Subject: The nut we have to crack
Posted by chuck duffy on Mon, 23 Oct 2006 23:55:06 GMT
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DJ,

Listen I know you love messing with this stuff, but I think we need to focus on how to get the mixes we want out of an all native system.

It just doesn't make any sense to me to get onboard with another weird, proprietary dsp system. Creamware is as weird, oddball nad proprietary as it gets. Why bother with it? Why bother with UAD or anything else. It just doesn't make sense to me.

If we can't get decent mixes out of a native daw then something is wrong. Let's find the thing that's wrong, and make it right.

Chuck

Subject: Re: The nut we have to crack
Posted by animix on Tue, 24 Oct 2006 00:11:23 GMT
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OK Chuck,

I'll bite. I'll have a native system here irregardless of what audio card I'm using. I don't plan to sell of anything until I'm convinced that the Scope cards are the ticket for me. Right now I'm in the middle of recabling the studio, but it will be pretty simple for me to configure it so that I can just restore a Ghost image, reconfigure the digital connects and pop in the Magma PCI cards, and I'm back to the RME/UAD-1 rig that I've been running. I have always intended to keep this viable for a while. The only major change I'll be making is the mobo on both rigs. In the near future, both the Paris system and the Native system will be running on Gigabyte GA-K8NS Ultra 939 mobos, but in the meantime, what I've got now is working. Believe me, I would dearly love to get a native system sounding like Paris and I'll gladly help you beta whatever plugin you might develop or jump through some experimental hoops, and I'm sure Neil, Dedric, Gene, Dave, LaMont will have great suggestions since they are much further along in native world than I am........but I'm also going to be getting another wierd proprietary DAW happening......the stuff is already on the way here.

;o)

"chuck duffy" <c@c.com> wrote in message news:453d565a$1@linux...
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> Chuck

Subject: Re: The nut we have to crack
Posted by TC on Tue, 24 Oct 2006 00:24:06 GMT
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With all due respect Chuck, screw that noise, I'm all for weird oddball DSP systems..

It's worth it because it sounds good and is dead easy to mix in. And.. that makes it fun.

Besides, DJ's endeavors make for damn fine entertainment..

Go DJ.

Cheers,

TC

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Subject: Re: The nut we have to crack
Posted by j-cron on Tue, 24 Oct 2006 01:33:56 GMT
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YESSSSSSSSSS

Go Chuck!!!

oh wait I thought the thread was going to be about politics.

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Subject: Re: The nut we have to crack
Posted by Chris Ludwig on Tue, 24 Oct 2006 01:37:52 GMT
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HI DJ,
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1. It works with your chipset and motherboard. (It will probably work
better on your old system.)
2. You want to use it's internal effects and synths only.

3. It can run at very low latencies like the RME and Lynx cards can
under the same CPU load for all the of the Native VSTi and effects you have.

If it it doesn't do well for any of the above extremely well then there
is no benefit to it at all Personally I think purely native is the way
to go with the UADs being used during the mixing stages when latency
isn't as important.

FYI-
Here is a benchmark I just did today with the new single socket Quad
Core Intel CPU i.e.. 4 cpus on one chip. A 48k buffer on the the
Fireface is approx 2ms. With a Multiface I think the 64k buffer is 1.5ms.

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48k buffer @ 40% cpu load absolutely clean. At 64k it was 31%. We
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Chris Ludwig
ADK
chrisl@adkproaudio.com <mailto:chrisl@adkproaudio.com>
www.adkproaudio.com <http://www.adkproaudio.com/>
(859) 635-5762

Subject: Re: The nut we have to crack
Posted by Dedric Terry on Tue, 24 Oct 2006 01:45:38 GMT
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Imho, mixing has as much to do with how the engineer approaches the tools, and their past experience/expectations, as it does the tools, maybe more. That isn't a negative in any way - just than I believe perspective and past experience play a significant role in how one solves a "problem" (i.e. a mix itself).

Some people love analog summing like Dangerous 2-buss, others don't. Some get great mixes out of native DAWs, others don't. Some think Samplitude sounds better than Nuendo. Others might think Logic sounds better, or not. Same for DP, Sonar, Saw, ProTools, etc. It all just says there are more opinions than DAWs.

Anyway, Chuck - thanks for posting that finding in Paris. That answers some questions I've had over the years that never made sense under the assumed "mystique" of Paris. Actually this bodes quite well for anyone who loves to mix in Paris - taking a methodical approach to what this gain reduction/addition really means would lead one to uncover the keys to mixing in any native DAW.

Regards,
Dedric

On 10/23/06 5:55 PM, in article 453d565a$1@linux, "chuck duffy" <c@c.com> wrote:

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Subject: Re: The nut we have to crack
Posted by animix on Tue, 24 Oct 2006 02:01:53 GMT
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Chris,

The way I intend to use it is with it's internal FX only. If that doesn't float my boat, then I'm back to the native mix thing. I've been talking quite a bit to Ali Fawaz, the Creamware rep for North America. He's an interesting guy and very willing to be helpful in getting this going. He's very negative about VIA chipsets, but not for any particular reason relaing to driver compatibility. He says they are less efficient on PCI bus throughput. I know that was a problem years ago but nowadays I don't know if I buy that if a system is configured properly. I'll have a parallel rig running the Gigabyte GA-K8Ns Ultra 939 with an 4800 x 2 CPU. and I'll be testing both systems. I'm definitely on a mission here......stay with RME/UAD and mix native or move to Scope as a standalone DSP processor for Paris. I'm really comfortable mixing with UAD-1 plugins. It's going to take some serious stuff from Creamware to change my mind......but it's worth the effort to find out, IMO. I'm tired of chasing my tail.

Deej
"Chris Ludwig" <chrisl@adkproaudio.com> wrote in message
news:453d6c61@linux...
> HI DJ,
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> --
> Chris Ludwig
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> chrisl@adkproaudio.com <mailto:chrisl@adkproaudio.com>
> www.adkproaudio.com <http://www.adkproaudio.com/>
> (859) 635-5762

Subject: Re: The nut we have to crack
Posted by Neil on Tue, 24 Oct 2006 02:52:54 GMT
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"chuck duffy" <c@c.com> wrote:

>If we can't get decent mixes out of a native daw then something is wrong.
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(Long, but thought-provoking, and hopefully helpful, rant follows):

I think the thing that's wrong is that some people just can't get their heads around the differences between analog & digital. With analog, "big" = hotter, and so hotter is better. When you overbias your tape machines & smack the hell out of the tape, you're getting compression right off the bat on every track you do that with, so one gets used to hearing most tracks with some degree of tape compression already... and we all know that compression can make things sound "bigger". Or, you use a compressor on the way in to the tape so that you get a better SNR, but since that's not an issue with digital (unless you're recording at levels so low that you just simply get poor resolution, but that's a slightly different scenario), people quit using compressors on the way in to digital since SNR isn't an issue there.... you also can't smack an AD convertor hard & expect it to like it - unlike tape. So right off the bat we've got a whole different set of dynamics action going on from one world to the other - then, when you've already got that compressed kick or bassline on tape, you compress it more, and you're compressing an already-compressed signal, so when you apply compression to your uncompressed kick on your DAW you're thinking "nah, that CAN'T be right, it can't need THAT much compression! I'd better back that off a bit!" (because you're looking at the ratios & the threshhold, etc, instead of using your ears). EQ reacts differently with digital, too... if you're used to mixing on a console, you might be used to boosting or cutting something by 3, 4, 6db & getting an audible difference... with digital/plugin EQ's, sometimes you gotta boost or cut HUGE swaths of that frequency to really make a difference... why? I think it's a phase thing... you get more phase shift with analog filters, and so the change is more apparent at smaller degrees of boost & cut. That also helps to isolate things to have their own place in the mix at the same time... considering that phase is the reason we have two ears - it's the thing that makes it possible for us to tell which direction a sound is coming from - this makes perfect sense.

So, those of us (and I think that's "most of us here") who cut our teeth in the analog world first, and are used to all the things mentioned above - and who have not changed that style of mixing - could be disappointed in Native systems - not because they fall short of analog or Paris, but because they are actually much more accurate (assuming good quality convertors) & as a result do not impart certain types of coloration that we might interpret as "pleasing". If you could go back to a great mix you did on analog & a console & take out half of the amount of dynamics processing & half of the amount of EQ'ing you did, what would you get? A mix that sounded flatter & more colorless & with less dimension than the one you ended up with. Want proof? Here it is: If you didn't need the amount of EQ & dynamics you applied, you wouldn't have done so! If half the amounts/degrees of those things would have sufficed, that's what you would have used! So Paris sounds & acts kinda like analog, and people who like Paris like that aspect of it... how do we know there's not a few lines of code in there somewhere that adds graduated degrees of even-harmonic distortion when you push the faders or saturate the mix buss to whatever degree? I personally don't think it's strictly a DSP thing, because let's face it.. a plugin is basically doing the same thing to your mix whether it's running of a processor on it's own card or off your CPU; the difference being how well a particular VST or Direct-X compressor or reverb is written (and what it's designed to do in terms of treating the sound) vs. whatever DSP compressor or reverb plugin you're talking about. Can I get an "Amen, brutha!" on that?

Chuck's nailed the Paris mix buss thing, it seems, with that -22db at the channel & +22db at the mix buss, but WHY does that make a difference? Well, here's why gang... it's just as I said earlier in another thread - you've got to give yourself some headroom, dammit! Paris apparently does this for you. Want to prove me wrong? Open up a Paris mix and drag the mix buss master fader down 22db from wherever you have it, then insert any plugin that has an output level control on each individual channel of that mix - if the plugin is a compressor, for example, don't use any compression, just use the output control - now boost every channel by 22db using that output control... if it only goes up 10 db, then insert that plugin twice in a row & max out the output on each insertion... that'll be close enough... how's that sound? I'll bet it won't sound all that good! Are you hearing that "overstuffed" mix buss sound? Is it smaller, with less dimension? I'd be curious to see what you guys think if you try this. Now that we know what Chuck told us he discovered, this is the best way to see if that makes a difference or not (my guess - it DOES make a difference, otherwise, they wouldn't have written the code that way!).

So how can you get "big" in Native? Give yourself what Paris apparently already gives you... some headroom - think "clean", then dirty it up if you have to later... hell, just mash the mix with a comp & limiter or an L2 or something equivalent - you'll get all the harmonic distortion you want. I wasn't kidding the other day when I said: "Think zen when mixing in Cubase" it's all gotta flow without clips, gang... think about it... if you have one channel getting "overs" in a 32-bit float-point system, you may not notice it... heck you can't notice each sample in a given sound file can you? Of course not. But if you start adding more channels, and each of those channels is running hot... let's say 32 channels - as a comparison for you guys running two-card paris systems & no native mixes. and let's say you're running hot (over zero) about 25% of the time on each channel - that's 352,000 errors PER SECOND across the 32 tracks. That's a lot of floating-point math going on there, isn't it? And in this scenario, I want you to think of each error as a mistake, because that's what it is... in this style of mixing, it's a mistake. How can you expect something that's got 352,000 mistakes per second going on, to sound good?

Are you still not convinced? Then you should also definitely investigate running stems (submixes) & reimporting. When I've done this I definitely can hear a difference, and I suspect you most likely will be able to as well.. it is NOT a huge difference, but it's audible. In fact, some months ago I posted a stems mix vs. a non-stems mix & a number of you said you could hear a difference. Now, if you think "aww, this is just another pain-in-the-ass procedure I have to go through if I mix in Native", keep in mind that you can run 90 Million stems mixes in the time it will take Deej to set up his first Pulsar card, and another 900 million in the time that it takes Chuck to research & write that plugin (OK, just giving hell to Deej there, and no really no offense intended to Chucks coding capability, but I'm just saying this is something you can do RIGHT NOW, TONIGHT if you want to if you have a Native system, without having to wait for anything new). Now, if you have a small project - one acoustic guitar, piano, & a vocal - with just a few tracks, running stems won't make a difference, but if you have a large project, give it a shot... you may not hear enough of a difference to make it worth doing in any given instance, but then again, you might.

So, now that I hope I've made my case, here's my own personal guidelines for Native mixing - try it out & see wat you think:

1.) Do NOT bring down your Master Fader. It stays at zero (unless you're doing a fade).

2.) On your Master inserts, use a peakstop/brickwall limiter set anywhere from -.03 to -3db, depending on how much headroom you want to give your mastering engineer. Settings for volume maximization & other parameters will, of course, depend on the program material.

3.) Record at 24-bit 88.2k or higher (Dan Lavry has a white paper that makes a good case for a 60k sample rate - in order to get the ringing from the convertors' FIR filters out of the top range of our hearing - but since there is no standard 60k sample rate, 88.2 is the next one up). Also, 16-bit may have worked with Paris for whatever reason (maybe it just enhanced the harmonic distortion you're hearing?), but let's face it, everybody knows that more bits = greater "truth", especially when combined with higher resolutions.

4.) Default your individual channel settings to -6db or lower... I find that -6 is a good place to start because you can load up a decent amount of tracks without overloading the mix buss & hitting your limiter too hard at that level. Consider setting it lower as a starting point if you plan on getting into the range of 40+ tracks. HERE'S THE KEY... if you've got your mix roughed out & you can pull out that peakstop limiter I mentioned in #2 & NOT go over zero on the Master - you're golden. Fuck it, set 'em all at -15 as a starting point if you want, Paris is already setting them for you at -22, right? If you're getting a few scant overs without the limiter, you're still ok, really... the idea is not to overstuff the mix buss so heavily that if you pull the limiter off you're going into the +5, +6 range without it.

Think "clean" people = think "no clips" (or as few as possible), you get 30-40 channels of "overs" constantly (like the 352,000 of 'em per second in the example I gave earlier), and it's going to get harsh & thin.... it's a cumulative effect.

That's it, really... it's just like any other tool - you can't use an allen wrench to properly drive a nail, and you can't use a hammer to trim your nose hair.

Happy Native mixing!

(think "zen"!)

Neil

Subject: Re: The nut we have to crack
Posted by Neil on Tue, 24 Oct 2006 02:58:36 GMT
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Chris, I assume this is one of the systems that you built? If
so, how much $$$ are we talking about for that kind of rig?

Neil

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>Chris Ludwig
>ADK
>chrisl@adkproaudio.com <mailto:chrisl@adkproaudio.com>
>www.adkproaudio.com <http://www.adkproaudio.com/>
>(859) 635-5762

Subject: Re: The nut we have to crack
Posted by animix on Tue, 24 Oct 2006 03:56:06 GMT
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O wise one,

Which brickwall limiter preferest thou? What you described wasn't exactly
the methodology I used whil'st creating suckage from mine mix, so verily,
upon completing the recabling, performing a FengSui and reassembling my
environment in harmonic convergence, I shall endeavor once again to become

one with the native mix bus whil'st contemplating the aural posibilities of
yet another incarnation of the studio from hell.

Peacelovedove.........

;o)

"Neil" <IOUIU@OIU.com> wrote in message news:453d8006$1@linux...
>
> "chuck duffy" <c@c.com> wrote:
>
> >If we can't get decent mixes out of a native daw then something is wrong.
> > Let's find the thing that's wrong, and make it right.
>
> (Long, but thought-provoking, and hopefully helpful, rant
> follows):
>
> I think the thing that's wrong is that some people just can't
> get their heads around the differences between analog & digital.
> With analog, "big" = hotter, and so hotter is better. When you
> overbias your tape machines & smack the hell out of the tape,
> you're getting compression right off the bat on every track you
> do that with, so one gets used to hearing most tracks with some
> degree of tape compression already... and we all know that
> compression can make things sound "bigger". Or, you use a
> compressor on the way in to the tape so that you get a better
> SNR, but since that's not an issue with digital (unless you're
> recording at levels so low that you just simply get poor
> resolution, but that's a slightly different scenario), people
> quit using compressors on the way in to digital since SNR isn't
> an issue there.... you also can't smack an AD convertor hard &
> expect it to like it - unlike tape. So right off the bat we've
> got a whole different set of dynamics action going on from one
> world to the other - then, when you've already got that
> compressed kick or bassline on tape, you compress it more, and
> you're compressing an already-compressed signal, so when you
> apply compression to your uncompressed kick on your DAW you're
> thinking "nah, that CAN'T be right, it can't need THAT much
> compression! I'd better back that off a bit!" (because you're
> looking at the ratios & the threshhold, etc, instead of using
> your ears). EQ reacts differently with digital, too... if you're
> used to mixing on a console, you might be used to boosting or
> cutting something by 3, 4, 6db & getting an audible
> difference... with digital/plugin EQ's, sometimes you gotta
> boost or cut HUGE swaths of that frequency to really make a

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> phase shift with analog filters, and so the change is more
> apparent at smaller degrees of boost & cut. That also helps to
> isolate things to have their own place in the mix at the same
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> our teeth in the analog world first, and are used to all the
> things mentioned above - and who have not changed that style of
> mixing - could be disappointed in Native systems - not because
> they fall short of analog or Paris, but because they are
> actually much more accurate (assuming good quality convertors)
> & as a result do not impart certain types of coloration that we
> might interpret as "pleasing". If you could go back to a great
> mix you did on analog & a console & take out half of the amount
> of dynamics processing & half of the amount of EQ'ing you did,
> what would you get? A mix that sounded flatter & more colorless
> & with less dimension than the one you ended up with. Want
> proof? Here it is: If you didn't need the amount of EQ &
> dynamics you applied, you wouldn't have done so! If half the
> amounts/degrees of those things would have sufficed, that's
> what you would have used! So Paris sounds & acts kinda like
> analog, and people who like Paris like that aspect of it... how
> do we know there's not a few lines of code in there somewhere
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> you push the faders or saturate the mix buss to whatever
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> any plugin that has an output level control on each individual
> channel of that mix - if the plugin is a compressor, for
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> control - now boost every channel by 22db using that output

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> twice in a row & max out the output on each insertion...
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> apparently already gives you... some headroom - think "clean",
> then dirty it up if you have to later... hell, just mash the
> mix with a comp & limiter or an L2 or something equivalent -
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> kidding the other day when I said: "Think zen when mixing in
> Cubase" it's all gotta flow without clips, gang... think about
> it... if you have one channel getting "overs" in a 32-bit float-
> point system, you may not notice it... heck you can't notice
> each sample in a given sound file can you? Of course not. But
> if you start adding more channels, and each of those channels
> is running hot... let's say 32 channels - as a comparison
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> the 32 tracks. That's a lot of floating-point math going on
> there, isn't it? And in this scenario, I want you to think of
> each error as a mistake, because that's what it is... in this
> style of mixing, it's a mistake. How can you expect something
> that's got 352,000 mistakes per second going on, to sound good?
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> Are you still not convinced? Then you should also definitely
> investigate running stems (submixes) & reimporting. When I've
> done this I definitely can hear a difference, and I suspect you
> most likely will be able to as well.. it is NOT a huge
> difference, but it's audible. In fact, some months ago I posted
> a stems mix vs. a non-stems mix & a number of you said you
> could hear a difference. Now, if you think "aww, this is just
> another pain-in-the-ass procedure I have to go through if I mix
> in Native", keep in mind that you can run 90 Million stems
> mixes in the time it will take Deej to set up his first Pulsar
> card, and another 900 million in the time that it takes Chuck
> to research & write that plugin (OK, just giving hell to Deej
> there, and no really no offense intended to Chucks coding
> capability, but I'm just saying this is something you can do
> RIGHT NOW, TONIGHT if you want to if you have a Native system,

> without having to wait for anything new). Now, if you have a
> small project - one acoustic guitar, piano, & a vocal - with
> just a few tracks, running stems won't make a difference, but
> if you have a large project, give it a shot... you may not hear
> enough of a difference to make it worth doing in any given
> instance, but then again, you might.
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> So, now that I hope I've made my case, here's my own personal
> guidelines for Native mixing - try it out & see wat you think:
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> 1.) Do NOT bring down your Master Fader. It stays at zero
> (unless you're doing a fade).
>
> 2.) On your Master inserts, use a peakstop/brickwall limiter
> set anywhere from -.03 to -3db, depending on how much headroom
> you want to give your mastering engineer. Settings for volume
> maximization & other parameters will, of course, depend on the
> program material.
>
> 3.) Record at 24-bit 88.2k or higher (Dan Lavry has a white
> paper that makes a good case for a 60k sample rate - in order
> to get the ringing from the convertors' FIR filters out of the
> top range of our hearing - but since there is no standard 60k
> sample rate, 88.2 is the next one up). Also, 16-bit may have
> worked with Paris for whatever reason (maybe it just enhanced
> the harmonic distortion you're hearing?), but let's face it,
> everybody knows that more bits = greater "truth", especially
> when combined with higher resolutions.
>
> 4.) Default your individual channel settings to -6db or lower...
> I find that -6 is a good place to start because you can load up
> a decent amount of tracks without overloading the mix buss &
> hitting your limiter too hard at that level. Consider setting
> it lower as a starting point if you plan on getting into the
> range of 40+ tracks. HERE'S THE KEY... if you've got your mix
> roughed out & you can pull out that peakstop limiter I
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> golden. Fuck it, set 'em all at -15 as a starting point if you
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> you're getting a few scant overs without the limiter, you're
> still ok, really... the idea is not to overstuff the mix buss
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> Think "clean" people = think "no clips" (or as few as
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> and it's going to get harsh & thin.... it's a cumulative effect.

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> That's it, really... it's just like any other tool - you can't
> use an allen wrench to properly drive a nail, and you can't use
> a hammer to trim your nose hair.
>
> Happy Native mixing!
>
> (think "zen"!)
>
> Neil

Subject: Re: The nut we have to crack
Posted by Neil on Tue, 24 Oct 2006 04:19:28 GMT
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"DJ" <notachance@net.net> wrote:
>O wise one,
>
>Which brickwall limiter preferest thou?

I preferest this one, which is not only quite transparent, but
your wallet will find transparent, as well...

http://www.x-buz.com/BuzMaxi3.html

(actually I prefer version 2, which I don't think is available
anymore, but this one's nearly as transparent, methinks)

Neil

Subject: Re: The nut we have to crack
Posted by Dedric Terry on Tue, 24 Oct 2006 04:28:11 GMT
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I'm not Neil, and can't claim any resemblance to the title "O wise one" :-),
but give Spectraphy a try: http://www.crysonic.com/spectraphy.html.

Best I've found. ymmv, and any other applicable netiquette acronym.

Dedric

On 10/23/06 9:56 PM, in article 453d8fc2@linux, "DJ" <notachance@net.net>
wrote:

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>> (think "zen"!)
>>
>> Neil
>
>

Subject: Re: The nut we have to crack
Posted by LaMont on Tue, 24 Oct 2006 04:48:13 GMT
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Hi Chuck,
As I
ve stated many times before, there mixng in the Natives DAWs, expecially
Cuabse SX/Nuendo is difficult, until you add an mixer into the equation.
It's summing is poor at best.
Pro Tools is much better today for mixing and routing flexibility, even the
lower end versions (LE & M-Powered). You just hear and feel it when you import
audio(wav files) onto the playing field. They 48 bit summing is excellent,
and you can mix agressively..

Dedric and I have gone around and around on this issue. I'm sorry, I call
hen as I (Hear them).. :) This is the reason, I keep using Paris in my personal
studio. Now doay, other producers aring noticing that he Natives(Execpt PT)
are not living up to their hype..

Even more, I don;t agree with this new trend of adding more CPU powerer,
thinking that it will yield you better summing or sound.
It won't!!

Bottom line, if you are using SX/Nuendo, Sonar,Logic, you need to be using
the Apogee AD16x/DA..The Apogees have soft limiting beuilt in. Tha's very
inportant.. RME does not do soft limiting..
Or you need a decent analog /difital mixer to summ..Period..

"chuck duffy" <c@c.com> wrote:
>
>DJ,
>
>Listen I know you love messing with this stuff, but I think we need to focus
>on how to get the mixes we want out of an all native system.
>
>It just doesn't make any sense to me to get onboard with another weird,
proprietary
>dsp system. Creamware is as weird, oddball nad proprietary as it gets.

>Why bother with it? Why bother with UAD or anything else. It just doesn't
>make sense to me.
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>If we can't get decent mixes out of a native daw then something is wrong.
> Let's find the thing that's wrong, and make it right.
>
>Chuck

Subject: Re: The nut we have to crack
Posted by Kim on Tue, 24 Oct 2006 04:50:44 GMT
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Chuck,

There was talk some time ago (oh how the years wander on...) of somebody
making an EDS chip emulator, which would then allow various possibilities,
which one would assume would include:

1) a "Virtual" EDS card driver which emulates all the functionality of an
EDS card down to the last bit, and hence plugs right into Paris allowing
more submixes, natively, but with the same sound characteristics as the EDS
subs, or...
2) using the same technology, a virtual Paris mix bus, which uses the emaulation
of the EDS alongside the code from the Paris OS to basically allow a Paris
mix bus, using something like rewire, to plug in to a native app.

I believe the talk was inspired by Matthew Craig's efforts in creating the
VST Paris EQ, which does basically this same thing, emulating the EDS functionality
and hence generating pretty much identical output to the same audio going
through the card itself.

This would sure sort out the issues if anybody with enough knowhow and dedication
got on board. Suddenly any app could have the Paris mix bus, not to mention
the paris EQ...   that would pretty much put an end to all this shennigans
i would think.

Cheers,
Kim.

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>
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>on how to get the mixes we want out of an all native system.
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Subject: Re: The nut we have to crack
Posted by gene lennon on Tue, 24 Oct 2006 05:01:06 GMT
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Chris Ludwig <chrisl@adkproaudio.com> wrote:
>HI DJ,
> The Scope will do the trick if:
>1. It works with your chipset and motherboard. (It will probably work
>better on your old system.)
>2. You want to use it's internal effects and synths only.
>3. It can run at very low latencies like the RME and Lynx cards can
>under the same CPU load for all the of the Native VSTi and effects you have.
>
>If it it doesn't do well for any of the above extremely well then there

>is no benefit to it at all Personally I think purely native is the way
>to go with the UADs being used during the mixing stages when latency
>isn't as important.
>
>
>FYI-
>Here is a benchmark I just did today with the new single socket Quad
>Core Intel CPU i.e.. 4 cpus on one chip. A 48k buffer on the the
>Fireface is approx 2ms. With a Multiface I think the 64k buffer is 1.5ms.
>

>system-
>N3
>thonex test
>975 chipset mobo
>Quad core 2.66
>RME Fireface latest non-beta
>
>48k buffer @ 40% cpu load absolutely clean. At 64k it was 31%. We
>haven't seen any machine to date be able to play back this Thonex test
>at a 48k buffer totally clean.
>
>
>64k buffer is the lowest any systems have been able to go at these CPU
>loads.
>Core2 Duo 2.66 - 58%
>Dual core Opteron 2.6 - 58%
>AMD FX60 - 73%
>AMD X2 4400 - 70%
>Dual Core "Wood crest" Xeon 2.0 gig - 53%
>P4 955 3.4g - 76%
>

Chris.
Is that the chip that will turn my G5 quad into a G5X8?
Hint, hint.
Gene

Subject: Re: The nut we have to crack
Posted by Dimitrios on Tue, 24 Oct 2006 07:27:45 GMT
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As a 4 years Pulsar user I do not agree.
I have never seen anything like this no matter what I tried and looked for.
It is a mind blowing intergrated system that has support and a lot of third
party developers with free and commercial plugz.
I wholeheartly recommend it.
Regards,
Dimitrios

"chuck duffy" <c@c.com> wrote:
>
>DJ,
>
>Listen I know you love messing with this stuff, but I think we need to focus
>on how to get the mixes we want out of an all native system.
>
>It just doesn't make any sense to me to get onboard with another weird,

proprietary
>dsp system. Creamware is as weird, oddball nad proprietary as it gets.

>Why bother with it? Why bother with UAD or anything else. It just doesn't
>make sense to me.
>
>If we can't get decent mixes out of a native daw then something is wrong.
> Let's find the thing that's wrong, and make it right.
>
>Chuck

Subject: Re: The nut we have to crack
Posted by espresso on Tue, 24 Oct 2006 08:15:00 GMT
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I posted this a while ago.. I now use Nuendo with a Layla 3G to output 8 x analog channels and 8 x ADAT stems to Paris ie. Paris is now my mixer, its in the same computer, its easy....hardly any overhead as all Paris is doing is sitting in 'live' mode. I've been doing a bunch of live concert DVDs with 50 odd channels - 2 hour files - no chance I'd be wanting to convert all those puppies to .pafs... or even the stems for that matter. The proof is in the sound - the files played through Paris are alive and have depth. Same mix in Nuendo...urrrgghh. I know that I'm getting a double belt of DA-AD plus losing 4 bits of info through the Paris ADAT, but honestly the end justifies the means. All i'm trying to add to the discussion is - if you want the functionality of the native program plus the Paris sound its readily achievable without having to jump through the '2nd computer as FX buss' hoops.

Cheers,

David.

"Kim" <hiddensounds@hotmail.com> wrote in message news:453d9ba4$1@linux...
>
>
> Chuck,
>
> There was talk some time ago (oh how the years wander on...) of somebody
> making an EDS chip emulator, which would then allow various possibilities,
> which one would assume would include:
>
> 1) a "Virtual" EDS card driver which emulates all the functionality of an
> EDS card down to the last bit, and hence plugs right into Paris allowing
> more submixes, natively, but with the same sound characteristics as the
EDS

> subs, or...
> 2) using the same technology, a virtual Paris mix bus, which uses the
emaulation
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> VST Paris EQ, which does basically this same thing, emulating the EDS
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> and hence generating pretty much identical output to the same audio going
> through the card itself.
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> i would think.
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> Cheers,
> Kim.
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> >If we can't get decent mixes out of a native daw then something is wrong.
> > Let's find the thing that's wrong, and make it right.
> >
> >Chuck
>

Subject: Re: The nut we have to crack
Posted by Neil on Tue, 24 Oct 2006 13:02:51 GMT

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"LaMont" <jjdpro@ameritech.ne> wrote:

>Even more, I don;t agree with this new trend of adding more CPU powerer,
>thinking that it will yield you better summing or sound.
>It won't!!

Perhaps not, but what it WILL do is allow you to get into using very processor-intensive tools while still running a lot of tracks. For example, If you mix with Izotope Ozone across your 2-buss, you could mix with super high-quality program compression, brickwall limiting, and a stereo image enhancer right in place... however, Ozone is very CPU-intensive (I've done this before in SX, but I've had to "freeze" a lot of tracks in order to be abe to accommodate it. No big deal unless you want to make changes to any of those tracks inserts, then you have to unfreeze/tweak/refreeze.

Another thing it enables you to do is to get into convo or modeled reverbs on the same computer - i.e.: no having to dedicate a separate box for EFX. I have nothing against DSP-based stuff, but there are a couple of really cool - but SUPER-cpu-intensive verbs out there... did anyone else check out the demo for theRayspace reverb that Deej (i think it was Deej, anyway) posted a link for? If not, go check it out.... here's the link:

http://www.quikquak.com/software.html

Simply amazing, IMO; but very much a CPU hog. So much so that I can't use it at all - I was able to try it on a drum group only after disabling half of the tracks on a particular project.

Neil

Subject: Re: The nut we have to crack
Posted by Dedric Terry on Tue, 24 Oct 2006 13:37:38 GMT
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I don't think anyone believes cpu power has anything to do with sound quality, just as lowering channels by 22dB and raising the master buss by 22dB has nothing to do with improving sound quality - it's just a matter of managing levels for the user instead of the user lowering the fader by 22dB.

Nothing wrong with a faster machine to allow more fx processing... or FX Teleport, Wormhole or VST System Link.

Regards,
Dedric

On 10/24/06 7:02 AM, in article 453e0efb$1@linux, "Neil" <OIUOIU@OIU.com>
wrote:

>
> "LaMont" <jjdpro@ameritech.ne> wrote:
>
>> Even more, I don;t agree with this new trend of adding more CPU powerer,
>> thinking that it will yield you better summing or sound.
>> It won't!!
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> Perhaps not, but what it WILL do is allow you to get into
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> only after disabling half of the tracks on a particular project.
>
> Neil

Subject: Re: The nut we have to crack
Posted by John [1] on Tue, 24 Oct 2006 13:43:03 GMT
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Dedric, You have experience in both Cubase and Paris if I'm not

mistaken, I'm about to receive Cubase using presonus digimax fs pres.
Can you elaborate on what sonic differences I should expect to
experience pro and con?

Thanks,
John

Dedric Terry wrote:
> I don't think anyone believes cpu power has anything to do with sound
> quality, just as lowering channels by 22dB and raising the master buss by
> 22dB has nothing to do with improving sound quality - it's just a matter of
> managing levels for the user instead of the user lowering the fader by 22dB.
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> Nothing wrong with a faster machine to allow more fx processing... or FX
> Teleport, Wormhole or VST System Link.
>
> Regards,
> Dedric
>
> On 10/24/06 7:02 AM, in article 453e0efb$1@linux, "Neil" <OIUOIU@OIU.com>
> wrote:
>
>> "LaMont" <jjdpro@ameritech.ne> wrote:
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>>
>> Neil
>

Subject: Re: The nut we have to crack
Posted by LaMont on Tue, 24 Oct 2006 14:18:44 GMT
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My Point exactly.. If all of you who use Nuendo or Cubase cannot hear that there is something going on (software-wise) in Cubase or Nuendo that's not bringing "Full-life" to our wav files, then,I'm sorry, your ears are not as good as you may think..

"espresso" <audio@espressodigital.com> wrote:
>I posted this a while ago.. I now use Nuendo with a Layla 3G to output
8 x
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>Cheers,
>
>David.
>
>
>"Kim" <hiddensounds@hotmail.com> wrote in message news:453d9ba4$1@linux...
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>> Chuck,
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>> There was talk some time ago (oh how the years wander on...) of somebody

>> making an EDS chip emulator, which would then allow various possibilities,
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>> >DJ,
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>> > Let's find the thing that's wrong, and make it right.
>> >
>> >Chuck
>>
>
>

Subject: Re: The nut we have to crack
Posted by animix on Tue, 24 Oct 2006 14:29:15 GMT
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My wife agrees with you LaMont. She can pick out a native mix like a buzzard circling roadkill at 3000'.

"LaMont" <jjdpro@ameritech.net> wrote in message news:453e20c4$1@linux...
>
> My Point exactly.. If all of you who use Nuendo or Cubase cannot hear that
> there is something going on (software-wise) in Cubase or Nuendo that's not
> bringing "Full-life" to our wav files, then,I'm sorry, your ears are not
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> >> > Let's find the thing that's wrong, and make it right.
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> >> >Chuck
> >>
> >
> >
>

Subject: Re: The nut we have to crack
Posted by Neil on Tue, 24 Oct 2006 15:14:56 GMT
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"LaMont" <jjdpro@ameritech.net> wrote:
>
>My Point exactly.. If all of you who use Nuendo or Cubase cannot hear that
>there is something going on (software-wise) in Cubase or Nuendo that's not
>bringing "Full-life" to our wav files, then,I'm sorry, your ears are not
>as good as you may think..

There IS something going on... IME, I think that a lot of people
are using the tool in a manner in which it was not designed for.
It's not designed to accomodate 50 tracks worth of clips/overs
resulting in hundreds of thousands of errors per second... it's
as simple as that.
I don't think anyone who's said you can get good mixes out of
Native suystems has insisted that it sounds exactly like Paris
(or PT, or analog, or anything else), so is something different
going on? Yeah... it's different - doesn't mean that it can't be
good.

Subject: Re: The nut we have to crack
Posted by TCB on Tue, 24 Oct 2006 15:23:49 GMT
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Chuck,

By and large I agree with this, which is why I don't have a Pulsar card when
I could easily afford one. But I'll say this, you can't get any of John Bowen's
synths (www.zargmusic.com) on a native system, reverbs still throttle CPUs,
and Stephan Sprenger thought seriously about coding for Pulsar. He decided
not to _because_ it was so proprietary and becasue the vendor competed with
the third parties, but he liked the idea.

I remember many, many moons ago using another proprietary product that was
DOA on the market--the OASYS PCI card. Had they taken the time to put pretty
plug-in interfaces that could be used directly inside a VST/PT host I think
they would have sold a ton of them. I know it's stupid, but just being able
to lay off some of the really intense processors to a DSP chip, having a
decent hardware mixer for routing, and the extra goodies of a DSP card can
be really useful. And at $750 a Pulsar gets me all of that for cost of 5-8
good native plug-ins.

I think the reason UAD has done relatively well with their cards is just
this, they make things easier and they sound good. The Pulsar line _could_
be that if they do it right.

TCB

"chuck duffy" <c@c.com> wrote:
>
>DJ,
>
>Listen I know you love messing with this stuff, but I think we need to focus
>on how to get the mixes we want out of an all native system.
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>If we can't get decent mixes out of a native daw then something is wrong.
> Let's find the thing that's wrong, and make it right.
>
>Chuck

Subject: Re: The nut we have to crack
Posted by Chris Ludwig on Tue, 24 Oct 2006 16:24:17 GMT
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Hi Gene,
The chip we got is a pre-production unmarked one for testing. The ones you want are the quad core xeons that supposedly drop in to the G5s!! :)

Chris

Gene Lennon wrote:
> Chris Ludwig <chrisl@adkproaudio.com> wrote:
>> HI DJ,
>> The Scope will do the trick if:
>> 1. It works with your chipset and motherboard. (It will probably work
>> better on your old system.)
>> 2. You want to use it's internal effects and synths only.
>> 3. It can run at very low latencies like the RME and Lynx cards can
>> under the same CPU load for all the of the Native VSTi and effects you have.
>>
>> If it it doesn't do well for any of the above extremely well then there
>
>> is no benefit to it at all Personally I think purely native is the way
>> to go with the UADs being used during the mixing stages when latency
>> isn't as important.
>>
>>
>> FYI-
>> Here is a benchmark I just did today with the new single socket Quad
>> Core Intel CPU i.e.. 4 cpus on one chip. A 48k buffer on the the
>> Fireface is approx 2ms. With a Multiface I think the 64k buffer is 1.5ms.
>>
>> system-
>> N3
>> thonex test
>> 975 chipset mobo
>> Quad core 2.66
>> RME Fireface latest non-beta
>>
>> 48k buffer @ 40% cpu load absolutely clean. At 64k it was 31%. We
>> haven't seen any machine to date be able to play back this Thonex test
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>> 64k buffer is the lowest any systems have been able to go at these CPU
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>> P4 955 3.4g - 76%
>>
>
> Chris.
> Is that the chip that will turn my G5 quad into a G5X8?
> Hint, hint.
> Gene
>

--
Chris Ludwig

ADK Pro Audio
(859) 635-5762
www.adkproaudio.com
chrisl@adkproaudio.com

Subject: Re: The nut we have to crack
Posted by Chris Ludwig on Tue, 24 Oct 2006 16:48:40 GMT
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HI Neil,

Yes we built but no price yet sense the chips aren't in retail channels
yet.
it will less than a dual opteron 280.

Chris

Neil wrote:
> Chris, I assume this is one of the systems that you built? If
> so, how much $$$ are we talking about for that kind of rig?
>
> Neil
>
>
> Chris Ludwig <chrisl@adkproaudio.com> wrote:
>> HI DJ,
>> The Scope will do the trick if:
>> 1. It works with your chipset and motherboard. (It will probably work
>> better on your old system.)
>> 2. You want to use it's internal effects and synths only.

>> 3. It can run at very low latencies like the RME and Lynx cards can
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>> AMD FX60 - 73%
>> AMD X2 4400 - 70%
>> Dual Core "Wood crest" Xeon 2.0 gig - 53%
>> P4 955 3.4g - 76%
>>
>> DJ wrote:
>>
>>> OK Chuck,
>>>
>>> I'll bite. I'll have a native system here irregardless of what audio card
>>> I'm using. I don't plan to sell of anything until I'm convinced that the
>>> Scope cards are the ticket for me. Right now I'm in the middle of recabling
>>> the studio, but it will be pretty simple for me to configure it so that
> I
>>> can just restore a Ghost image, reconfigure the digital connects and pop
> in
>>> the Magma PCI cards, and I'm back to the RME/UAD-1 rig that I've been

>>> running. I have always intended to keep this viable for a while. The only
>>> major change I'll be making is the mobo on both rigs. In the near future,
>>> both the Paris system and the Native system will be running on Gigabyte
>>> GA-K8NS Ultra 939 mobos, but in the meantime, what I've got now is working.
>>> Believe me, I would dearly love to get a native system sounding like Paris
>>> and I'll gladly help you beta whatever plugin you might develop or jump
>>> through some experimental hoops, and I'm sure Neil, Dedric, Gene, Dave,
>>> LaMont will have great suggestions since they are much further along in
>>> native world than I am........but I'm also going to be getting another
> wierd
>>> proprietary DAW happening......the stuff is already on the way here.
>>>
>>> ;o)
>>>
>>> "chuck duffy" <c@c.com> wrote in message news:453d565a$1@linux...
>>>
>>>
>>>> DJ,
>>>>
>>>> Listen I know you love messing with this stuff, but I think we need to
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>>>
>>>
>> --
>> Chris Ludwig
>> ADK
>> chrisl@adkproaudio.com <mailto:chrisl@adkproaudio.com>
>> www.adkproaudio.com <http://www.adkproaudio.com/>
>> (859) 635-5762
>

--
Chris Ludwig

ADK Pro Audio
(859) 635-5762
www.adkproaudio.com
chrisl@adkproaudio.com

Subject: Re: The nut we have to crack
Posted by TCB on Tue, 24 Oct 2006 17:13:40 GMT
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I've played the same gear forever, including an 18 watt amp that is a cross
between as AC-30 and an AC-15. I put a THD power soak on so I can play it
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I tried to correct it by playing harder. Does that mean that beautifully
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Your wife didn't like the first non-PARIS mix you've done in years. Does
that mean native sounds bad?

Discuss.

"DJ" <notachance@net.net> wrote:
>My wife agrees with you LaMont. She can pick out a native mix like a buzzard
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Subject: Re: The nut we have to crack
Posted by John [1] on Tue, 24 Oct 2006 17:44:21 GMT
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If DJ's wife is not happy, nobody's happy !   hehe
John

TCB wrote:
> I've played the same gear forever, including an 18 watt amp that is a cross
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Subject: Re: The nut we have to crack
Posted by animix on Tue, 24 Oct 2006 18:40:22 GMT
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"TCB" <nobody@ishere.com> wrote in message news:453e49c4$1@linux...
>
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> restored AC-30s sound bad?
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> that mean native sounds bad?
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> Discuss.

I'd say you're absolutely right about this Thad......I should have said that she can pick out *my native mix* .....(add aforementioned reference to soaring carrion bird and carcass located on vehicular accessway). I do intend to keep plugging away at this. The Creamware system has always intrigued me.......going wayyyyy back. I printed out the manual and started reading last night. Even with the voluminous verbage vis-a-vis the RME Totalmix, most of it made sense to me immediately...........but then again, I think Windows in much more intuitive than Mac OS, so ....oh well.

;o)

Subject: Re: The nut we have to crack
Posted by animix on Tue, 24 Oct 2006 18:41:48 GMT
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Whaaaaatttttt???.......why???......what's the problem?

"rick" <parnell68@hotmail.com> wrote in message
news:ponsj2tvrbuutjnnsfh2nda9io1s3n6d1a@4ax.com...
> i hope to god you don't use that as a compliment when you tell amy how
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Posted by rick on Tue, 24 Oct 2006 18:46:06 GMT
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Subject: Re: The nut we have to crack
Posted by LaMont on Tue, 24 Oct 2006 19:44:36 GMT
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Neil I do mix follow the native mix rules. No overs, faders around -5db ect,
and I can make it sound good..

However, when I add in a mixer for summing, all of those native mixing rules
are out the window. The whole mix "sonically" opens up..

As well as, If I'm using Apogees AD16x/DA16x with soft-limiter set on, I
can mix like I want to in SX. With RME interface's and converters, I have
to abide by the rules.

Lastly, when i have to mix (In the Box) using SX/Nuendo, I refer to the Charles
dye method and add in Harmonic distortion via plugs in (namely) antares Mic
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These days,I just use the SSL plugs which have that harmonic distort color
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>There IS something going on... IME, I think that a lot of people
>are using the tool in a manner in which it was not designed for.
>It's not designed to accomodate 50 tracks worth of clips/overs
>resulting in hundreds of thousands of errors per second... it's
>as simple as that.
>I don't think anyone who's said you can get good mixes out of
>Native suystems has insisted that it sounds exactly like Paris
>(or PT, or analog, or anything else), so is something different
>going on? Yeah... it's different - doesn't mean that it can't be
>good.

Subject: Re: The nut we have to crack
Posted by Dedric Terry on Wed, 25 Oct 2006 01:35:52 GMT
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Lamont - if your D-A converters affect the way you mix inside a DAW, you
aren't mixing what you think you are. Certainly converters can sound
different, but the differences at the RME/Apogee level aren't in siginficant

areas (mainly a slight difference in sound of the top end - yes I've heard
all of these, along with Myteks, Cranesong, DCS, and others side by side -
Cranesong is my favorite - Myteks are great, but a little sterile. DCS is
just too expensive).

1) If you are saying you mix differently on a console because you are using Apogee converters from the DAW with soft limit vs. RME converters, also into the same desk (no mixing in SX, just playback), you are simply using the converters to color the signal (albeit only slightly), in different ways - Softlimit just limiting. Nothing wrong with that, but that is altering the tracks going in, not the mixing platform itself. Saying RME converters limit you because they don't have a limiter built in says you aren't mixing the way most of us do - you are trying to get analog saturation out of digital - ain't gonna happen.

2) If you are mixing inside SX and change your approach depending on which converter you monitor through, then that's a problem since your bounces aren't going to be the same, and your decisions aren't going to be consistent. The idea is to have your monitoring chain *not* affect your mix decisions, but enable more accurate ones.

If you mixdown to a 2-track of some sort (Masterlink, etc), then you are using SoftLimit as a limiter on the output. You could achieve the same think in a multitude of different ways.

Regards,
Dedric

On 10/24/06 1:44 PM, in article 453e6d24$1@linux, "LaMont"
<jjdpro@ameritech.net> wrote:

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>

Subject: Re: The nut we have to crack
Posted by Dedric Terry on Wed, 25 Oct 2006 01:45:26 GMT
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Hi John,

The difference is pretty much what Chuck found in technical detail - Paris
lets you clip channels by cutting your levels for you. Native DAWs don't do
this - what you put in you get out. You may not actually hear any
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There will always be people disagreeing about the audio quality of DAWs, but
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engineering logic.

There are a lot of people mixing pro level music in Nuendo, and Cubase is the same audio engine. Imho, Cubase 4 has the best workflow of any native DAW out there now.

Regards,
Dedric

On 10/24/06 7:43 AM, in article 453e165b$1@linux, "John" <no@no.com> wrote:

> Dedric, You have experience in both Cubase and Paris if I'm not
> mistaken, I'm about to receive Cubase using presonus digimax fs pres.
> Can you elaborate on what sonic differences I should expect to
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>>
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>>> cpu-intensive verbs out there... did anyone else check out the
>>> demo for theRayspace reverb that Deej (i think it was Deej,
>>> anyway) posted a link for? If not, go check it out.... here's
>>> the link:
>>>
>>> http://www.quikquak.com/software.html
>>>
>>> Simply amazing, IMO; but very much a CPU hog. So much so that I
>>> can't use it at all - I was able to try it on a drum group
>>> only after disabling half of the tracks on a particular project.
>>>
>>> Neil
>>

Subject: Re: The nut we have to crack
Posted by LaMont on Wed, 25 Oct 2006 01:50:19 GMT
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Yes, I do mixdown to a Masterlink...

Dedric Terry <dterry@keyofd.net> wrote:
>Lamont - if your D-A converters affect the way you mix inside a DAW, you
>aren't mixing what you think you are. Certainly converters can sound
>different, but the differences at the RME/Apogee level aren't in siginficant
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>
>If you mixdown to a 2-track of some sort (Masterlink, etc), then you are
>using SoftLimit as a limiter on the output. You could achieve the same
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>
>Regards,
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>
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>
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 ect,
>> and I can make it sound good..
>>
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>>
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I
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Mic
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>

Subject: Re: The nut we have to crack
Posted by gene lennon on Wed, 25 Oct 2006 05:09:00 GMT
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Chris Ludwig <chrisl@adkproaudio.com> wrote:
>Hi Gene,
>The chip we got is a pre-production unmarked one for testing. The ones
>you want are the quad core xeons that supposedly drop in to the G5s!! :)
>
>Chris
>

Gene

Subject: Re: The nut we have to crack
Posted by rick on Wed, 25 Oct 2006 09:09:43 GMT
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man, your amy is one loving woman.

On Tue, 24 Oct 2006 12:41:48 -0600, "DJ" <notachance@net.net> wrote:

>Whaaaaatttttt???.......why???......what's the problem?
>
>
>"rick" <parnell68@hotmail.com> wrote in message
>news:ponsj2tvrbuutjnnsfh2nda9io1s3n6d1a@4ax.com...

>> i hope to god you don't use that as a compliment when you tell amy how
>> much you appreciate her senses...whew...
>>
>> On Tue, 24 Oct 2006 08:29:15 -0600, "DJ" <notachance@net.net> wrote:
>>
>> >My wife agrees with you LaMont. She can pick out a native mix like a
>buzzard
>> >circling roadkill at 3000'.
>> >
>> >
>> >"LaMont" <jjdpro@ameritech.net> wrote in message news:453e20c4$1@linux...
>> >>
>> >> My Point exactly.. If all of you who use Nuendo or Cubase cannot hear
>that
>> >> there is something going on (software-wise) in Cubase or Nuendo that's
>not
>> >> bringing "Full-life" to our wav files, then,I'm sorry, your ears are
>not
>> >> as good as you may think..
>> >>
>> >>
>> >> "espresso" <audio@espressodigital.com> wrote:
>> >> >I posted this a while ago.. I now use Nuendo with a Layla 3G to
>output
>> >> 8 x
>> >> >analog channels and 8 x ADAT stems to Paris ie. Paris is now my mixer,
>> >its
>> >> >in the same computer, its easy....hardly any overhead as all Paris is
>> >doing
>> >> >is sitting in 'live' mode. I've been doing a bunch of live concert
>DVDs
>> >> with
>> >> >50 odd channels - 2 hour files - no chance I'd be wanting to convert
>all
>> >> >those puppies to .pafs... or even the stems for that matter. The proof
>is
>> >> in
>> >> >the sound - the files played through Paris are alive and have depth.
>Same
>> >> >mix in Nuendo...urrrgghh. I know that I'm getting a double belt of
>DA-AD
>> >> >plus losing 4 bits of info through the Paris ADAT, but honestly the
>end
>> >> >justifies the means. All i'm trying to add to the discussion is - if
>you
>> >> >want the functionality of the native program plus the Paris sound its
>> >> >readily achievable without having to jump through the '2nd computer as
>FX

>> >> >buss' hoops.
>> >> >
>> >> >Cheers,
>> >> >
>> >> >David.
>> >> >
>> >> >
>> >> >"Kim" <hiddensounds@hotmail.com> wrote in message
>> >news:453d9ba4$1@linux...
>> >> >>
>> >> >>
>> >> >> Chuck,
>> >> >>
>> >> >> There was talk some time ago (oh how the years wander on...) of
>> >somebody
>> >> >> making an EDS chip emulator, which would then allow various
>> >possibilities,
>> >> >> which one would assume would include:
>> >> >>
>> >> >> 1) a "Virtual" EDS card driver which emulates all the functionality
>of
>> >> an
>> >> >> EDS card down to the last bit, and hence plugs right into Paris
>> >allowing
>> >> >> more submixes, natively, but with the same sound characteristics as
>the
>> >> >EDS
>> >> >> subs, or...
>> >> >> 2) using the same technology, a virtual Paris mix bus, which uses
>the
>> >> >emaulation
>> >> >> of the EDS alongside the code from the Paris OS to basically allow a
>> >Paris
>> >> >> mix bus, using something like rewire, to plug in to a native app.
>> >> >>
>> >> >> I believe the talk was inspired by Matthew Craig's efforts in
>creating
>> >> the
>> >> >> VST Paris EQ, which does basically this same thing, emulating the
>EDS
>> >> >functionality
>> >> >> and hence generating pretty much identical output to the same audio
>> >going
>> >> >> through the card itself.
>> >> >>
>> >> >> This would sure sort out the issues if anybody with enough knowhow
>and
>> >> >dedication

>> >> >> got on board. Suddenly any app could have the Paris mix bus, not to
>> >> >mention
>> >> >> the paris EQ...   that would pretty much put an end to all this
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>> >> >> i would think.
>> >> >>
>> >> >> Cheers,
>> >> >> Kim.
>> >> >>
>> >> >> "chuck duffy" <c@c.com> wrote:
>> >> >> >
>> >> >> >DJ,
>> >> >> >
>> >> >> >Listen I know you love messing with this stuff, but I think we need
>to
>> >> >focus
>> >> >> >on how to get the mixes we want out of an all native system.
>> >> >> >
>> >> >> >It just doesn't make any sense to me to get onboard with another
>> >weird,
>> >> >> proprietary
>> >> >> >dsp system. Creamware is as weird, oddball nad proprietary as it
>> >gets.
>> >> >>
>> >> >> >Why bother with it? Why bother with UAD or anything else. It just
>> >> >doesn't
>> >> >> >make sense to me.
>> >> >> >
>> >> >> >If we can't get decent mixes out of a native daw then something is
>> >wrong.
>> >> >> > Let's find the thing that's wrong, and make it right.
>> >> >> >
>> >> >> >Chuck
>> >> >>
>> >> >
>> >> >
>> >>
>> >
>>
>

Subject: Re: The nut we have to crack
Posted by Chris Ludwig on Wed, 25 Oct 2006 13:31:26 GMT
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exactly!!

Dedric Terry wrote:

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>>>>
>>>>Neil
>>>>
>>>>
>
>
>

--
Chris Ludwig
ADK
chrisl@adkproaudio.com <mailto:chrisl@adkproaudio.com>
www.adkproaudio.com <http://www.adkproaudio.com/>
(859) 635-5762

Subject: Re: The nut we have to crack
Posted by LaMont on Wed, 25 Oct 2006 14:40:28 GMT
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Dedric,
My point has moe to do with 'head-room' of ITB mixes versus, using a analog
or digital mixer for summing.

There is a difference. Also, I challege anyone to open up say SX, DP, Logic
and play a stereo way file @ unity gain ..then, If you have copy of say Pr-Tools
LE M-powered, import that same file.. Then listen.
You can here the difference, even using the same audio interface..

I agree with you that you have to mix differently using the natives, but
Soft ware has a sound.. To me and others, to get make SX/Nuendo slam at it's
best, is to using a outboard summing mixer.

These days, my work flow is to record,edit,then bounce stems from Nuendo.
Simply put, there is no better workflow DAW on the planet for such tasks.
Then, I either mix in Pro-Tools or Paris depending on the color I'm going
for.

Dedric Terry <dterry@keyofd.net> wrote:
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>>
>

Subject: Re: The nut we have to crack
Posted by Tony Benson on Wed, 25 Oct 2006 16:38:02 GMT
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Neil,

Thanks for posting this. Last night I recorded a test song, drums, guitar, and bass in DP. I dropped the individual channel faders to -6.0 and added a limiter (for "make-up" gain and almost no limiting) to the DP main out. I didn't raise any channel fader above -6.0. Only lowered channels to balance levels. Even though I only had about 20 channels going, I could already tell it was one of the better sounding mixes I've been able to get out of DP. Can it really be this simple? I was so used to maximizing the levels in PARIS that I took that methodology over to DP and my mixes in DP always sounded "smaller". Now I'm jazzed about doing some more experimentation in DP. Thanks again.

Tony

"Neil" <IOUIU@OIU.com> wrote in message news:453d8006$1@linux...
>
> "chuck duffy" <c@c.com> wrote:
>
>>If we can't get decent mixes out of a native daw then something is wrong.
>> Let's find the thing that's wrong, and make it right.
>
> (Long, but thought-provoking, and hopefully helpful, rant
> follows):
>
> I think the thing that's wrong is that some people just can't
> get their heads around the differences between analog & digital.
> With analog, "big" = hotter, and so hotter is better. When you
> overbias your tape machines & smack the hell out of the tape,
> you're getting compression right off the bat on every track you
> do that with, so one gets used to hearing most tracks with some
> degree of tape compression already... and we all know that
> compression can make things sound "bigger". Or, you use a
> compressor on the way in to the tape so that you get a better
> SNR, but since that's not an issue with digital (unless you're
> recording at levels so low that you just simply get poor
> resolution, but that's a slightly different scenario), people
> quit using compressors on the way in to digital since SNR isn't
> an issue there.... you also can't smack an AD convertor hard &
> expect it to like it - unlike tape. So right off the bat we've
> got a whole different set of dynamics action going on from one
> world to the other - then, when you've already got that
> compressed kick or bassline on tape, you compress it more, and
> you're compressing an already-compressed signal, so when you
> apply compression to your uncompressed kick on your DAW you're
> thinking "nah, that CAN'T be right, it can't need THAT much
> compression! I'd better back that off a bit!" (because you're
> looking at the ratios & the threshhold, etc, instead of using
> your ears). EQ reacts differently with digital, too... if you're
> used to mixing on a console, you might be used to boosting or
> cutting something by 3, 4, 6db & getting an audible
> difference... with digital/plugin EQ's, sometimes you gotta
> boost or cut HUGE swaths of that frequency to really make a
> difference... why? I think it's a phase thing... you get more
> phase shift with analog filters, and so the change is more
> apparent at smaller degrees of boost & cut. That also helps to
> isolate things to have their own place in the mix at the same
> time... considering that phase is the reason we have two ears -
> it's the thing that makes it possible for us to tell which
> direction a sound is coming from - this makes perfect sense.
>
> So, those of us (and I think that's "most of us here") who cut
> our teeth in the analog world first, and are used to all the

> things mentioned above - and who have not changed that style of
> mixing - could be disappointed in Native systems - not because
> they fall short of analog or Paris, but because they are
> actually much more accurate (assuming good quality convertors)
> & as a result do not impart certain types of coloration that we
> might interpret as "pleasing". If you could go back to a great
> mix you did on analog & a console & take out half of the amount
> of dynamics processing & half of the amount of EQ'ing you did,
> what would you get? A mix that sounded flatter & more colorless
> & with less dimension than the one you ended up with. Want
> proof? Here it is: If you didn't need the amount of EQ &
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Posted by Mic Cross on Wed, 25 Oct 2006 19:59:13 GMT
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Subject: Re: The nut we have to crack
Posted by volthause on Wed, 25 Oct 2006 20:23:46 GMT
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It shouldn't no. We're talking about mix stage, not recording stage. Recording into 16 bit resolution you would want to take as much advantage of amplitude, thus bit resolution. But when mixing you want to avoid overloading the 2 buss master. Lowering your individual channel faders will help you avoid that.

-scott v.

Subject: Re: The nut we have to crack
Posted by Tony Benson on Wed, 25 Oct 2006 20:44:24 GMT
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I assume it does Mic, but going by ears, things sounded good. I guess I don't know for sure how lowering the fader level affects the bit depth in DP. It is a question I was wondering about also.

Tony

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Subject: Re: The nut we have to crack
Posted by Mic Cross on Wed, 25 Oct 2006 21:31:26 GMT
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Subject: Re: The nut we have to crack
Posted by Tony Benson on Wed, 25 Oct 2006 21:53:48 GMT
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That's what I understand, but I'm not a tech geek (no offense to the tech
geeks of course) on how different DAW's handle the math involved in changing
gain at the per track (channel) level. Maybe since the math involved is
handled at a higher level (32 bit floating? whatever Integer?) the actual
bit reduction isn't an issue.

I can say that I didn't notice anything strange going on with my little test
recording as far as "graininess" or anything else I would call "low bit"
sounding. It was actually the opposite. I was able to hear more separation

bit I could hear more space around each track. It was much easier to get
things to "sit right" in the mix. I'm going to try this on some higher track
counts and see if it still holds true.

Tony

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>>>>> the harmonic distortion you're hearing?), but let's face it,
>>>>> everybody knows that more bits = greater "truth", especially
>>>>> when combined with higher resolutions.
>>>>>
>>>>> 4.) Default your individual channel settings to -6db or lower...
>>>>> I find that -6 is a good place to start because you can load up
>>>>> a decent amount of tracks without overloading the mix buss &
>>>>> hitting your limiter too hard at that level. Consider setting
>>>>> it lower as a starting point if you plan on getting into the
>>>>> range of 40+ tracks. HERE'S THE KEY... if you've got your mix
>>>>> roughed out & you can pull out that peakstop limiter I
>>>>> mentioned in #2 & NOT go over zero on the Master - you're
>>>>> golden. Fuck it, set 'em all at -15 as a starting point if you
>>>>> want, Paris is already setting them for you at -22, right? If
>>>>> you're getting a few scant overs without the limiter, you're
>>>>> still ok, really... the idea is not to overstuff the mix buss
>>>>> so heavily that if you pull the limiter off you're going into
>>>>> the +5, +6 range without it.
>>>>>
>>>>> Think "clean" people = think "no clips" (or as few as
>>>>> possible), you get 30-40 channels of "overs" constantly (like
>>>>> the 352,000 of 'em per second in the example I gave earlier),
>>>>> and it's going to get harsh & thin.... it's a cumulative effect.
>>>>>
>>>>> That's it, really... it's just like any other tool - you can't
>>>>> use an allen wrench to properly drive a nail, and you can't use
>>>>> a hammer to trim your nose hair.
>>>>>
>>>>> Happy Native mixing!
>>>>>
>>>>> (think "zen"!)
>>>>>
>>>>> Neil
>>>>
>>>>
>>>
>>
>>

>

Subject: Re: The nut we have to crack
Posted by Dedric Terry on Thu, 26 Oct 2006 14:39:03 GMT
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Actually the 3.5 bit loss assumes either of two situations:

1) I'm guessing Chuck was referring to the EDS code, so if the gain reduction happens (for some unknown reason) after reading the file off of disk, and before pushing it into the higher bit depth processing section, then it would pad 0's for any extra bits beyond 24.

2) More likely, if you reduce *all* of your tracks by 22dB, sum them, reduce the master fader (as many might), then you could effectively have some tracks lose their original lower bits simply because that is all pushed down below 23-bits in the sum before sent back out as a 24-bit stream.

Next assumption - we can actually hear -122dB. :-) That's where this is happening.

Really this isn't a big deal - what bothers me about the concept of every track being reduced by 22dB without the express written consent of the engineer/mixer is that it is misleading and presupposing you need to reduce track gain to get the mix to work.

In any given large mix, I may actually end up with many tracks down by 15-25dB in order to keep the master in the right range, but it's easier to make that choice based on what the song, the tracks and the mix need. For sure it seems to work in Paris to some degree. But if you start knowing how your mix should sound (hearing it mentally), and how each track should fit into that, it's easy to create that sonic space with most any mixing medium. That's where the argument about one DAW mixing better than another falls down for me - it says the engineer is letting the medium dictate the mix rather than the engineer. That isn't engineering.

Regardless of what you mix in, there is only 40Hz to about 17kHz of actual human listening/hearing range in the final product, and only 0dBFS of max level, and -96dB of min level. That's the space we have to work with, and only so much can fit in there. DAWs don't prevent music from fitting in that comparatively small range, people do.

The point really is that this technical discovery about Paris says one and only one thing:

* When you mix digitally, control the levels of your tracks to fit the mix rather than assuming you can just push up faders and have each track find

it's own space automatically *

Just my opinion,
Dedric

PS: We are in the midst of a blizzard here - about 10" on the ground now
with winds up to and over 40mph.

On 10/25/06 3:53 PM, in article 453fdae4@linux, "Tony Benson"
<tony@standinghampton.com> wrote:

> That's what I understand, but I'm not a tech geek (no offense to the tech
> geeks of course) on how different DAW's handle the math involved in changing
> gain at the per track (channel) level. Maybe since the math involved is
> handled at a higher level (32 bit floating? whatever Integer?) the actual
> bit reduction isn't an issue.
>
> I can say that I didn't notice anything strange going on with my little test
> recording as far as "graininess" or anything else I would call "low bit"
> sounding. It was actually the opposite. I was able to hear more separation
> and nuance than mixing at higher channel levels. I know it sounds cliché,
> bit I could hear more space around each track. It was much easier to get
> things to "sit right" in the mix. I'm going to try this on some higher track
> counts and see if it still holds true.
>
> Tony
>
>
> "Mic Cross" <crzymnmchl@cocmast.net> wrote in message
> news:453fd7ae$1@linux...
>>
>> Quote from Dedric a little further down:
>>
>> "I always thought Paris was harder to get a clear top end out of. Nuendo
>> sounded clearer to me immediately. Some of that was Paris' converters,
>> some
>> wasn't. If tracks are being cut by 22dB before you even start processing
>> you are losing 3.5 bits of resolution from 24-bit files (depending on how
>> Paris transfers to larger bit depths for processing, and where it lops
>> them
>> off in the end)."
>>
>> The 22db cut is at mix stage rather than tracking, right? So I think
>> (would
>> love to be corrected!) that Dedric is talking about a 3.5 bit loss as
>> Paris
>> works its magic. Is this right?
>>

>> Mic.
>>
>

Subject: Re: The nut we have to crack
Posted by Dedric Terry on Thu, 26 Oct 2006 15:19:35 GMT
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Hi Lamont,

Yes, there is a difference between how analog handles "headroom", and how digital does, but there is no difference between how digital desks and DAWs handle it - they are all limited to 0dBFS for the actual digital data that passes through. There may certainly be differences in how a digital desk manages the digital path, or how you mix on it, but that doesn't necessarily mean it's more than 24-bit all the way through. Unless a desk is completely mixing within a cpu to maintain full floating point math, it will be fixed point, either 24 or 48 bit for most of the path - the same as TC Powercore or ProTools.

As far as comparing stereo wav files, if there is a difference with one DAW vs. others, the one isn't representing the stereo track correctly. I can open up any stereo wav file in SX, Nuendo, Sequoia, Vegas, and even iTunes (since all of my systems are piped to the same playback converters), and all sound identical, regardless if the original track was mixed on an SSL, analog, ProTools, or any other DAW.

Now, if a DAW only supports mono files (e.g. Paris and ProTools), converting an interleaved stereo wav file could sound different when played back in dual mono, but that would likely be to a difference in pan law between the source and the playback DAW, or alignment issues. I have heard this happen (in Paris I believe) - the two channels sound wider, but the middle sounds disconnected and almost "missing" with dual mono files for a stereo track, where it sounds a little less wide but coherent across the middle as interleaved stereo. I wouldn't say this *should* be the case with interleaved stereo vs. dual mono, but it could be.

Obviously your workflow works great for you. Mine works great for me. While a lot of engineering has a consistent basic technical methodology, personal preference still plays a significant role. Now if we could just get the "engineers" (aka artists' brothers in law, best friends, etc) that are putting bad mixes on the radio to connect the technical basics with inexperienced preference, we might be able to tune into listenable music again some day...

Regards,
Dedric

On 10/25/06 8:40 AM, in article 453f775c$1@linux, "LaMont"
<jjdpro@ameritech.net> wrote:

>
> Dedric,
> My point has moe to do with 'head-room' of ITB mixes versus, using a analog
> or digital mixer for summing.
>
> There is a difference. Also, I challege anyone to open up say SX, DP, Logic
> and play a stereo way file @ unity gain ..then, If you have copy of say
> Pr-Tools
> LE M-powered, import that same file.. Then listen.
> You can here the difference, even using the same audio interface..
>
> I agree with you that you have to mix differently using the natives, but
> Soft ware has a sound.. To me and others, to get make SX/Nuendo slam at it's
> best, is to using a outboard summing mixer.
>
> These days, my work flow is to record,edit,then bounce stems from Nuendo.
> Simply put, there is no better workflow DAW on the planet for such tasks.
> Then, I either mix in Pro-Tools or Paris depending on the color I'm going
> for.
>
> Dedric Terry <dterry@keyofd.net> wrote:
>> Lamont - if your D-A converters affect the way you mix inside a DAW, you
>> aren't mixing what you think you are. Certainly converters can sound
>> different, but the differences at the RME/Apogee level aren't in siginficant
>> areas (mainly a slight difference in sound of the top end - yes I've heard
>> all of these, along with Myteks, Cranesong, DCS, and others side by side
> -
>> Cranesong is my favorite - Myteks are great, but a little sterile. DCS is
>> just too expensive).
>>
>> 1) If you are saying you mix differently on a console because you are using
>> Apogee converters from the DAW with soft limit vs. RME converters, also
> into
>> the same desk (no mixing in SX, just playback), you are simply using the
>> converters to color the signal (albeit only slightly), in different ways
> -
>> Softlimit just limiting. Nothing wrong with that, but that is altering
> the
>> tracks going in, not the mixing platform itself. Saying RME converters
>> limit you because they don't have a limiter built in says you aren't mixing
>> the way most of us do - you are trying to get analog saturation out of
>> digital - ain't gonna happen.
>>

>> 2) If you are mixing inside SX and change your approach depending on which
>> converter you monitor through, then that's a problem since your bounces
>> aren't going to be the same, and your decisions aren't going to be
>> consistent. The idea is to have your monitoring chain *not* affect your
> mix
>> decisions, but enable more accurate ones.
>>
>> If you mixdown to a 2-track of some sort (Masterlink, etc), then you are
>> using SoftLimit as a limiter on the output. You could achieve the same
>> think in a multitude of different ways.
>>
>> Regards,
>> Dedric
>>
>> On 10/24/06 1:44 PM, in article 453e6d24$1@linux, "LaMont"
>> <jjdpro@ameritech.net> wrote:
>>
>>>
>>> Neil I do mix follow the native mix rules. No overs, faders around -5db
> ect,
>>> and I can make it sound good..
>>>
>>> However, when I add in a mixer for summing, all of those native mixing
> rules
>>> are out the window. The whole mix "sonically" opens up..
>>>
>>> As well as, If I'm using Apogees AD16x/DA16x with soft-limiter set on,
> I
>>> can mix like I want to in SX. With RME interface's and converters, I have
>>> to abide by the rules.
>>>
>>> Lastly, when i have to mix (In the Box) using SX/Nuendo, I refer to the
>>> Charles
>>> dye method and add in Harmonic distortion via plugs in (namely) antares
> Mic
>>> modler(tube) on the inserts. This gives a different texture to the faders.
>>> These days,I just use the SSL plugs which have that harmonic distort color
>>> that helps a native mix...
>>> "Neil" <OIUOIU@OIU.com> wrote:
>>>>
>>>> "LaMont" <jjdpro@ameritech.net> wrote:
>>>>>
>>>>> My Point exactly.. If all of you who use Nuendo or Cubase cannot hear
> that
>>>>> there is something going on (software-wise) in Cubase or Nuendo that's
>>> not
>>>>> bringing "Full-life" to our wav files, then,I'm sorry, your ears are
> not

>>>>> as good as you may think..
>>>>
>>>> There IS something going on... IME, I think that a lot of people
>>>> are using the tool in a manner in which it was not designed for.
>>>> It's not designed to accomodate 50 tracks worth of clips/overs
>>>> resulting in hundreds of thousands of errors per second... it's
>>>> as simple as that.
>>>> I don't think anyone who's said you can get good mixes out of
>>>> Native suystems has insisted that it sounds exactly like Paris
>>>> (or PT, or analog, or anything else), so is something different
>>>> going on? Yeah... it's different - doesn't mean that it can't be
>>>> good.
>>>
>>
>

Subject: Re: The nut we have to crack
Posted by Tony Benson on Thu, 26 Oct 2006 18:01:48 GMT
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Thanks Dedric. I really don't consider myself to be an "engineer". I'm a
songwriter/musician who wants record professional sounding tracks in my home
studio. I've been into mixing both live and in the studio for 20 years or
so, but am completely self taught. Any "real" knowledge is always
appreciated. Anyway, this new approach (for me anyway) might just let me
actually get the kind of mixes I hear in my head out of DP.

Tony

"Dedric Terry" <dterry@keyofd.net> wrote in message
news:C16624A7.4B4C%dterry@keyofd.net...
> Actually the 3.5 bit loss assumes either of two situations:
>
> 1) I'm guessing Chuck was referring to the EDS code, so if the gain
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> then it would pad 0's for any extra bits beyond 24.
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>
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>>> works its magic. Is this right?
>>>
>>> Mic.
>>>
>>
>

Subject: Re: The nut we have to crack
Posted by TC on Thu, 26 Oct 2006 19:53:45 GMT

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I'm going to try that also in DP.

I just find Paris such a joy to mix in though.

Cheers,

TC

Tony Benson wrote:
> Thanks Dedric. I really don't consider myself to be an "engineer". I'm a
> songwriter/musician who wants record professional sounding tracks in my home
> studio. I've been into mixing both live and in the studio for 20 years or
> so, but am completely self taught. Any "real" knowledge is always
> appreciated. Anyway, this new approach (for me anyway) might just let me
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>
> Tony
>

Subject: Re: The nut we have to crack
Posted by Martin Harrington on Sat, 28 Oct 2006 04:19:24 GMT
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Thanks for that well thought out "rant", Neil.
Thats pretty much the way I feel with Nuendo, although I don't put an
abitary setting on each channel, (probably because I''m only recording a
couple of tracks at a time,) but I certainly record to within about 8dbfs
from 0, and always put a limiter on the output,
--
Martin Harrington
www.lendanear-sound.com
"Neil" <IOUIU@OIU.com> wrote in message news:453d8006$1@linux...
>
> "chuck duffy" <c@c.com> wrote:
>
>>If we can't get decent mixes out of a native daw then something is wrong.
>> Let's find the thing that's wrong, and make it right.
>
> (Long, but thought-provoking, and hopefully helpful, rant
> follows):
>
> I think the thing that's wrong is that some people just can't
> get their heads around the differences between analog & digital.
> With analog, "big" = hotter, and so hotter is better. When you
> overbias your tape machines & smack the hell out of the tape,

> you're getting compression right off the bat on every track you
> do that with, so one gets used to hearing most tracks with some
> degree of tape compression already... and we all know that
> compression can make things sound "bigger". Or, you use a
> compressor on the way in to the tape so that you get a better
> SNR, but since that's not an issue with digital (unless you're
> recording at levels so low that you just simply get poor
> resolution, but that's a slightly different scenario), people
> quit using compressors on the way in to digital since SNR isn't
> an issue there.... you also can't smack an AD convertor hard &
> expect it to like it - unlike tape. So right off the bat we've
> got a whole different set of dynamics action going on from one
> world to the other - then, when you've already got that
> compressed kick or bassline on tape, you compress it more, and
> you're compressing an already-compressed signal, so when you
> apply compression to your uncompressed kick on your DAW you're
> thinking "nah, that CAN'T be right, it can't need THAT much
> compression! I'd better back that off a bit!" (because you're
> looking at the ratios & the threshhold, etc, instead of using
> your ears). EQ reacts differently with digital, too... if you're
> used to mixing on a console, you might be used to boosting or
> cutting something by 3, 4, 6db & getting an audible
> difference... with digital/plugin EQ's, sometimes you gotta
> boost or cut HUGE swaths of that frequency to really make a
> difference... why? I think it's a phase thing... you get more
> phase shift with analog filters, and so the change is more
> apparent at smaller degrees of boost & cut. That also helps to
> isolate things to have their own place in the mix at the same
> time... considering that phase is the reason we have two ears -
> it's the thing that makes it possible for us to tell which
> direction a sound is coming from - this makes perfect sense.
>
> So, those of us (and I think that's "most of us here") who cut
> our teeth in the analog world first, and are used to all the
> things mentioned above - and who have not changed that style of
> mixing - could be disappointed in Native systems - not because
> they fall short of analog or Paris, but because they are
> actually much more accurate (assuming good quality convertors)
> & as a result do not impart certain types of coloration that we
> might interpret as "pleasing". If you could go back to a great
> mix you did on analog & a console & take out half of the amount
> of dynamics processing & half of the amount of EQ'ing you did,
> what would you get? A mix that sounded flatter & more colorless
> & with less dimension than the one you ended up with. Want
> proof? Here it is: If you didn't need the amount of EQ &
> dynamics you applied, you wouldn't have done so! If half the
> amounts/degrees of those things would have sufficed, that's
> what you would have used! So Paris sounds & acts kinda like

> analog, and people who like Paris like that aspect of it... how
> do we know there's not a few lines of code in there somewhere
> that adds graduated degrees of even-harmonic distortion when
> you push the faders or saturate the mix buss to whatever
> degree? I personally don't think it's strictly a DSP thing,
> because let's face it.. a plugin is basically doing the same
> thing to your mix whether it's running of a processor on it's
> own card or off your CPU; the difference being how well a
> particular VST or Direct-X compressor or reverb is written (and
> what it's designed to do in terms of treating the sound) vs.
> whatever DSP compressor or reverb plugin you're talking about.
> Can I get an "Amen, brutha!" on that?
>
> Chuck's nailed the Paris mix buss thing, it seems, with that
> -22db at the channel & +22db at the mix buss, but WHY does that
> make a difference? Well, here's why gang... it's just as I said
> earlier in another thread - you've got to give yourself some
> headroom, dammit! Paris apparently does this for you. Want to
> prove me wrong? Open up a Paris mix and drag the mix buss
> master fader down 22db from wherever you have it, then insert
> any plugin that has an output level control on each individual
> channel of that mix - if the plugin is a compressor, for
> example, don't use any compression, just use the output
> control - now boost every channel by 22db using that output
> control... if it only goes up 10 db, then insert that plugin
> twice in a row & max out the output on each insertion...
> that'll be close enough... how's that sound? I'll bet it won't
> sound all that good! Are you hearing that "overstuffed" mix
> buss sound? Is it smaller, with less dimension? I'd be curious
> to see what you guys think if you try this. Now that we know
> what Chuck told us he discovered, this is the best way to see
> if that makes a difference or not (my guess - it DOES make a
> difference, otherwise, they wouldn't have written the code that
> way!).
>
>
> So how can you get "big" in Native? Give yourself what Paris
> apparently already gives you... some headroom - think "clean",
> then dirty it up if you have to later... hell, just mash the
> mix with a comp & limiter or an L2 or something equivalent -
> you'll get all the harmonic distortion you want. I wasn't
> kidding the other day when I said: "Think zen when mixing in
> Cubase" it's all gotta flow without clips, gang... think about
> it... if you have one channel getting "overs" in a 32-bit float-
> point system, you may not notice it... heck you can't notice
> each sample in a given sound file can you? Of course not. But
> if you start adding more channels, and each of those channels
> is running hot... let's say 32 channels - as a comparison

> for you guys running two-card paris systems & no native mixes.
> and let's say you're running hot (over zero) about 25% of the
> time on each channel - that's 352,000 errors PER SECOND across
> the 32 tracks. That's a lot of floating-point math going on
> there, isn't it? And in this scenario, I want you to think of
> each error as a mistake, because that's what it is... in this
> style of mixing, it's a mistake. How can you expect something
> that's got 352,000 mistakes per second going on, to sound good?
>
> Are you still not convinced? Then you should also definitely
> investigate running stems (submixes) & reimporting. When I've
> done this I definitely can hear a difference, and I suspect you
> most likely will be able to as well.. it is NOT a huge
> difference, but it's audible. In fact, some months ago I posted
> a stems mix vs. a non-stems mix & a number of you said you
> could hear a difference. Now, if you think "aww, this is just
> another pain-in-the-ass procedure I have to go through if I mix
> in Native", keep in mind that you can run 90 Million stems
> mixes in the time it will take Deej to set up his first Pulsar
> card, and another 900 million in the time that it takes Chuck
> to research & write that plugin (OK, just giving hell to Deej
> there, and no really no offense intended to Chucks coding
> capability, but I'm just saying this is something you can do
> RIGHT NOW, TONIGHT if you want to if you have a Native system,
> without having to wait for anything new). Now, if you have a
> small project - one acoustic guitar, piano, & a vocal - with
> just a few tracks, running stems won't make a difference, but
> if you have a large project, give it a shot... you may not hear
> enough of a difference to make it worth doing in any given
> instance, but then again, you might.
>
> So, now that I hope I've made my case, here's my own personal
> guidelines for Native mixing - try it out & see wat you think:
>
> 1.) Do NOT bring down your Master Fader. It stays at zero
> (unless you're doing a fade).
>
> 2.) On your Master inserts, use a peakstop/brickwall limiter
> set anywhere from -.03 to -3db, depending on how much headroom
> you want to give your mastering engineer. Settings for volume
> maximization & other parameters will, of course, depend on the
> program material.
>
> 3.) Record at 24-bit 88.2k or higher (Dan Lavry has a white
> paper that makes a good case for a 60k sample rate - in order
> to get the ringing from the convertors' FIR filters out of the
> top range of our hearing - but since there is no standard 60k
> sample rate, 88.2 is the next one up). Also, 16-bit may have

> worked with Paris for whatever reason (maybe it just enhanced
> the harmonic distortion you're hearing?), but let's face it,
> everybody knows that more bits = greater "truth", especially
> when combined with higher resolutions.
>
> 4.) Default your individual channel settings to -6db or lower...
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> a decent amount of tracks without overloading the mix buss &
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> it lower as a starting point if you plan on getting into the
> range of 40+ tracks. HERE'S THE KEY... if you've got your mix
> roughed out & you can pull out that peakstop limiter I
> mentioned in #2 & NOT go over zero on the Master - you're
> golden. Fuck it, set 'em all at -15 as a starting point if you
> want, Paris is already setting them for you at -22, right? If
> you're getting a few scant overs without the limiter, you're
> still ok, really... the idea is not to overstuff the mix buss
> so heavily that if you pull the limiter off you're going into
> the +5, +6 range without it.
>
> Think "clean" people = think "no clips" (or as few as
> possible), you get 30-40 channels of "overs" constantly (like
> the 352,000 of 'em per second in the example I gave earlier),
> and it's going to get harsh & thin.... it's a cumulative effect.
>
> That's it, really... it's just like any other tool - you can't
> use an allen wrench to properly drive a nail, and you can't use
> a hammer to trim your nose hair.
>
> Happy Native mixing!
>
> (think "zen"!)
>
> Neil

Subject: Re: The nut we have to crack
Posted by Martin Harrington on Sat, 28 Oct 2006 04:32:01 GMT
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Hmmm
So you're happy to truncate 24 bit files to 20 bit and then convert back to
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Martin Harrington
www.lendanear-sound.com

"espresso" <audio@espressodigital.com> wrote in message

news:453dc969$1@linux...
>I posted this a while ago.. I now use Nuendo with a Layla 3G to output 8 x
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>> >Chuck
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Subject: Re: The nut we have to crack
Posted by Ted Gerber on Sat, 28 Oct 2006 09:58:02 GMT
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Hi Martin-

I specifically remember others in the past using the Paris Adat In from
24 bit sources, being thrilled with the sound, happliy knowing truncation

was happening. As David says below:

"honestly, the end justifies the means"

FWIW

Ted

"Martin Harrington" <lendan@bigpond.net.au> wrote:
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Subject: Re: The nut we have to crack
Posted by animix on Wed, 01 Nov 2006 18:13:13 GMT
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Deej

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Subject: Re: The nut we have to crack
Posted by Martin Harrington on Thu, 02 Nov 2006 04:31:00 GMT
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Martin Harrington
www.lendanear-sound.com

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Subject: Re: The nut we have to crack
Posted by animix on Thu, 02 Nov 2006 22:47:39 GMT
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Yeah.....if full spec DVD (as in 96k/24bit) was my destination format, I'd
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;o)

"Martin Harrington" <lendan@bigpond.net.au> wrote in message
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Subject: Re: The nut we have to crack
Posted by Martin Harrington on Thu, 02 Nov 2006 23:33:38 GMT
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Good Point
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Martin Harrington
www.lendanear-sound.com

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>> >> >> of the EDS alongside the code from the Paris OS to basically allow
>> >> >> a
>> >> >> Paris
>> >> >> mix bus, using something like rewire, to plug in to a native app.
>> >> >>
>> >> >> I believe the talk was inspired by Matthew Craig's efforts in
> creating
>> >> >> the
>> >> >> VST Paris EQ, which does basically this same thing, emulating the
> EDS
>> >> > functionality
>> >> >> and hence generating pretty much identical output to the same audio
>> > going

>> >> >> through the card itself.
>> >> >>
>> >> >> This would sure sort out the issues if anybody with enough knowhow
> and
>> >> > dedication
>> >> >> got on board. Suddenly any app could have the Paris mix bus, not to
>> >> > mention
>> >> >> the paris EQ...   that would pretty much put an end to all this
>> >> >> shennigans
>> >> >> i would think.
>> >> >>
>> >> >> Cheers,
>> >> >> Kim.
>> >> >>
>> >> >> "chuck duffy" <c@c.com> wrote:
>> >> >> >
>> >> >> >DJ,
>> >> >> >
>> >> >> >Listen I know you love messing with this stuff, but I think we
>> >> >> >need
>> >> >> >to
>> >> > focus
>> >> >> >on how to get the mixes we want out of an all native system.
>> >> >> >
>> >> >> >It just doesn't make any sense to me to get onboard with another
>> > weird,
>> >> >> proprietary
>> >> >> >dsp system. Creamware is as weird, oddball nad proprietary as it
>> > gets.
>> >> >>
>> >> >> >Why bother with it? Why bother with UAD or anything else. It
>> >> >> >just
>> >> > doesn't
>> >> >> >make sense to me.
>> >> >> >
>> >> >> >If we can't get decent mixes out of a native daw then something is
>> >> >> >wrong.
>> >> >> > Let's find the thing that's wrong, and make it right.
>> >> >> >
>> >> >> >Chuck
>> >> >>
>> >> >
>> >> >
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