boring. i guess this isnt always the case, but... anyone else?""
=====

You might want to check out this thread:

http://forum.cockos.com/showthread.php?t=68258

Even easier, and perfectly seriously, feel free to go back to whatever you were doing 10 years ago. Good sound does not get worse with technology.

Quote:
Originally Posted by Marah Mag
...I sometimes think sound quality is overrated. Certainly over-prioritized.""
=====

Signature-worthy

Stopped 12-25-2010 Thread # 1687