Subject: Re: Whoa! That's gorgeous!
Posted by DJ on Fri, 10 Nov 2006 18:16:59 GMT
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hat I'm going to do sort of a light "mastering" or pre-
mastering on... I'll render the 2-buss mix down a 32-bit/88.2k,
then use Ozone on either the 2-buss mix itself (in this case)
or the master module of the stems mix in those instances.

>Why even start recording a such a high rate, when the 24/32
>sounds great?

Because I am a complete & utter moron, apparently... I know
that more people have been getting into using this particular
samplerate lately, but it seems the vast majority of people who
don't use Pro-Tools are dead-set against going above 44.1 (or
sometimes 48k), while a lot of pepole using PT are going for
96k and some are apparently starting to use 88.2k - a couple of
posts ago in this thread I answered your above question on the
samplerate already... unless you're asking something different
now, and I don't get what you mean?

NeilWhile I agree that the same things could be done with native DSP code, if
the only way to get a John Bowen synth is to buy a Pulsar, because Bowen
prefers copy protection to size of potential market, then that's that. I
was worried about stability but if that is taken care of then I'm in.

I also think it's odd that people will pay absurd amounts of money for esoteric
analog recording hardware (mics, pres, cables, EQs, comps) when if even one
of the links in the chain is subpar, even the quality of power in the building,
can 86 those thousands of dollars spent. But that makes perfect sense, in
contrast to buying an esoteric DSP card that only sounds a bit better than
the native apps (the NI Prophet sounds very good, the Creamware Prophet sounds
sick). Isn't that a strange double standard?

None of which means I might not be cursing the day I ordered a Pulsar, but
seeing the number of people there using the gear makes me somewhat optomistic.
I just think it's odd that the pursuit of superior sound is fine when it's
tube amps and vintage limiters, but it's silly when it's a DSP synth.

TCB

Chris Ludwig <chrisl@adkproaudio.com> wrote:
>HI Dj and Neil,
>It's sadly amusing to see that Creamware's way of doing things hasn't
>changed at all sense I dealt with them years ago. Some of the user
>responses you all got are are hilarious in their denial of facts.

>Any of the effects ava

Subject: Re: Whoa! That's gorgeous!
Posted by DJ on Fri, 10 Nov 2006 18:57:28 GMT
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und a couple
>more years so has been more commonly available on digital clocks.
>
>
>Chris
>
>DJ wrote:
>
>><evil grin>
>>
>>"Neil" <IUOIU@OIU.com> wrote in message news:457c65bb$1@linux...
>>
>>
>>>"Don Nafe" <dnafe@magma.ca> wrote:
>>>
>>>
>>>>I guess the big question is are you taking an unacceptable sonic hit
a
>>>>44.1
>>>>
>>>>
>>>>vs 88.2 and does the summing using the Pulsar offset the sonic hit you

>>>>take
>>>>
>>>>
>>>>(if in fact you do)
>>>>
>>>>If the answer is no...dump them ASAP
>>>>
>>>>
>>>Do you mean if the answer is "yes", dump them ASAP? Or do you
>>>mean if the answer is "no" I should dump using 88.2k ASAP?
>>>
>>>Frankly I don't know if using the Pulsar for summing would make
>>>up for the sonic hit I would take at 44.1k - I can use Paris
>>>for summing right now & NOT have to take the hit to
>>>downconvert, though I have to go out through several Analog
>>>submixes to do this. My idea with the Pulsar was
>>>ess

Subject: Whoa! That's gorgeous!
Posted by DC on Sun, 12 Nov 2006 01:17:27 GMT
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couple years (1 card/MEC/24-8 in/out on a dedicated
>>>MAC
>>>>>>>G4
>>>>>>>>Quicksilver

Subject: Re: Whoa! That's gorgeous!
Posted by Neil on Sun, 12 Nov 2006 16:52:38 GMT
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w, any input channel can be routed and mixed
> to any hardware output (third row.)
>
>
> Middle row: playback channels (playback tracks of the software.)
>
> Using the fader and routing window, any playback channel can be routed and
> mixed to any hardware output (third row.)
>
> Lower row: hardware outputs. Because they refer to the output of a subgroup,
> the level can only be attenuated here (in order to avoid overloads), routing
> is no

Subject: Re: Whoa! That's gorgeous!
Posted by AlexPlasko on Sun, 12 Nov 2006 17:04:50 GMT
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ignal (complete ASIO Direct
> Monitoring)
>
>
>
> naming channels
>
>
>
> The channel names shown in the white label area can be edited. A right mouse
> click on the white name field brings up the dialog box Enter Name. Any name
> can be entered in this dialog. Enter/Return closes the dialog box, the white
> label now shows the first letters of the new name. ESC cancels the process
> and closes the dialog box.
>
>

>
> post send mode
>
>
>
> Dragging the faders by use of the right mouse button activates Post Send
> mode and causes all routings of the current input or playback channel to
> be changed in a relative way. Please note that the fader settings of all
> routings are memorized.
>
> So when pulling the fader to the bottom (maximum attenuation), the individual
> settings are back when you right click the mouse and pull the fader up.
>
> The individual settings get lost in m.a. position as soon as the fader is
> clicked with the left mouse button.
>
> As long as no single level is at m.a. position, the left mouse button can
> be used to change the current routing's gain.
>

Subject: Re: Whoa! That's gorgeous!
Posted by DC on Sun, 12 Nov 2006 17:39:59 GMT
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t possible. This row has two additional channels, the analog outputs.
>
>
>
> more info
>
>
>
> Additional documentation can be found here:
> http://www.rme-audio.com/english/techinfo/hdsp_tmhard.htm
> http://www.rme-audio.com/english/techinfo/hdsp_tmsoft.htm
>
> This card can be good for:
>   * setting up delay-free submixes (headphone mixes)
>   * unlimited routing of inputs and outputs (free utilization, patchbay
> function)
>   * distributing signals to several outputs at a time
>   * simultaneous playback of different programs over only one stereo channel
>   * mixing of the input signal to the playback s

Subject: Re: Whoa! That's gorgeous!

Posted by Gary Flanigan on Sun, 12 Nov 2006 19:15:16 GMT
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ilable could be coded and ran in a current native

>system. Creamware, TC and UAD types do not want to do it for the simple

>reason people could easily steal the software. The big thing that the
>Creamware DSP cards offer which same goes for Pro Tools is the ability
>to do it very low latencies so as to make is seem real time. The UAD
>and TCs don't even offer this and actually quite the opposite. In native

>systems you can't run the latencies to their lowest settings when using

>these plug ins.
>
>I have a feeling that the reason the does not do 88.2k is because of
>hardware limitations. It may be that the clocks they use on their cards

>which are fairly old right now may not have had the ability to support
>the 88.2k clock. People as fair as I've seen didn't really start trying

>to use 88.2k till the past 2/3 years but 96k has been aro

Subject: Re: Whoa! That's gorgeous!
Posted by DC on Sun, 12 Nov 2006 21:41:37 GMT
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you do)
>>>>>
>>>>>If the answer is no...dump them ASAP
>>>>>
>>>>>
>>>>Do you mean if the answer is "yes", dump them ASAP? Or do you
>>>>mean if the answer is "no" I should dump using 88.2k ASAP?
>>>>
>>>>Frankly I don't know if using the Pulsar for summing would make
>>>>up for the sonic hit I would take at 44.1k - I can use Paris
>>>>for summing right now & NOT have to take the hit to
>>>>downconvert, though I have to go out through several Analog
>>>>submixes to do this. My idea with the Pulsar was
>>>>essentially: "What if I can sum in the digital domain via DSP;
>>>>and if so, would that sound better than what I can do right
>>>>now?" At this point I still can't find out, however, due to the
>>>>inability of the Pulsar stuff to work at 88.2k.
>>>>
>>>>If you want to see the minor shitstorm that Deej & I started

>>>>over on the Pulsar forum over this issue, go here:
>>>>
>>>>http://www.planetz.com/phpBB2/viewtopic.php?t=20885
>>>>
>>>>
>>>>Neil
>>>>
>>>>
>>>
>>>
>>>
>>>
>>
>>--
>>Chris Ludwig
>>ADK
>>chrisl@adkproaudio.com <mailto:chrisl@adkproaudio.com>
>>www.adkproaudio.com <http://www.adkproaudio.com/>
>>(859) 635-5762
>Nice Drum sound ... great attack and balance. Thanks for posting it!

"DJ" <nowayjose@dude.net> wrote:
>There's a "Mixtravaganza" kinda thing happening over at 3dB forum. I figured

>I'd bite and compare my mix to other *proper* mixes and see if there was
any
>audible sonic horribleness with the *improper* scenario I'm using wherein

>I'm taking files that are processed in Cubase SX at 32 bit float, truncated

>over Paris ADAT, then summed in Paris at 24 bit, then IDR'ed in Wavelab
to
>16 bit, then further mangled to MP3.
>
>I'm thinking I'm not going to think twice about this any more, ever.
>
>Thanks to Neil for hosting this:
>http://saqqararecords.com/MiscAudio/animixsugar1.mp3
>
>;o)
>Deej
>
>Simon, I read something about this in "the mixing engineer's handbook'.
Got me curious now, I'll have to check that out.
Think it was -3dB avg

Rob

<Simon hexagometer (at) gmx.net> wrote in message news:457d8136$1@linux...