Subject: Paris Skins - alternatives (3)
Posted by Yanoska on Fri, 06 Oct 2006 17:43:58 GMT
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>
> Well, let me put it this way - there's a reason you get the source code
with
> an open source project...
>
> Doug ;-)
>
> http://www.parisfaqs.com
>WORKS GREAT WRAPPED.
YOU HAVE TO OFFSET THE TRACKS BY 8960 SAMPLES OR WHATEVER IT IS.
aFTER THAT YOUR GOOD TO GO.

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>> 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
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>> looking at the ratios & the threshhold, etc, instead of using
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>> 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
>> 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.
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>> 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,
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>> amounts/degrees of those things would have sufficed, that's
>> what you would have used! So Paris sounds & acts kinda like
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>> 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
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>> 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
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>> way!).
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>> 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?
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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
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>> to research & write that plugin (OK, just giving hell to Deej
>> there, and no really no offense intended to Chucks coding
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>> 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
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>> (unless you're doing a fade).
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>> 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
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>> 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
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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.
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>> 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
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>> 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
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>> the +5, +6 range without it.
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>> possible), you get 30-40 channels of "overs" constantly (like
>> the 352,000 of 'em per second in the example I gave earlier),
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>> a hammer to trim your nose hair.
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>> Happy Native mixing!
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>> (think "zen"!)
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>I believe it works but it cuts off part of the GUI if you don't wrap it. If
my
recolection is recolating properly. It might be a bit more stable wrapped as
well.

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> Very interesting stuff! Question: does lowering the amplitude
> reduce bit depth/resolution? Or does this not apply here? I remember
> one
> discussion where digital amplitude was related to resolution.

>
> Mic.
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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.I just installed Paris on XP and if I bott the comp frsh and run Paris I can
work on that song with no problems, however when I exit that song the next
song loads but won't play. I have to reboot the comp and then Paris to start
the next song. Also, if I try to load any previously recorded song that have
native plugins running it either locks or will play that song, but won't
load another. I am completely stumped, I need help!

Thanks,
MikeI 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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>If it's a stool sample...........well...........never mind.
Rod
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>
>
>What's a sample among friends?
>T.
>
> "Dimitrios" <musurgio@otenet.gr> wrote in message =
>news:453f8b69$1@linux...
>
> Hi,
> The SSLcompressor has 0 latency.
> Thee SSL channel has 1 sample latency.
> Regards,
> Dimitrios
>
> "LaMont" <jjdpro@ameritech.net> wrote:
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> >yes it works. Thank god Waves still code their plugins in Direct-x. =
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=
>10
> >tracks.=20
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> >"Goran Stojiljkovic" <goran.stojiljkovic@os.t-com.hr> wrote:
> >>does it work?
> >>latency ?
> >>
> >>please answer......=20
> >>
> >>
> >
>
>
>
>I choose Polesoft Lockspam to fight spam, and you?
>http://www.polesoft.com/refer.html
>
><!DOCTYPE HTML PUBLIC "-//W3C//DTD HTML 4.0 Transitional//EN">
><HTML><HEAD>
><META http-equiv=3DContent-Type content=3D"text/html; =
>charset=3Diso-8859-1">
><META content=3D"MSHTML 6.00.2800.1400" name=3DGENERATOR>
><STYLE></STYLE>

></HEAD>
><BODY bgColor=3D#ffffff>
><DIV><FONT face=3DArial size=3D2>What's a sample among =
>friends?</FONT></DIV>
><DIV><FONT face=3DArial size=3D2>T.</FONT></DIV>

><BLOCKQUOTE=20
>style=3D"PADDING-RIGHT: 0px; PADDING-LEFT: 5px; MARGIN-LEFT: 5px; =
>BORDER-LEFT: #000000 2px solid; MARGIN-RIGHT: 0px">
> <DIV>"Dimitrios" <<A=20
> href=3D"mailto:musurgio@otenet.gr">musurgio@otenet.gr</A>> wrote in =
>message=20
> <A=20
> =
>href=3D"news:453f8b69$1@linux">news:453f8b69$1@linux</A>...</DIV>
Hi,<=
>BR>The=20
> SSLcompressor has 0 latency.
Thee SSL channel has 1 sample=20
> latency.
Regards,
Dimitrios

"LaMont" <<A=20
> href=3D"mailto:jjdpro@ameritech.net">jjdpro@ameritech.net</A>>=20
> wrote:
>
>yes it works. Thank god Waves still code their =
>plugins=20
> in Direct-x. Surprisely,
>the plug added no latency. However, =
>the mix I=20
> was working on only had 10
>tracks.
>
>"Goran =
>Stojiljkovic"=20
> <<A=20
> =
>href=3D"mailto:goran.stojiljkovic@os.t-com.hr">goran.stojiljkovic@os.t-co=
>m.hr</A>>=20
> wrote:
>>does it work?
>>latency=20
> ?
>>
>>please answer......=20
>
>>
>>
>
</BLOCKQUOTE>
><DIV><FONT size=3D2>

I choose Polesoft Lockspam to fight spam, =
>and=20
>you?
<A=20
>href=3D"http://www.polesoft.com/refer.html">http://www.polesoft.com/refer=

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>>>> get their heads around the differences between analog & digital.
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>>>> 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
>>>
>>>
>>
>
>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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>>>>> 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
>>>>
>>>>
>>>
>>
>>
>hi gene
 I also went to a sonar demo last week, and that 64 bit subject came up.
according to the rep the sonar audio engine is running in 64 bit mode in
windows xp 32 bit already. you dont have to have win xp 64.
I have to admit i was impressed with the improvements they made .most
notibly the new features like real time audio warping.Its like they are
giving you vocalign,melodyne,recycle,drumagogg,and a fewother programs
integrated into sonarV 6.The rep had a mix of a queen tune they were doing a
remix on ,and it sounded pretty damn good, even through the PA speakers
they substituted for monitors.
I love cubase 4 , but that demo has me thinking again.
and it supports VST! imagine that.
Gene Lennon" <glennon@NOSPmyrealbox.com> wrote in message
news:453f7f7d$1@linux...
>
>
> I went to a Sonar 6 demo last night. I am primarily interested in the
> 64-bit
> mix engine and how it sounds. My last experiment with Sonar was several
> years
> ago and I hated the sound.
> Zac Kenney from cakewalk gave the demo. Not that it was a surprise, but
> the
> setup was not appropriate to judge audio quality, although I did hear one
> acoustic track that may sound very good in a better environment.
> Some high-end plug-ins like the Refined Audiometrics PLParEQ EQ work
> native
> in Sonar at 64 bits (Audio engine). I have not heard this yet but is
> should
> be very good. I hope Sony and Algorithmix do the same. The included
> convolution
> reverb also runs native at 64 (as do most of the included plug-ins and
> some
> of the VSTIs). A demo version is not available yet, so I will have to
> wait.
>
> Audio engine aside, the feature set and ergonomic aspects of Sonar are
> very
> impressive.

> Running on a 64 bit OS, Sonar can address all the RAM you would ever need.
> A 100 plus track session with streaming video and many VSTIs seemed like
> a "walk in the park." on a dual core PC.
> It did crash during the demo, so overall the jury is still out.
>
> Gene
>Dumb Q - where does one get this beast and the impulses

Don

"Rod Lincoln" <rlincoln@nospam.kc.rr.com> wrote in message
news:453fd1cd$1@linux...
>
> Brandon is correct about the GUI. The newest version only has 200 ms
> latency,
> so it works great with Paris. Just hit the nudge 100 back button 2 times
> and your good to go...sample accurate.
> Rod
> "Brandon" <a@a.com> wrote:
>>I believe it works but it cuts off part of the GUI if you don't wrap it.
> If
>>my
>>recolection is recolating properly. It might be a bit more stable wrapped
> as
>>well.
>>
>>
>>Brandon
>>
>>

>>news:453fba4e@linux...
>>> Thanks, Brandon.
>>> Wrapped means it does'nt work as a 'regular' VSt plug ? i need a
>>> wrapper,
>>> correct ?
>>>

>>> > WORKS GREAT WRAPPED.
>>> > YOU HAVE TO OFFSET THE TRACKS BY 8960 SAMPLES OR WHATEVER IT IS.
>>> > aFTER THAT YOUR GOOD TO GO.
>>> >
>>> >
>>> >
>>> > Brandon
>>> >
>>> >

>>>
>>>
>>
>>
>How many EDS cards, what motherboard, what slots are the cards in, what
  Irqs is it sharing, are these new projects, post your paris.cfg file,
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> native plugins running it either locks or will play that song, but won't
> load another. I am completely stumped, I need help!
>
> Thanks,
> MikeThis is a multi-part message in MIME format.

------=_NextPart_000_0036_01C6F862.01B217D0
Content-Type: text/plain;
 charset="iso-8859-1"
Content-Transfer-Encoding: quoted-printable

oh.
  "Rod Lincoln" <rlincoln@nospam.kc.rr.com> wrote in message =
news:453fd252$1@linux...

  If it's a stool sample...........well...........never mind.
  Rod
  "Tom Bruhl" <arpegio@comcast.net> wrote:
  >
  >
  >What's a sample among friends?
  >T.
  >
  > "Dimitrios" <musurgio@otenet.gr> wrote in message =3D
  >news:453f8b69$1@linux...
  >
  > Hi,
  > The SSLcompressor has 0 latency.
  > Thee SSL channel has 1 sample latency.
  > Regards,
  > Dimitrios
  >
  > "LaMont" <jjdpro@ameritech.net> wrote:
  > >

> >yes it works. Thank god Waves still code their plugins in =
Direct-x. =3D
>Surprisely,
> >the plug added no latency. However, the mix I was working on only =
had
=3D
>10
> >tracks.=3D20
> >
> >"Goran Stojiljkovic" <goran.stojiljkovic@os.t-com.hr> wrote:
> >>does it work?
> >>latency ?
> >>
> >>please answer......=3D20
> >>
> >>
> >
>
>
>
>I choose Polesoft Lockspam to fight spam, and you?
>http://www.polesoft.com/refer.html =20
>
><!DOCTYPE HTML PUBLIC "-//W3C//DTD HTML 4.0 Transitional//EN">
><HTML><HEAD>
><META http-equiv=3D3DContent-Type content=3D3D"text/html; =3D
>charset=3D3Diso-8859-1">
><META content=3D3D"MSHTML 6.00.2800.1400" name=3D3DGENERATOR>
><STYLE></STYLE>
></HEAD>
><BODY bgColor=3D3D#ffffff>
><DIV><FONT face=3D3DArial size=3D3D2>What's a sample among =3D
>friends?</FONT></DIV>
><DIV><FONT face=3D3DArial size=3D3D2>T.</FONT></DIV>
><DIV> </DIV>
><BLOCKQUOTE=3D20
>style=3D3D"PADDING-RIGHT: 0px; PADDING-LEFT: 5px; MARGIN-LEFT: 5px; =
=3D
>BORDER-LEFT: #000000 2px solid; MARGIN-RIGHT: 0px">
> <DIV>"Dimitrios" <<A=3D20
> href=3D3D"mailto:musurgio@otenet.gr">musurgio@otenet.gr</A>> wrote =
in =3D
>message=3D20
> <A=3D20
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=
>href=3D3D"news:453f8b69$1@linux">news:453f8b69$1@linux</A>...</DIV>
H=
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> was working on only had 10
>tracks.
>
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>Stojiljkovic"=3D20
> <<A=3D20
> =3D
=
>href=3D3D"mailto:goran.stojiljkovic@os.t-com.hr">goran.stojiljkovic@os.t=
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>.html</A> </FONT></DIV></BODY></HTML>
>
>

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 charset="iso-8859-1"
Content-Transfer-Encoding: quoted-printable

<!DOCTYPE HTML PUBLIC "-//W3C//DTD HTML 4.0 Transitional//EN">
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charset=3Diso-8859-1">
<META content=3D"MSHTML 6.00.2800.1400" name=3DGENERATOR>
<STYLE></STYLE>
</HEAD>
<BODY bgColor=3D#ffffff>
<DIV><FONT face=3DArial size=3D2>oh.</FONT></DIV>
<BLOCKQUOTE=20
style=3D"PADDING-RIGHT: 0px; PADDING-LEFT: 5px; MARGIN-LEFT: 5px; =

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BORDER-LEFT: #000000 2px solid; MARGIN-RIGHT: 0px">
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  <BR>&gt;<BR>&gt;&lt;!DOCTYPE HTML PUBLIC "-//W3C//DTD HTML 4.0=20
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  http-equiv=3D3DContent-Type content=3D3D"text/html;=20
  =3D<BR>&gt;charset=3D3Diso-8859-1"&gt;<BR>&gt;&lt;META =
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><DIV>=20
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PADDING-LEFT: 5px; MARGIN-LEFT: 5px; =3D
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> href=3D3D"<A=20
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wrote in =3D
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>href=3D3D"<A=20
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R>> =20
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'>mailto:goran.stojiljkovic@os.t-com.hr">goran.stojiljkovic@os.t-co</A=

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I choose Polesoft Lockspam to fight =
spam,=20
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>and=3D20
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<A=3D20
>href=3D3D"<A=20
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href=3D'http://www.polesoft.com/refer.html">http://www.polesoft.com/refer=
'>http://www.polesoft.com/refer.html">http://www.polesoft.com/refer</A=
>=3D
>.html</A>=20
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 </FONT></DIV></BODY&g t;</HTML>
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I can run sessions using eds plugs fine, as soon as I add native plugs it crashes.

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> Irqs is it sharing, are these new projects, post your paris.cfg file,

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>>> Thanks,
>>> Mike

>no wait it's here:

*** PARIS Configuration ***

*** ENGINE configuration parameters ***
* Cache Size in MB
CacheSize=128

* Overview cache size in KB
OvwCacheSize=8192

* I/O configuration
IOSize=256

* SubMix Cache Size in KB
SubMixCacheSize=256

ManualRecDelay=4096

RecXFadeLen=20
Use32BitWinMTC=0

DisableDirectX=0

MasterOutputCard=0

ScrubMaxRate=1

WheelSensitivity=20

WheelInertia=68

CSProVersion=ABCDEFGH

VSTDirectory=C:\vsts\

MIDIPlayDisabled=1

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>>> Thanks,
>>> Mike
>I am using one onboard video card and one PCI, I tried disabling the onboard
but not the PCI, do you think I should try that?
Is that cfg file a general guide or one that works well on XP?

Mike

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Are you sure you followed the XP install exactly. If you get carless installing
the subsystem, bad things can happen. It needs to be installed to the same
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>Thanks,
>MikeIf I installed it in the wrong location would I still see all the plugins?

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>Here Don

http://www.knufinke.de/sir/index_en.html

hit the links page for impulses
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>>>> >
>>>> >
>>>> >
>>>> > Brandon
>>>> >
>>>> >
>>>>
>>>>
>>>
>>>
>>
>
>If you are recording @ 24 bit you really don't need to get that high.
Peaks of -25 to -15 are more than enough. Terry Manning of Compass Point
Studios (AC/DC, ZZ Top etc) turned me on to this on a different forum
and is a big advocate of it. I tried it. I have to agree with him that
it made a significant improvement in the resulting sound of the recording.

http://www.cubase.net/phpbb2/viewtopic.php?t=55258&highl ight=clippingalso on that thread,

Actually I was reading the book lastnight and it stated that a 24bit
recording at -48db is equal to a full range 16bit recording .. mind you,
I was quite drunk lastnight (Twisted Evil) so someone correct me if I

miss-quoted here!

John wrote:
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<alex.plasko@snet.net> wrote:
>hi gene
> I also went to a sonar demo last week, and that 64 bit subject came up.

>according to the rep the sonar audio engine is running in 64 bit mode in

>windows xp 32 bit already. you dont have to have win xp 64.
>I have to admit i was impressed with the improvements they made .most
>notibly the new features like real time audio warping.Its like they are

>giving you vocalign,melodyne,recycle,drumagogg,and a fewother programs
>integrated into sonarV 6.The rep had a mix of a queen tune they were doing
a
>remix on ,and it sounded pretty damn good, even through the PA speakers

>they substituted for monitors.
>I love cubase 4 , but that demo has me thinking again.
>and it supports VST! imagine that.
>Gene Lennon" <glennon@NOSPmyrealbox.com> wrote in message
>news:453f7f7d$1@linux...
>>
>
Hi Alex,
Yes, the 64bit mix engine and the ability to run on 64 bit OS are two separate
issues. I saw it running all 4 ways.

Yes but each of the programs you list work better than the integrated features

is impressive but for drums, Pro Tools Beat Detective is a better approach.

The clever integration of iZotope Radius is perhaps the most exciting feature
for producer/engineers that do a lot of audio manipulation (Like me). Sonar
is certainly impressive. Even if the mix bus comes up short, I still may
consider switching and using external summing. The low cost of the competitive
crossgrade means that I will probably be adding it to my collection even
if I only use it for a few of its tricks.

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>>Hi all

I've been playing with Sawstudiolite and no matter how a configure things I
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A straight flying tracks there and back via lightpitp results in 2055
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Paris now I know Nuendo and Cubase lock to Paris

So my question is...what other apps do the same

ThanksThanks Rod

I thought that was the place

"Rod Lincoln" <rlincoln@nospam.kc.rr.com> wrote in message
news:453fefee$1@linux...
>
> Here Don
>
> http://www.knufinke.de/sir/index_en.html
>
> hit the links page for impulses
> Rod
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>>Dumb Q - where does one get this beast and the impulses
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>>Don
>>
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>>>>> >
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>>>>> >
>>>>> >
>>>>>
>>>>>
>>>>
>>>>
>>>
>>
>>
>This is a multi-part message in MIME format.

------=_NextPart_000_004F_01C6F872.9CF2DD00
Content-Type: text/plain;
 charset="iso-8859-1"
Content-Transfer-Encoding: quoted-printable

Antivirus?=20
Use http://free.grisoft.com/doc/5390/lng/us/tpl/v5#avg-anti-viru s-free

It's always worked perfectly for me, so I have no reason to switch my =
machines off from it.=20
AA

"John" <no@no.com> wrote in message news:453f7707@linux...
>I meant antivirus. sorry
>=20
>=20
> John wrote:
>> Whatcha all think?
>>=20
>> =
 http://www.microsoft.com/athome/security/spyware/software/de fault.mspx
------=_NextPart_000_004F_01C6F872.9CF2DD00
Content-Type: text/html;
 charset="iso-8859-1"
Content-Transfer-Encoding: quoted-printable

<!DOCTYPE HTML PUBLIC "-//W3C//DTD HTML 4.0 Transitional//EN">
<HTML><HEAD>
<META http-equiv=3DContent-Type content=3D"text/html; =

charset=3Diso-8859-1">
<META content=3D"MSHTML 6.00.5730.11" name=3DGENERATOR>
<STYLE></STYLE>
</HEAD>
<BODY>
<DIV><FONT face=3DArial size=3D2>Antivirus? </FONT></DIV>
<DIV><FONT face=3DArial size=3D2>Use <A=20
href=3D" http://free.grisoft.com/doc/5390/lng/us/tpl/v5#avg-anti-viru s-fre=
e"> http://free.grisoft.com/doc/5390/lng/us/tpl/v5#avg-anti-viru s-free</A>=
</FONT></DIV>
<DIV><FONT face=3DArial size=3D2></FONT><FONT face=3DArial =
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<DIV><FONT face=3DArial size=3D2>"John" <</FONT><A =
href=3D"mailto:no@no.com"><FONT=20
face=3DArial size=3D2>no@no.com</FONT></A><FONT face=3DArial =
size=3D2>> wrote in=20
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size=3D2>...</FONT></DIV><FONT face=3DArial size=3D2>>I meant =
antivirus. =20
sorry
>
>
> John wrote:
>> Whatcha all=20
think?
>>
>> </FONT><A=20
href=3D" http://www.microsoft.com/athome/security/spyware/software/de fault=
..mspx"><FONT=20
face=3DArial=20
size=3D2> http://www.microsoft.com/athome/security/spyware/software/de faul=
t.mspx</FONT></A>=20
</BODY></HTML>

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>>>>> Thanks,
>>>>> Mike
>>>
>If Doug can't I can. My fur kids didn't even wait for me to finish it before
they moved in so I must've done something they liked.

AA

"John" <no@no.com> wrote in message news:453f9a3d$1@linux...
> can you build me a doghouse? hehe
>
> Doug Wellington wrote:
>> "Nappy" <mgrant01@san.rr.com> wrote in message news:453f7722$1@linux...
>>> What, no wood working? LOL
>>
>> Heehee, I forgot about that! My martial arts instructor/family doctor
>> bought a bunch of land out in the Chiracahuas and has been building a
>> fortress...I mean house...for some time. He bought some big glass
>> windows
>> from a bank (I think they're bullet-proof or something) and needed a
>> couple
>> custom window frames made, so of course, since everyone knows Doug has a
>> woodshop, they just stacked a bunch of redwood outside the door and
>> waited
>> for me to come home and take care of everything! And this was in the
>> middle
>> of my packrat war too, so the shop floor was covered in turds. BTW, did
>> I
>> mention the little rat bastards had filled my 4" dust collection system
>> with
>> cholla buds and dried dog poo? Oh yeah, remind me to tell you the story
>> of
>> how the packrats had been collecting the dog doo and putting it in a pile
>> just on the other side of my backyard wall. (And here I thought the kids
>> were being good...)
>>
>> Anyway, the windows were the same height, but about an inch and a half
>> different in width. Since they were going to be put into two openings of
>> similar size, I made one of the frames with deeper slots for the longer
>> piece of glass. Well, they forgot to test fit everything, so they ended
>> up
>> assembling the small glass into the big frame. Then, when they tried to
>> put
>> the big glass into the small frame, they realized what they had done and
>> had
>> to spend a couple hours prying everything back apart!!! Sheesh!!!
>>
>> Never dull,
>> Doug
>>
>> http://www.parisfaqs.com
>>

>>

Have you tried putting a SMPTE stripe in to read the playback time code?
AA

"Don Nafe" <dnafe@magma.ca> wrote in message news:4540085c$1@linux...
> Hi all
>
> I've been playing with Sawstudiolite and no matter how a configure things
> I can't achieve accurate sync with Paris when tracks are loaded into
> Saw...
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> A straight flying tracks there and back via lightpitp results in 2055
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> wanders all over the place. Bob the developer seems to think this is a
> result of the MTC sync my Dakota card is generating from the ADAT sync in
> Paris now I know Nuendo and Cubase lock to Paris
>
> So my question is...what other apps do the same
>
> Thanks
>Thanks, that will be my next move, I didn't notice if was in the correct folder.
I assumed it would put it there.
I appreciate your help.

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But Gene, that's a PC program! Oh yeah, that's right, you got one of them
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James"James McCloskey" <excelsm@hotmail.com> wrote:

>But Gene, that's a PC program! Oh yeah, that's right, you got one of them
>newfangled Macs;)
>
>James
>

I always ran Paris on a PC.

GeneIt's a 1.5v AGP mobo. The GEForce card is 3.3v. Now I wonder if the reason
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electrical engineering course.....I'm gonna get a gimme hat with "dumbass"
in big block letters.

(sigh)

;o}How do you wrap your VST plugs ?
(newbee style question)

4540087e@linux...
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> I thought that was the place
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> How do you wrap your VST plugs ?
> (newbee style question)

Well, you download:

http://www.parisfaqs.com/wrapper33.zip

Extract the files and follow the instructions in the Readme33.txt
file...

Doug

http://www.parisfaqs.comThis depends on PCI compatibility versions, but...

I was under the impression that all but the very last version of PCI would
accept 3.3v cards and run them at 3.3v. The final version of PCI (2.3???)
doesn't, but 3.3v cards don't fit in it as I recall.

I wouldn't have thought a 3.3v card would even run at 1.5...

Cheers,
Kim.

"DJ" <notachance@net.net> wrote:
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>(sigh)
>
>;o}
>
>Can that, you're talking about AGP...

....love has my brain frazzled at the moment. ;o)

Cheers,
Kim.

"Kim" <hiddensounds@hotmail.com> wrote:
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 >

I choose Polesoft Lockspam to fight spam, and you?
http://www.polesoft.com/refer.html
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</HEAD>
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<DIV><FONT face=3DArial size=3D2>Kim,</FONT></DIV>
<DIV><FONT face=3DArial size=3D2>Hopefully more than just your =
brain.</FONT></DIV>
<DIV><FONT face=3DArial size=3D2>Tom</FONT></DIV>
<BLOCKQUOTE=20
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<DIV>"Kim" <<A=20
=
href=3D"mailto:hiddensounds@hotmail.com">hiddensounds@hotmail.com</A>>=
wrote=20
in message <A=20
=
href=3D"news:45405600$1@linux">news:45405600$1@linux</A>...</DIV>

=
Can=20
that, you're talking about AGP...

...love has my brain frazzled =
at the=20
moment. ;o)

Cheers,
Kim.

"Kim" <<A=20
=
href=3D"mailto:hiddensounds@hotmail.com">hiddensounds@hotmail.com</A>>=
=20
wrote:
>
>
>This depends on PCI compatibility =
versions,=20
but...
>
>I was under the impression that all but the very =
last=20
version of PCI would
>accept 3.3v cards and run them at 3.3v. =
The final=20
version of PCI (2.3???)
>doesn't, but 3.3v cards don't fit in it =
as I=20
recall.
>
>I wouldn't have thought a 3.3v card would even =
run at=20
1.5...
>
>Cheers,
>Kim.
>
>"DJ" <<A=20
href=3D"mailto:notachance@net.net">notachance@net.net</A>>=20
wrote:
>>It's a 1.5v AGP mobo. The GEForce card is 3.3v. Now =
I wonder=20
if the reason
>>I'm having all of these other problems is =
because I=20
fried the mobo (and/or
>>the video cards). would running the =
cards=20
undervoltage ruin them? this
could
>>explain some other =
wierdness.=20
I had those 3.3v cards running great in the
>>Paris mobo for =
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>>hope =

the=20
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 an
>>electrical engineering course.....I'm gonna get a gimme =
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letters.
>>
>>(sigh)
>>
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>=
>
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>
</BLOCKQUOTE >
<DIV><FONT size=3D2>

I choose Polesoft Lockspam to fight spam, =
and=20
you?
<A=20
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..html</A>   </FONT></DIV></BODY ></HTML>

------=_NextPart_000_00C2_01C6F8B1.A16B6E30--Hi,
Anyone knows if these waves devices work with Paris ?
Regards,
DimitriosHey Aaron

I'll be attempting this today, hopefully with better results

DOn

"Aaron Allen" <know-spam@not_here.dude> wrote in message
news:45400cfb@linux...
> Have you tried putting a SMPTE stripe in to read the playback time code?
> AA
>
> "Don Nafe" <dnafe@magma.ca> wrote in message news:4540085c$1@linux...
>> Hi all
>>
>> I've been playing with Sawstudiolite and no matter how a configure things
>> I can't achieve accurate sync with Paris when tracks are loaded into
>> Saw...
>>
>> A straight flying tracks there and back via lightpitp results in 2055
>> samples of latency everytime, but record or drop a track into it and it
>> wanders all over the place. Bob the developer seems to think this is a
>> result of the MTC sync my Dakota card is generating from the ADAT sync in
>> Paris now I know Nuendo and Cubase lock to Paris
>>
>> So my question is...what other apps do the same
>>
>> Thanks
>>
>
>

>Hey.....that's OK...you've given me an idea......now if I can just cram one
of these AGP cards into a PCI slot.....

;o)

"Kim" <hiddensounds@hotmail.com> wrote in message news:45405600$1@linux...
>
>
> Can that, you're talking about AGP...
>
> ...love has my brain frazzled at the moment. ;o)
>
> Cheers,
> Kim.
>
> "Kim" <hiddensounds@hotmail.com> wrote:
> >
> >
> >This depends on PCI compatibility versions, but...
> >
> >I was under the impression that all but the very last version of PCI
would
> >accept 3.3v cards and run them at 3.3v. The final version of PCI (2.3???)
> >doesn't, but 3.3v cards don't fit in it as I recall.
> >
> >I wouldn't have thought a 3.3v card would even run at 1.5...
> >
> >Cheers,
> >Kim.
> >
> >"DJ" <notachance@net.net> wrote:
> >>It's a 1.5v AGP mobo. The GEForce card is 3.3v. Now I wonder if the
reason
> >>I'm having all of these other problems is because I fried the mobo
(and/or
> >>the video cards). would running the cards undervoltage ruin them? this
> could
> >>explain some other wierdness. I had those 3.3v cards running great in
the
> >>Paris mobo for about 3 days before things started going south.
CRAP!!!!....I
> >>hope the video cards and the mobo aren't all damaged. I need to take an
> >>electrical engineering course.....I'm gonna get a gimme hat with
"dumbass"
> >>in big block letters.
> >>
> >>(sigh)
> >>

> >>;o}
> >>
> >>
> >
>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.
>>
>use the dremel and cut a new slot. works every time!

"DJ" <notachance@net.net> wrote in message news:4540c227@linux...
> Hey.....that's OK...you've given me an idea......now if I can just cram
> one
> of these AGP cards into a PCI slot.....
>
> ;o)
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> "Kim" <hiddensounds@hotmail.com> wrote in message news:45405600$1@linux...
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>Just a little curious. I dont recall seeing a count on our beloved news
group here as to how many hit records, or at least ones that charted ,were
recorded with paris.
we all know about BT and the Lonestar track.
1)How many of us worldclass engineers have actually hit paydirt using paris?
2)Has anyone researched the top system(s) used for said hit records?
I dont want to hear the hype. just the facts, if anyone knows.
just curious guys,(and girls) no need to start a flame fest here.:-)Alex, Joshua Thompson whom I, as well as Genne Lennon worked with has had
a string of hits using Paris. He introduced me to Paris when I worked in
his camp from around 2000 to 2002.

Tyrone

with artists ranging from "alex plasko" <alex.plasko@snet.net> wrote:
>Just a little curious. I dont recall seeing a count on our beloved news

>group here as to how many hit records, or at least ones that charted ,were

>recorded with paris.
>we all know about BT and the Lonestar track.
>1)How many of us worldclass engineers have actually hit paydirt using paris?
>2)Has anyone researched the top system(s) used for said hit records?
>I dont want to hear the hype. just the facts, if anyone knows.
>just curious guys,(and girls) no need to start a flame fest here.:-)
>
>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.
>>>
>>
>"alex plasko" <alex.plasko@snet.net> wrote:
>use the dremel and cut a new slot. works every time!

Nah Deej, just pull out the ole trusty the SAWZAL!

TyroneBasically this inolved strapping this across every track in a mix, applying

a UAD-1 Delaycomp on the first slot in the application and then adding UAD-1
and other plugins to the subsequent slots. The thing that killed this idea
was that in order for it to work, it had to be used on *every* track so that
there was a uniform amount of delay compensaion. then it was just a matter
of sliding "all" of the tracks to the left in the Paris editor to the left
by a certain amount to cover the buffer latency of the host machine.

Well....there are a few of these host applications........sooooo.........
Chainer will allow access to up to 10 x ASIO I/O.
FXPansion Simple Virtual Host will allow access to 4 x ASIO I/O
Forte, for my purposes, would allow access to 10 x ASIO I/O
Steinberg VStack will allow access to 16 ASIO I/O..
RT player will allow access to a few more ASIO I/O....

So it appears that using all of these on the same machine, I could, "in
theory" access *at least* 40 ASIO* I/O and that's all I would need for a
real time mix scenario.

Now assuming I was running all five of these on the same system
sending/returning signal in and out of 40 RME ADAT I/O whil'st processing
these signals through 4 x UAD-1 cards (and other VSTi's) with a UAD-1 delay
comp instantiated in the first slot of each host set ot compensate for 4 x
plugins and that all of these VST hosts had a predictable latency
...........well.....you know where I'm going with this, don't you?

;o)Didn't Jason Miles win a grammy not long ago?

"alex plasko" <alex.plasko@snet.net> wrote in message news:4540cd94@linux...
> Just a little curious. I dont recall seeing a count on our beloved news
> group here as to how many hit records, or at least ones that charted ,were
> recorded with paris.
> we all know about BT and the Lonestar track.
> 1)How many of us worldclass engineers have actually hit paydirt using
> paris?
> 2)Has anyone researched the top system(s) used for said hit records?
> I dont want to hear the hype. just the facts, if anyone knows.
> just curious guys,(and girls) no need to start a flame fest here.:-)
>This is a multi-part message in MIME format.

---=_linux4540d2c9
Content-Type: text/plain
Content-Transfer-Encoding: 7bit

"Tyrone Corbett" <tyronecorbett@comcast.net> wrote:
>

>"alex plasko" <alex.plasko@snet.net> wrote:
>>use the dremel and cut a new slot. works every time!
>
>Nah Deej, just pull out the ole trusty the SAWZAL!
>
>Tyrone
>
>
>
This is DJ !!
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pbqerjvfzPHd3p+q8rDOjmTCkRgrSX21thXTbWkpv8dBARNnqaS4hySFIdxg
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8NBEObGU5CagmUPRGH33GUaDdLT6SA30/wAJUTeg9RNnS4mPxcdmW2Xsa+4+
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qUfzKYYOez95ftSn6rylAoFAoFAoFAoFAoFAoFAoFAoFAoP/Z

---=_linux4540d2c9--So, if I've got VST effects working on my Paris system, should I bother trying to wrap them? I've noticed that things like meters on VST effects don't work, but the effect still seems to. Does the wrapper make the dancy lights and stuff work on VST effects?

What are the preferred options for general wrapping of VSTs?

Sorry for the dumb questions, I'm just trying to get my head around it.

-scott v.;o)....seriously.....do you think I could have damaged the mobo by running a 3.3v card in a 1.5v slot...and/or damaged the video card?

"Gene Lennon" <glennon@NOSPmyrealbox.com> wrote in message news:4540d2c9$1@linux...
>
> "Tyrone Corbett" <tyronecorbett@comcast.net> wrote:
> >
> >"alex plasko" <alex.plasko@snet.net> wrote:
> >>use the dremel and cut a new slot. works every time!
> >
> >Nah Deej, just pull out the ole trusty the SAWZAL!
> >
> >Tyrone
> >
> >

> >
> This is DJ !!ya...you're returning all your new purchases and ordering a straight jacket

:-)

"DJ" <notachance@net.net> wrote in message news:4540d282@linux...
> Basically this inolved strapping this across every track in a mix,
> applying
> a UAD-1 Delaycomp on the first slot in the application and then adding
> UAD-1
> and other plugins to the subsequent slots. The thing that killed this idea
> was that in order for it to work, it had to be used on *every* track so
> that
> there was a uniform amount of delay compensaion. then it was just a matter
> of sliding "all" of the tracks to the left in the Paris editor to the left
> by a certain amount to cover the buffer latency of the host machine.
>
> Well....there are a few of these host applications........sooooo.........
> Chainer will allow access to up to 10 x ASIO I/O.
> FXPansion Simple Virtual Host will allow access to 4 x ASIO I/O
> Forte, for my purposes, would allow access to 10 x ASIO I/O
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> So it appears that using all of these on the same machine, I could, "in
> theory" access *at least* 40 ASIO* I/O and that's all I would need for a
> real time mix scenario.
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> Now assuming I was running all five of these on the same system
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> these signals through 4 x UAD-1 cards (and other VSTi's) with a UAD-1
> delay
> comp instantiated in the first slot of each host set ot compensate for 4 x
> plugins and that all of these VST hosts had a predictable latency
> ....well.....you know where I'm going with this, don't you?
>
> ;o)
>
>
>"Tom Bruhl" <arpegio@comcast.net> wrote:
>Kim,
>Hopefully more than just your brain.

....well, one would think, but apparently not in this case...A Love Affair....the music of Ivan Lins...the song was She Walks This Earth.
Sting on lead vocal. Excellent CD. I think this won a Grammy in 2000-01. He

may have won other Grammys as well. I think he still uses Paris. Any hits by Markus Miller are likely to involve a Paris system in the production as well.

"Don Nafe" <dnafe@magma.ca> wrote in message news:4540d2c3@linux...
> Didn't Jason Miles win a grammy not long ago?
>
>
> "alex plasko" <alex.plasko@snet.net> wrote in message
news:4540cd94@linux...
> > Just a little curious. I dont recall seeing a count on our beloved news
> > group here as to how many hit records, or at least ones that charted
,were
> > recorded with paris.
> > we all know about BT and the Lonestar track.
> > 1)How many of us worldclass engineers have actually hit paydirt using
> > paris?
> > 2)Has anyone researched the top system(s) used for said hit records?
> > I dont want to hear the hype. just the facts, if anyone knows.
> > just curious guys,(and girls) no need to start a flame fest here.:-)
> >
>
>"Don Nafe" <dnafe@magma.ca> wrote:
>ya...you're returning all your new purchases and ordering a straight jacket
>
>:-)
>

I almost cried when I read that, LOL!

James

>
>"DJ" <notachance@net.net> wrote in message news:4540d282@linux...
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>Well....VStack doesn't pass audio...just outputs it so it's out anyway. The
developer of the DSound has sent me a few e-mails asking what on earth I am
trying to do.....so I told him and now he is sitting over in Europe
somewhere laughing at the crazy American.

"James McCloskey" <excelsm@hotmail.com> wrote in message
news:4540e6d7$1@linux...
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>somewhere laughing at the crazy American.

But did you tell him that the DAW is called Paris, so it should work.

James

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>> >>
>> >> ;o)
>> >>
>> >>
>> >>
>> >
>> >
>>
>
>I'm sure I'll hear back from him sooooonnnnn........

"james McCloskey" <excelsm@hotmail.com> wrote in message
news:4540e982$1@linux...
>
> "DJ" <notachance@net.net> wrote:
> >Well....VStack doesn't pass audio...just outputs it so it's out anyway.
> The
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> >> >> ....well.....you know where I'm going with this, don't you?
> >> >>
> >> >> ;o)
> >> >>
> >> >>
> >> >>
> >> >
> >> >
> >>
> >
> >
>This is a multi-part message in MIME format.

------=_NextPart_000_0035_01C6F90A.D2CBD240
Content-Type: text/plain;
 charset="iso-8859-1"
Content-Transfer-Encoding: quoted-printable

Speaking of SSL, have you guys seen this new piece , I wonder how this =
would work with Paris latency speeking..??
http://www.solid-state-logic.com/music/duende_home.html

Rob

"Tom Bruhl" <arpegio@comcast.net> wrote in message =
news:453fe0a4@linux...
oh.
"Rod Lincoln" <rlincoln@nospam.kc.rr.com> wrote in message =
news:453fd252$1@linux...

If it's a stool sample...........well...........never mind.
Rod
"Tom Bruhl" <arpegio@comcast.net> wrote:
>
>
>What's a sample among friends?
>T.
>
> "Dimitrios" <musurgio@otenet.gr> wrote in message =3D
>news:453f8b69$1@linux...
>
> Hi,
> The SSLcompressor has 0 latency.
> Thee SSL channel has 1 sample latency.
> Regards,
> Dimitrios
>
> "LaMont" <jjdpro@ameritech.net> wrote:
> >
> >yes it works. Thank god Waves still code their plugins in =
Direct-x. =3D
>Surprisely,
> >the plug added no latency. However, the mix I was working on =
only had
=3D
>10
> >tracks.=3D20
> >
> >"Goran Stojiljkovic" <goran.stojiljkovic@os.t-com.hr> wrote:
> >>does it work?
> >>latency ?
> >>
> >>please answer......=3D20
> >>
> >>
> >
>
>
>
>I choose Polesoft Lockspam to fight spam, and you?
>http://www.polesoft.com/refer.html =20
>

><!DOCTYPE HTML PUBLIC "-//W3C//DTD HTML 4.0 Transitional//EN">
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>charset=3D3Diso-8859-1">
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></HEAD>
><BODY bgColor=3D3D#ffffff>
><DIV><FONT face=3D3DArial size=3D3D2>What's a sample among =3D
>friends?</FONT></DIV>
><DIV><FONT face=3D3DArial size=3D3D2>T.</FONT></DIV>
><DIV> </DIV>
><BLOCKQUOTE=3D20
>style=3D3D"PADDING-RIGHT: 0px; PADDING-LEFT: 5px; MARGIN-LEFT: 5px; =
=3D
>BORDER-LEFT: #000000 2px solid; MARGIN-RIGHT: 0px">
> <DIV>"Dimitrios" <<A=3D20
> href=3D3D"mailto:musurgio@otenet.gr">musurgio@otenet.gr</A>> =
wrote in =3D
>message=3D20
> <A=3D20
> =3D
=
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H=
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>"Goran =3D
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=
>href=3D3D"mailto:goran.stojiljkovic@os.t-com.hr">goran.stojiljkovic@os.t=
-co=3D
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>>
>>
>
</BLOCKQUOTE>

><DIV><FONT size=3D3D2>

I choose Polesoft Lockspam to fight =
spam, =3D
>and=3D20
>you?
<A=3D20
=
>href=3D3D"http://www.polesoft.com/refer.html">http://www.polesoft.com/re=
fer=3D
>.html</A> </FONT></DIV></BODY></HTML>
>
>

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Content-Transfer-Encoding: quoted-printable

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charset=3Diso-8859-1">
<META content=3D"MSHTML 6.00.2900.2873" name=3DGENERATOR>
<STYLE></STYLE>
</HEAD>
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ww.solid-state-logic.com/music/duende_home.html</A></FONT> </DIV>
<DIV><FONT size=3D2></FONT> </DIV>
<DIV><FONT size=3D2>Rob</FONT></DIV>
<DIV><FONT size=3D2></FONT> </DIV>
<DIV><FONT size=3D2></FONT> </DIV>
<BLOCKQUOTE dir=3Dltr=20
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wrote:
> >>does it work?
> =
>>latency=20
?
> >>
> >>please=20
answer......=3D20
> >>
> =
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> =20
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>
>
>I choose Polesoft Lockspam to =
fight=20
spam, and you?
>http://www.polesoft.com/refer.html   =

>
><!DOCTYPE HTML PUBLIC "-//W3C//DTD HTML 4.0=20
=
Transitional//EN">
><HTML><HEAD>
><META=20
http-equiv=3D3DContent-Type content=3D3D"text/html;=20
=3D
>charset=3D3Diso-8859-1">
><META =
content=3D3D"MSHTML=20
6.00.2800.1400"=20
=

name=3D3DGENERATOR>
><STYLE></STYLE>
></HEA=
D>
><BODY=20
bgColor=3D3D#ffffff>
><DIV><FONT face=3D3DArial=20
size=3D3D2>What's a sample among=20
=
=3D
>friends?</FONT></DIV>
><DIV><FONT=20
face=3D3DArial =
size=3D3D2>T.</FONT></DIV&gt ;
><DIV>=20
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</DIV>
><BLOCKQUOTE=3D20
>style=3D3D"PADDING-RIGHT: =
0px;=20
PADDING-LEFT: 5px; MARGIN-LEFT: 5px; =3D
>BORDER-LEFT: #000000 =
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solid; MARGIN-RIGHT: 0px">
> <DIV>"Dimitrios"=20
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> href=3D3D"<A=20
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href=3D'mailto:musurgio@otenet.gr">musurgio@otenet.gr</A'>mailto:musurgio=
@otenet.gr">musurgio@otenet.gr</A</A>>>=20
wrote in =3D
>message=3D20
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>href=3D3D"<A=20
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href=3D'news:453f8b69$1@linux">news:453f8b69$1@linux</A>...</DIV>
Hi'>=
news:453f8b69$1@linux">news:453f8b69$1@linux</A>...</DIV>&=
lt;BR>Hi</A>,<=3D
>BR>The=3D20
> =20
SSLcompressor has 0 latency.
Thee SSL channel has 1=20
sample=3D20
> =20
=
latency.
Regards,
Dimitrio s

 "LaMont"=
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> href=3D3D"<A=20
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href=3D'mailto:jjdpro@ameritech.net">jjdpro@ameritech.net</A>>=3D20'>mail=
to:jjdpro@ameritech.net">jjdpro@ameritech.net</A>>=3D20</A><B=
R>> =20
wrote:
>
>ye s it works. Thank god Waves =
still code=20
their =3D
>plugins=3D20
> in Direct-x.=20
Surprisely,
>the plug added no latency. However, =
=3D
>the=20
mix I=3D20
> was working on only had =
10
>tracks.=20

>
>"Goran =
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> =3D
>href=3D3D"<A=20
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'>mailto:goran.stojiljkovic@os.t-com.hr">goran.stojiljkovic@os.t-co</A=

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>m.hr</A>>=3D20
> =20
wrote:
>>does it=20
work?
>>latency=3D20
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?
>>
&gt ;>please =
answer......=3D20
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< /BLOCKQUO=
TE>
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size=3D3D2>

I choose Polesoft Lockspam to =
fight spam,=20
=
=3D
>and=3D20
>you?
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href=3D'http://www.polesoft.com/refer.html">http://www.polesoft.com/refer=
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>.html</A>=20
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</FONT></DIV></BODY&g t;</HTML>
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BLOCKQUOTE></BLOCKQUOTE></BODY></HTML>

------=_NextPart_000_0035_01C6F90A.D2CBD240--and the check is in the mail
"DJ" <notachance@net.net> wrote in message news:4540ec1a@linux...
> I'm sure I'll hear back from him sooooonnnnn........
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>> >> >
>> >>
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>> >> >> ;o)
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>> >> >>
>> >> >
>> >> >
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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
> 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

>> 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.
>>>
>>
>;o)

"alex plasko" <alex.plasko@snet.net> wrote in message
news:4540f566$1@linux...
> and the check is in the mail

> "DJ" <notachance@net.net> wrote in message news:4540ec1a@linux...
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> >> >> >>

> >> >> >>
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>
>Maybe I need to get up off my ass and learn to do this. My needs are too outside the box to expect to find them commercially. Here's the tool kit.

http://dssi.sourceforge.net/why-use.html

Hell....I've got the guy who wrote the code for MRI machines here to help me. His wife is one of my studio clients. His brother-in- law is my partner. This can definitely be done and I'd love to learn how to write my own stuff.

Once I get the studio back up and running I'm going to try to find the time to write a VST FX rack that can access unlimited I/O and plugin slots........I'm going to talk to Dan about this ASAP.

Deej

"DJ" <notachance@net.net> wrote in message news:4540f901@linux...
> ;o)
>
> "alex plasko" <alex.plasko@snet.net> wrote in message
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> > >

> >
> >
>
>This is a multi-part message in MIME format.

------=_NextPart_000_0048_01C6F90B.9FE55080
Content-Type: text/plain;
 charset="iso-8859-1"
Content-Transfer-Encoding: quoted-printable

Scott,
Yes the wrapper will fix the GUI of most plugs. My UADs are still
showing meters with latency though.
Tom
  "volthause" <volthause-nospam-@soldrocks-nospam-.com> wrote in message =
news:Xns986875884AFEBvolthause@202.63.37.102...
  So, if I've got VST effects working on my Paris system, should I =
bother=20
  trying to wrap them? I've noticed that things like meters on VST =
effects=20
  don't work, but the effect still seems to. Does the wrapper make the =
dancy=20
  lights and stuff work on VST effects?

  What are the preferred options for general wrapping of VSTs?

  Sorry for the dumb questions, I'm just trying to get my head around =
it.

  -scott v.

I choose Polesoft Lockspam to fight spam, and you?
http://www.polesoft.com/refer.html
------=_NextPart_000_0048_01C6F90B.9FE55080
Content-Type: text/html;
 charset="iso-8859-1"
Content-Transfer-Encoding: quoted-printable

<!DOCTYPE HTML PUBLIC "-//W3C//DTD HTML 4.0 Transitional//EN">
<HTML><HEAD>
<META http-equiv=3DContent-Type content=3D"text/html; =
charset=3Diso-8859-1">
<META content=3D"MSHTML 6.00.2800.1400" name=3DGENERATOR>
<STYLE></STYLE>
</HEAD>
<BODY bgColor=3D#ffffff>
<DIV><FONT face=3DArial size=3D2>Scott,</FONT></DIV>

<DIV><FONT face=3DArial size=3D2>Yes the wrapper will fix the GUI of =
most=20
plugs. My UADs are still</FONT></DIV>
<DIV><FONT face=3DArial size=3D2>showing meters with latency =
though.</FONT></DIV>
<DIV><FONT face=3DArial size=3D2>Tom</FONT></DIV>
<BLOCKQUOTE=20
style=3D"PADDING-RIGHT: 0px; PADDING-LEFT: 5px; MARGIN-LEFT: 5px; =
BORDER-LEFT: #000000 2px solid; MARGIN-RIGHT: 0px">
<DIV>"volthause" <<A=20
=
href=3D"mailto:volthause-nospam-@soldrocks-nospam-.com">volthause-nospam-=
@soldrocks-nospam-.com</A>>=20
wrote in message <A=20
=
href=3D"news:Xns986875884AFEBvolthause@202.63.37.102">news:Xns986875884AF=
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I choose Polesoft Lockspam to fight spam, =
and=20
you?
<A=20
href=3D"http://www.polesoft.com/refer.html">http://www.polesoft.com/refer=
..html</A>   </FONT></DIV></BODY ></HTML>

------=_NextPart_000_0048_01C6F90B.9FE55080--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
> actually get the kind of mixes I hear in my head out of DP.

>
> Tony
>If I could get someone to help me code a very simple VST rack like Forte, would it be possible to run it outside of Paris as an independent application and have it interface with Paris using Wires? Since Paris requires minimal host processing power and UAD-1/Powercore/Duende/LiquidMix, etc. all have their won DSP engines, would it be possible to take some of the available native CPU horsepower and apply a goodly amount of it it to knocking down the latency of these plugins to a low, as in maybe zero and if not zero, then at least a predictable number of samples which could correspond exactly to the nudge parameters in the Paris editor and then connect the channels of this external VST rack to Paris inserts and auxes using Wires? I'm serious here man. Do you think this would be possible? I've got a guru here whose wife wants to do a project at this studio. I can trade session time for R & D time....I'm absolutely sure of it.

DeejSMPTE was a no go...sending smpte to dakota results in it converting it to MTC which is what Sawstudio is having problems reading and I have no other way of reading smpte on my second rig although all ideas are welcome

DOn

"Don Nafe" <dnafe@magma.ca> wrote in message news:45409ec1$1@linux...
> Hey Aaron
>
> I'll be attempting this today, hopefully with better results
>
> DOn
>
>
> "Aaron Allen" <know-spam@not_here.dude> wrote in message
> news:45400cfb@linux...
>> Have you tried putting a SMPTE stripe in to read the playback time code?
>> AA
>>
>> "Don Nafe" <dnafe@magma.ca> wrote in message news:4540085c$1@linux...
>>> Hi all
>>>
>>> I've been playing with Sawstudiolite and no matter how a configure
>>> things I can't achieve accurate sync with Paris when tracks are loaded
>>> into Saw...
>>>
>>> A straight flying tracks there and back via lightpitp results in 2055
>>> samples of latency everytime, but record or drop a track into it and it
>>> wanders all over the place. Bob the developer seems to think this is a
>>> result of the MTC sync my Dakota card is generating from the ADAT sync
>>> in Paris now I know Nuendo and Cubase lock to Paris

>>>
>>> So my question is...what other apps do the same
>>>
>>> Thanks
>>>
>>
>>
>>
>
>Hey Deej

Not to throw a damper on things but isn't wires, like ADATs (in XP) unable
to cross submixes and isn't that an inherent part of Paris' mixing
architecture?

Wouldn't this be somewhat like re-route or rewire or whatever that thing is
called in terms of routing audio to various points inside a different
application

Inquiring minds want to know

"DJ" <notachance@net.net> wrote in message news:45411808@linux...
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> application and have it interface with Paris using Wires? Since Paris
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> got a guru here whose wife wants to do a project at this studio. I can
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> session time for R & D time....I'm absolutely sure of it.
>
> Deej
>
>
>This is a question for Chuck. I've never used Wires. However, even it wasn't
possible to cross submixes, if it was possible to create an FX rack applet
per submix and allocate native DSP across these, that would work for me.

Deej

"Don Nafe" <dnafe@magma.ca> wrote in message news:45411ff8$1@linux...
> Hey Deej
>
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> to cross submixes and isn't that an inherent part of Paris' mixing
> architecture?
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> >
> > Deej
> >
> >
> >
>
>Two weeks ago we almost had you recording into SX, and now this? Talk about
a relapse . . .

TCB

"DJ" <notachance@net.net> wrote:
>Basically this inolved strapping this across every track in a mix, applying
>a UAD-1 Delaycomp on the first slot in the application and then adding UAD-1
>and other plugins to the subsequent slots. The thing that killed this idea
>was that in order for it to work, it had to be used on *every* track so
that
>there was a uniform amount of delay compensaion. then it was just a matter
>of sliding "all" of the tracks to the left in the Paris editor to the left
>by a certain amount to cover the buffer latency of the host machine.
>
>Well....there are a few of these host applications........sooooo.........
>Chainer will allow access to up to 10 x ASIO I/O.
>FXPansion Simple Virtual Host will allow access to 4 x ASIO I/O
>Forte, for my purposes, would allow access to 10 x ASIO I/O
>Steinberg VStack will allow access to 16 ASIO I/O..
>RT player will allow access to a few more ASIO I/O....
>
>
>So it appears that using all of these on the same machine, I could, "in
>theory" access *at least* 40 ASIO* I/O and that's all I would need for a
>real time mix scenario.
>
>Now assuming I was running all five of these on the same system
>sending/returning signal in and out of 40 RME ADAT I/O whil'st processing
>these signals through 4 x UAD-1 cards (and other VSTi's) with a UAD-1 delay
>comp instantiated in the first slot of each host set ot compensate for 4
x
>plugins and that all of these VST hosts had a predictable latency
>..........well.....you know where I'm going with this, don't you?
>
>;o)
>
>
>Thanks, i'll try.

ADA20DEF-64BC-11DB-96B0-000393A9F344%doug@parisfaqs.com...

>> How do you wrap your VST plugs ?
>> (newbee style question)
>
> Well, you download:
>
> http://www.parisfaqs.com/wrapper33.zip

>
> Extract the files and follow the instructions in the Readme33.txt
> file...
>
> Doug
>
> http://www.parisfaqs.com
>Hey DJ,
Do you know that Pulsar gives you 64 routes to and from asio destination/sources
??
Soyou can route 64 audio tracks back and forth inside pulsar from VST host
using asio and the transfer to paris.
Regards,
Dimitrios

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>Deej
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>
>I am also hearing that it doesn't work with AMD dualcore processors. Have
you tried this dimitrios?

"Dimitrios" <musurgio@otenet.gr> wrote in message news:45413238$1@linux...
>
> Hey DJ,
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>Dimitrios......I don't want to use a VST host, if the vst host is cubase SX.
If it is the Pulsar mixer than that would be OK, but I want to get as far
away from Cubase as I can. If I wanted to use Cubase, I could use the RME
cards I have now.

Thanks,

DJ

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>This is what I have for sale in EXCELLENT shape. Let's move it out !
Shipping $10.

eds 150
eds 150
c16 75
8in 200
8in 200
8out 200
mec 75

Email me or call
John
john@kfocus.com
843-559-3777 evesDJ,
I don't mean Cubase.
Pulsar can receive up to 64 ASIO destinations and send on up to 64 =ASIO
sends.
So any vst chainer/host or whatever you call it that can load vst's and output
on different asio can do the trick.
I have used thru cubase (well...) sending 32 audio tracks back and forth
via pulsar.
You can use 32 bit floating ,32 bit integer, 24 bit asio devices from within
pulsar !

"DJ" <notachance@net.net> wrote:
>Dimitrios......I don't want to use a VST host, if the vst host is cubase
SX.

>If it is the Pulsar mixer than that would be OK, but I want to get as far
>away from Cubase as I can. If I wanted to use Cubase, I could use the RME
>cards I have now.
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>Thanks,
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>> >
>> >
>> >
>>
>

>I'll take an EDS, an 8-in & the MEC. Seriously.

(going to do my own damn summing experiments &
comparisons! :D   )

Reserve those for me, please - I'll follow up with an e-mail
this evening to arrange for payment & shipping.

Neil

John <no@no.com> wrote:
>This is what I have for sale in EXCELLENT shape. Let's move it out !
>Shipping $10.
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>eds 150
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>
>Email me or call
>John
>john@kfocus.com
>843-559-3777 evesI never used AMD in my life !
No problem with my intel comoputers.
Used ASUS P3BF , ASUS P4B, Asus P4B-E, Abit BH-6
all 440 chipset work great.
These are dead cheap.
But on planetz forum you can search for newer working pc's
I am sure new models work as great.

"DJ" <notachance@net.net> wrote:
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>you tried this dimitrios?
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>away from Cubase as I can.

WTF? As far away??? Why this change in attitude?

NeilHi,
Even with all these high end shits , mic preamps ad converters microphones
sometimes you get amazed by some folks using "cheap" alternatives like Yamaha
01v or AW4416 workstations.
I heard some drums and I heard "that" sound on the snare I am looking for
, crisp punchy and with lot of harmonics.
Like "californication" song ,you know what snare I mean.
I believe they used O2 on that song, not sure.

So as a cheap alternative would I get the same sound using DS2416 dsp factory
with analog ins and adat out ?
I amsure it is the yamaha converters and not probably the mic preamps.
If it is also mic preamps (I wish not) then I could get me a cheap old 01
(not V) and use analog outs to DS2416 analog in (DS 2416 has same internal
routings like 02) and then an adat out card to paris or pulsar.
What do you think ??
Regards,
DimitriosDJ,
To help you understand.
Pulsar lets you alter its ASIO routing to achieve that amazing 64 in out
asio routing !!
Appears as asio 1,2 3,4 5,6 etc.
Now ANY asio related host appchainer will showon its routing when selecting
Scope asio these exact ins and outs.
Thats exactly what I was using all that years well with cubase.
If cubase 5 did that for me I am sure you can use like chainer to compliment
that routing.
So why bother with wires when Pulsar comes to rrescue ?
Can you imagine the possibiolties ?
If you can run Pulsar at 3 ms (why not I could) then using the "millidelay"
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nudge intervals (80 samples 160 samples etc)
Hope this helps.
Regards,
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>consider them reserved

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What I want to do is as follows:

Create a Paris mix template with 40 tracks with each track having an insert inabled and routed to that track in the Paris virtual patchbay.

I will have 40 ADAT inputs and outputs routed between Paris and the Pulsar cards.

Now I want to be able to send my Paris tracks to the Pulsar mixer, process them there with Pulsar plugins and Uad-1 plugins at a certain fixed latency that I can compensate each track in Paris, and then return them to Paris without having to use Cubase SX as a VST host at all.

Is this possible?

Thanks,

DJ

"Dimitrios " <musurgio@otenet.gr> wrote in message news:45413977@linux...
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> >Deej
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>DJ,
You can do that of course but you will have to use a chainer VST loader like chainer or forte , whatever that will load the vsts take input from scope asio sends and send back thru scope asio again back to pulsar mixer.
Regards,
Dimitrios

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>> >
>> >
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>>
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>So since neither Forte or Chainer will allow more than 16 I/O if it is used
as a standalone application I can open up Forte or chainer on each of the 40
Pulsar channels and this will allow it to see all 40 of the ASIO I/O and I
won't have to use Cubase at all? If this is the case, I'm drooling!!!! This

is *exactly* what I've been hoping for!!! Building an Intel machine to
support this will be a pleasure.

;o)

"Dimitrios" <musurgio@otenet.gr> wrote in message news:45413e68$1@linux...
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> Dimitrios
>
> "DJ" <notachance@net.net> wrote:
> >Dimitrios,
> >
> >What I want to do is as follows:
> >
> >Create a Paris mix template with 40 tracks with each track having an
insert
> >inabled and routed to that track in the Paris virtual patchbay.
> >
> >I will have 40 ADAT inputs and outputs routed between Paris and the
Pulsar
> >cards.
> >
> >Now I want to be able to send my Paris tracks to the Pulsar mixer,
process
> >them there with Pulsar plugins and Uad-1 plugins at a certain fixed
latency
> >that I can compensate each track in Paris, and then return them to Paris
> >without having to use Cubase SX as a VST host at all.
> >
> >Is this possible?
> >
> >Thanks,
> >
> >DJ
> >
> >
> >
> >"Dimitrios " <musurgio@otenet.gr> wrote in message news:45413977@linux...

> >>
> >> DJ,
> >> To help you understand.
> >> Pulsar lets you alter its ASIO routing to achieve that amazing 64 in
out
> >> asio routing !!
> >> Appears as asio 1,2 3,4 5,6 etc.
> >> Now ANY asio related host appchainer will showon its routing when
> >selecting
> >> Scope asio these exact ins and outs.
> >> Thats exactly what I was using all that years well with cubase.
> >> If cubase 5 did that for me I am sure you can use like chainer to
> >compliment
> >> that routing.
> >> So why bother with wires when Pulsar comes to rrescue ?
> >> Can you imagine the possibiolties ?
> >> If you can run Pulsar at 3 ms (why not I could) then using the
> >"millidelay"
> >> free sample delay inside scope you can delay all your audio routing for
> >exact
> >> nudge intervals (80 samples 160 samples etc)
> >> Hope this helps.
> >> Regards,
> >> Dimitrios
> >>
> >>
> >> "DJ" <notachance@net.net> wrote:
> >> >If I could get someone to help me code a very simple VST rack like
Forte,
> >> >would it be possible to run it outside of Paris as an independent
> >> >application and have it interface with Paris using Wires? Since Paris
> >> >requires minimal host processing power and
> >UAD-1/Powercore/Duende/LiquidMix,
> >> >etc. all have their won DSP engines, would it be possible to take some
> of
> >> >the available native CPU horsepower and apply a goodly amount of it it
> to
> >> >knocking down the latency of these plugins to a low, as in maybe zero
> and
> >> if
> >> >not zero, then at least a predictable number of samples which could
> >> >correspond exactly to the nudge parameters in the Paris editor and
then
> >> >connect the channels of this external VST rack to Paris inserts and
auxes
> >> >using Wires? I'm serious here man. Do you think this would be
possible?
> >> I've

> >> >got a guru here whose wife wants to do a project at this studio. I can
> >trade
> >> >session time for R & D time....I'm absolutely sure of it.
> >> >
> >> >Deej
> >> >
> >> >
> >> >
> >>
> >
> >
>"Dimitrios" <musurgio@otenet.gr> wrote:
>
>Hi,
>Even with all these high end shits , mic preamps ad converters microphones
>sometimes you get amazed by some folks using "cheap" alternatives like Yamaha
>01v or AW4416 workstations.
>I heard some drums and I heard "that" sound on the snare I am looking for
>, crisp punchy and with lot of harmonics.
>Like "californication" song ,you know what snare I mean.
>I believe they used O2 on that song, not sure.
>So as a cheap alternative would I get the same sound using DS2416 dsp factory
>with analog ins and adat out ?
>I amsure it is the yamaha converters and not probably the mic preamps.
>If it is also mic preamps (I wish not) then I could get me a cheap old
01
>(not V) and use analog outs to DS2416 analog in (DS 2416 has same internal
>routings like 02) and then an adat out card to paris or pulsar.
>What do you think ??
>Regards,
>Dimitrios

If you can't get it I can give you a nice sample or two that
you can use.

NeilGo the other direction. Put the stripe in SAW and let Paris chase?
AA

"Don Nafe" <dnafe@magma.ca> wrote in message news:45411e34$1@linux...
> SMPTE was a no go...sending smpte to dakota results in it converting it to
> MTC which is what Sawstudio is having problems reading and I have no other
> way of reading smpte on my second rig although all ideas are welcome
>
> DOn
>
>
> "Don Nafe" <dnafe@magma.ca> wrote in message news:45409ec1$1@linux...

>> Hey Aaron
>>
>> I'll be attempting this today, hopefully with better results
>>
>> DOn
>>
>>
>> "Aaron Allen" <know-spam@not_here.dude> wrote in message
>> news:45400cfb@linux...
>>> Have you tried putting a SMPTE stripe in to read the playback time code?
>>> AA
>>>
>>> "Don Nafe" <dnafe@magma.ca> wrote in message news:4540085c$1@linux...
>>>> Hi all
>>>>
>>>> I've been playing with Sawstudiolite and no matter how a configure
>>>> things I can't achieve accurate sync with Paris when tracks are loaded
>>>> into Saw...
>>>>
>>>> A straight flying tracks there and back via lightpitp results in 2055
>>>> samples of latency everytime, but record or drop a track into it and it
>>>> wanders all over the place. Bob the developer seems to think this is a
>>>> result of the MTC sync my Dakota card is generating from the ADAT sync
>>>> in Paris now I know Nuendo and Cubase lock to Paris
>>>>
>>>> So my question is...what other apps do the same
>>>>
>>>> Thanks
>>>>
>>>
>>>
>>>
>>
>>
>
>I think you have spent way too much time around Deej.
AA

"Dimitrios" <musurgio@otenet.gr> wrote in message news:45413837$1@linux...
>
> Hi,
> Even with all these high end shits , mic preamps ad converters microphones
> sometimes you get amazed by some folks using "cheap" alternatives like
> Yamaha
> 01v or AW4416 workstations.
> I heard some drums and I heard "that" sound on the snare I am looking for
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> (not V) and use analog outs to DS2416 analog in (DS 2416 has same internal
> routings like 02) and then an adat out card to paris or pulsar.
> What do you think ??
> Regards,
> DimitriosDoug O and Dream Theatre.
AA

"DJ" <notachance@net.net> wrote in message news:4540e3a6@linux...
>A Love Affair....the music of Ivan Lins...the song was She Walks This
>Earth.
> Sting on lead vocal. Excellent CD. I think this won a Grammy in 2000-01.
> He
> may have won other Grammys as well. I think he still uses Paris. Any hits
> by
> Markus Miller are likely to involve a Paris system in the production as
> well.
>
>
>
>
> "Don Nafe" <dnafe@magma.ca> wrote in message news:4540d2c3@linux...
>> Didn't Jason Miles win a grammy not long ago?
>>
>>
>> "alex plasko" <alex.plasko@snet.net> wrote in message
> news:4540cd94@linux...
>> > Just a little curious. I dont recall seeing a count on our beloved news
>> > group here as to how many hit records, or at least ones that charted
> ,were
>> > recorded with paris.
>> > we all know about BT and the Lonestar track.
>> > 1)How many of us worldclass engineers have actually hit paydirt using
>> > paris?
>> > 2)Has anyone researched the top system(s) used for said hit records?
>> > I dont want to hear the hype. just the facts, if anyone knows.
>> > just curious guys,(and girls) no need to start a flame fest here.:-)
>> >
>>
>>
>
>would need a turorial to pull that one off...and don't you need a smpte
reader/interface to do this?

Don

"Aaron Allen" <know-spam@not_here.dude> wrote in message
news:454145c1@linux...
> Go the other direction. Put the stripe in SAW and let Paris chase?
> AA
>
>
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>>>>> samples of latency everytime, but record or drop a track into it and
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>>>>> sync in Paris now I know Nuendo and Cubase lock to Paris
>>>>>
>>>>> So my question is...what other apps do the same
>>>>>
>>>>> Thanks

>>>>>
>>>>
>>>>
>>>>
>>>
>>>
>>
>>
>
>Ebay a JL Cooper PPS-1 or PPS-2. I loved mine, though I haven't had to use
them in years they always just handled the sitch with grace. Bet they're
cheap now.

AA

"Don Nafe" <dnafe@magma.ca> wrote in message news:45414828@linux...
> would need a turorial to pull that one off...and don't you need a smpte
> reader/interface to do this?
>
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>
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>>>>>>
>>>>>> Thanks
>>>>>>
>>>>>
>>>>>
>>>>>
>>>>
>>>>
>>>
>>>
>>
>>
>
>For those keeping score I have these left for sale after Neil's purchase:

eds 150
c16 75
8in 200
8out 200

> Email me or call
> John
> john@kfocus.com
> 843-559-3777 eves"DJ" <notachance@net.net> wrote:
>;o)....seriously.....do you think I could have damaged the mobo by running
a
>3.3v card in a 1.5v slot...and/or damaged the video card?
>
No.
But anything is possible.
GeneI'll be blunt here Aaron...Sawstudio isn't worth it

Don

"Aaron Allen" <know-spam@not_here.dude> wrote in message
news:454148a5@linux...
> Ebay a JL Cooper PPS-1 or PPS-2. I loved mine, though I haven't had to use
> them in years they always just handled the sitch with grace. Bet they're
> cheap now.
>
> AA
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>>>>>>>
>>>>>>> Thanks
>>>>>>>
>>>>>>
>>>>>>
>>>>>>
>>>>>
>>>>>
>>>>
>>>>
>>>
>>>
>>
>>
>
>On 25 Oct 2006 09:40:07 +1000, "Ab" <ab.vangoor@wanadoo.fr> wrote:

>
>Just read about the new Macbook pro. The FW 800 port was what I was waiting
>for.
>First thing in the morning is to contact my local apple reseller.
>Btw, anyone interested in an Albook G4 1.5Ghz with 2GB Ram? Rick, DJ;?)
>
>Best
>Ab
I already called MacMall and ordered me up one of them there 15"
2.33GHz ones.

I was waiting for the Core2 chips since the rumor started that they
were coming....

Same price as before, but better chips, double the ram, larger hard
drive, and one more FW port.

The worst part now is waiting for it to ship.

pabInquiring minds want to know :-)
Why do you want to throw Cubase out?

Mic.

"DJ" <notachance@net.net> wrote:
>Dimitrios......I don't want to use a VST host, if the vst host is cubase
SX.
>If it is the Pulsar mixer than that would be OK, but I want to get as far
>away from Cubase as I can. If I wanted to use Cubase, I could use the RME
>cards I have now.
>
>Thanks,
>
>DJ
>
>"Dimitrios" <musurgio@otenet.gr> wrote in message news:45413238$1@linux...
>>
>> Hey DJ,
>> Do you know that Pulsar gives you 64 routes to and from asio
>destination/sources
>> ??
>> Soyou can route 64 audio tracks back and forth inside pulsar from VST
host
>> using asio and the transfer to paris.
>> Regards,
>> Dimitrios
>>
>> "DJ" <notachance@net.net> wrote:
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>> >etc. all have their won DSP engines, would it be possible to take some
of
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>> >knocking down the latency of these plugins to a low, as in maybe zero
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>> >not zero, then at least a predictable number of samples which could
>> >correspond exactly to the nudge parameters in the Paris editor and then
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>> >Deej
>> >
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>
>I just want a simple FX rack where I can send Paris tracks, process them
with UAD-1 plugins and return them. By setting the buffers on the native
audio interface to 512k and inserting a UAD-1 Delaycomp set to compensate
for 5 UAD-1 plugins it's possible to delay compensate all Paris tracks by a
small, fixed latency this way and still have a viable visual reference to
the now line in the editor for fader automation. I know Cubase will do this
but it's got so much other crap going on that it's inherently unstable when
used in this way, at least on my machine. Once you set up a project with 48
tracks with 48 I/O busses set to monitor with FX, it's just not that stable.
Now maybe there's something wrong with my computer, but Forte was much more
stable than Cubase SX wuth much lower CPU usage with the same bus count.

I think I have found the ticket here.
 http://www.plogue.com/index.php?option=content&task=view &id=21&Itemid=35

I've been chatting with the developer. He says the bussing is limited only
by the number of I/O and the plugins are limited only by the CPU
horsepower.

;o)

"Mic Cross" <crzymnmchl@comcast.net> wrote in message
news:45416b6e$1@linux...
>
> Inquiring minds want to know :-)
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>
> Mic.
>
>
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> >> >
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news:45417186$1@linux...
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>> >> >
>> >> >Deej
>> >> >
>> >> >
>> >> >
>> >>
>> >
>> >
>>
>
>Let us know how you like it.

Cheers,
  -Jamie
  www.JamieKrutz.com

Paul Braun wrote:
> On 25 Oct 2006 09:40:07 +1000, "Ab" <ab.vangoor@wanadoo.fr> wrote:
>
>> Just read about the new Macbook pro. The FW 800 port was what I was waiting
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>> First thing in the morning is to contact my local apple reseller.
>> Btw, anyone interested in an Albook G4 1.5Ghz with 2GB Ram? Rick, DJ;?)
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>> Ab
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> 2.33GHz ones.
>
> I was waiting for the Core2 chips since the rumor started that they
> were coming....
>
> Same price as before, but better chips, double the ram, larger hard
> drive, and one more FW port.
>
> The worst part now is waiting for it to ship.
>
> pabDeej, if the 3v cards are pulling their operating voltage from an onboard
regulator IC, it is possible to damage components on the vid card AND the
mobo. When the reg can't get what it wants for V, it tries to up the
voltage which requires more current. The reg IC on the card gets hot while
the mobo gets siphoned for more current than it perhaps can deliver.

MW

"DJ" <notachance@net.net> wrote in message news:4540d673@linux...

> ;o)....seriously.....do you think I could have damaged the mobo by running
> a
> 3.3v card in a 1.5v slot...and/or damaged the video card?
>
> "Gene Lennon" <glennon@NOSPmyrealbox.com> wrote in message
> news:4540d2c9$1@linux...
>>
>> "Tyrone Corbett" <tyronecorbett@comcast.net> wrote:
>> >
>> >"alex plasko" <alex.plasko@snet.net> wrote:
>> >>use the dremel and cut a new slot. works every time!
>> >
>> >Nah Deej, just pull out the ole trusty the SAWZAL!
>> >
>> >Tyrone
>> >
>> >
>> >
>> This is DJ !!
>
>`Hey DJ,
There is a vst chainer called RT Player which the pro version can accomodate
32 asio ins and outs !
BUT I would encourage you use cuabse for that,one because you have it , secondly
it can accomodate all 64 asio ins and outs and thirdly You can only use VST
mixer as chainer with no audio tracks.
VStack might be useful or Cubase 5 which I use which is light and simple.
I will try RT Player for you to check how this works.
Dsound is making it.
I searched all chainers and is the only one to my knowledge that can acommodate
32 asio ins and outs.
Regards,
Dimitrios

"DJ" <notachance@net.net> wrote:
>So since neither Forte or Chainer will allow more than 16 I/O if it is used
>as a standalone application I can open up Forte or chainer on each of the
40
>Pulsar channels and this will allow it to see all 40 of the ASIO I/O and
I
>won't have to use Cubase at all? If this is the case, I'm drooling!!!! This
>is *exactly* what I've been hoping for!!! Building an Intel machine to
>support this will be a pleasure.
>
>;o)
>
>
>

>
>
>"Dimitrios" <musurgio@otenet.gr> wrote in message news:45413e68$1@linux...
>>
>> DJ,
>> You can do that of course but you will have to use a chainer VST loader
>like
>> chainer or forte , whatever that will load the vsts take input from scope
>> asio sends and send back thru scope asio again back to pulsar mixer.
>> Regards,
>> Dimitrios
>>
>> "DJ" <notachance@net.net> wrote:
>> >Dimitrios,
>> >
>> >What I want to do is as follows:
>> >
>> >Create a Paris mix template with 40 tracks with each track having an
>insert
>> >inabled and routed to that track in the Paris virtual patchbay.
>> >
>> >I will have 40 ADAT inputs and outputs routed between Paris and the
>Pulsar
>> >cards.
>> >
>> >Now I want to be able to send my Paris tracks to the Pulsar mixer,
>process
>> >them there with Pulsar plugins and Uad-1 plugins at a certain fixed
>latency
>> >that I can compensate each track in Paris, and then return them to Paris
>> >without having to use Cubase SX as a VST host at all.
>> >
>> >Is this possible?
>> >
>> >Thanks,
>> >
>> >DJ
>> >
>> >
>> >
>> >"Dimitrios " <musurgio@otenet.gr> wrote in message news:45413977@linux...
>> >>
>> >> DJ,
>> >> To help you understand.
>> >> Pulsar lets you alter its ASIO routing to achieve that amazing 64 in
>out
>> >> asio routing !!
>> >> Appears as asio 1,2 3,4 5,6 etc.

>> >> Now ANY asio related host appchainer will showon its routing when
>> >selecting
>> >> Scope asio these exact ins and outs.
>> >> Thats exactly what I was using all that years well with cubase.
>> >> If cubase 5 did that for me I am sure you can use like chainer to
>> >compliment
>> >> that routing.
>> >> So why bother with wires when Pulsar comes to rrescue ?
>> >> Can you imagine the possibiolties ?
>> >> If you can run Pulsar at 3 ms (why not I could) then using the
>> >"millidelay"
>> >> free sample delay inside scope you can delay all your audio routing
for
>> >exact

>> >> nudge intervals (80 samples 160 samples etc)
>> >> Hope this helps.
>> >> Regards,
>> >> Dimitrios
>> >>
>> >>
>> >> "DJ" <notachance@net.net> wrote:
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>Forte,
>> >> >would it be possible to run it outside of Paris as an independent
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>> >> >etc. all have their won DSP engines, would it be possible to take
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>> >> I've
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>"Mic Cross" <crzymnmchl@comcast.net> wrote in message
>news:45416b6e$1@linux...
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>> Inquiring minds want to know :-)
>> Why do you want to throw Cubase out?
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>> >>
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>> >
>>
>
>Ya might want those in cards Neil, for that nice fat drum sound

"Neil" <OIUOIU@OIU.com> wrote in message news:4541368b$1@linux...
>
> I'll take an EDS, an 8-in & the MEC. Seriously.
>
> (going to do my own damn summing experiments &
> comparisons! :D)
>
> Reserve those for me, please - I'll follow up with an e-mail
> this evening to arrange for payment & shipping.
>
> Neil
>
>
> John <no@no.com> wrote:
>>This is what I have for sale in EXCELLENT shape. Let's move it out !
>>Shipping $10.
>>
>>eds 150

>>eds 150
>>c16 75
>>8in 200
>>8in 200
>>8out 200
>>mec 75
>>
>>
>>Email me or call
>>John
>>john@kfocus.com
>>843-559-3777 eves
>Cubase 5? Where do you get Cubase 5? I just ordered Cubase 4 and it will be here next week. Are you in a time warp? Show me a website at Steinberg with Cubase 5 ! WOW

John

Dimitrios wrote:
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>>>>>> session time for R & D time....I'm absolutely sure of it.
>>>>>>
>>>>>> Deej
>>>>>>
>>>>>>
>>>>>>
>>>>
>>
>Hehehehehhehee
That was funny,,, thanks....:)
I was reffering to the old Cubase VST 5.1 !!
Regards,
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>>Hi DJ,

I guess what we are talking about is two things:

1. An ASIO host application with let's say 64 ins and 64 outs. This app
would also be a VST host application that would let you insert plugs on each
of the 64 ins. It would add up the total latency on each input, buffer the
output to some consistent user entered amount, and send it out the output.
The latency for every channel would end up being exactly the same user entered
amount.

2. A simple VST plugin that would allow you to select an input and output
ASIO channel. That's all that one would do. This plug wouldn't have any
latency of it's own.

So my question is..... Is there any other possible use for such a setup?
I would be willing to get involved in an open source freeware, ad/donation
supported project for this if there was.

ChuckChuck,
You have hit gold here.
Well at least amoung Paris users...
Can you imagine all Parisians have the ability tosend their Cubase (sorry
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with asio to complete the route.

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Anyway Chuck I am willing to pay for it 100$ is ok ? he
One thing that matters is the vst instances must be synced at the same latency
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Other daws that luck asio could benefit also.
How much would you define as a payment for the time and skills that would
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Dimitrios

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>supported project for this if there was.
>
>Chuck
>
>
>
>If this lets me use my uad plugins in paris with minimal latency grief,
count me and my MasterCard in as well.

Rob

"chuck duffy" <c@c.com> wrote in message news:454206a9$1@linux...
>

> Hi DJ,
>
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> supported project for this if there was.
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>
>
>"Don Nafe" <dnafe@magma.ca> wrote:
>Ya might want those in cards Neil, for that nice fat drum sound

Well, I've been interested for some time in trying to see if
summing through Paris (or something like a Dangerous 2-buss,
folcrom, etc.) will make a difference - I like what I'm
getting out of Cubase, but I'm always open to other options.

Just hate to stand the chance of losing that high-end
transparency that i'm getting right now, but we'll see.

Neil"Tom Bruhl" <arpegio@comcast.net> wrote in news:4540fd2c$1@linux:

> Scott,
> Yes the wrapper will fix the GUI of most plugs. My UADs are still
> showing meters with latency though.
> Tom

Thanks Tom!

-scott v.Dimitrios,

The Aw4416 is a high-end sounding studio in a box. It's converters are some of the best in the box. They are "Highly sought after units that are "secret weapons" of some producers.

I recently mixed a project (Gospel) that originated from the newer Yamaha AW line (AW1600). MAN!!! Amazing sound quality. The pre amps are from the DM2000. If you even heard the DM-2000 mic pres, then you know what i saying. I hear that the top of the line Aw (AW2400) is in a "sonic" league of it's own. basically, it's a DM-2000 with a recorder.

I was so impressed with these units, that i'm planning on getting (AW2400) as my remote recorder/mixer in place of my laptop setup.. It's just that good..

"Dimitrios" <musurgio@otenet.gr> wrote:
>
>Hi,
>Even with all these high end shits , mic preamps ad converters microphones
>sometimes you get amazed by some folks using "cheap" alternatives like Yamaha
>01v or AW4416 workstations.
>I heard some drums and I heard "that" sound on the snare I am looking for
>, crisp punchy and with lot of harmonics.
>Like "californication" song ,you know what snare I mean.
>I believe they used O2 on that song, not sure.
>So as a cheap alternative would I get the same sound using DS2416 dsp factory
>with analog ins and adat out ?
>I amsure it is the yamaha converters and not probably the mic preamps.
>If it is also mic preamps (I wish not) then I could get me a cheap old
01
>(not V) and use analog outs to DS2416 analog in (DS 2416 has same internal
>routings like 02) and then an adat out card to paris or pulsar.
>What do you think ??
>Regards,
>DimitriosWell, the real question is if this has any use *outside* of paris users.
It would need to have some sort of audience beyond the paris community to get the kind of numbers that make ad revenue possible.

Chuck

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>I just discovered that two of the three PCI slots on my "new" PC are too short to accomodate MY UAD-1s. So I am considering getting a (probablly used) Magma box. This would use a PCI slot in the PC, not a PCIe slot.

Does this work well? Are there cable length restrictions?

ThanksChuck
would an auto latency compensation plug in be easier to build?Im no software engineer but isnt that just a ping?

"chuck duffy" <c@c.com> wrote in message news:454206a9$1@linux...
>
> Hi DJ,
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> would also be a VST host application that would let you insert plugs on
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> I would be willing to get involved in an open source freeware, ad/donation
> supported project for this if there was.
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> Chuck
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>
>Hi Alex,

It might be, but I'm looking at a few other angles:

1. The plugin code in paris is ancient and it runs lots of newer plugs poorly, with weird side effects, or not at all. This would be an opportunity to run the plugs in a modern environment, and get latency compensation

2. We don't have access to the VST/DX plug-in code, or any other paris application
code, so I can't really produce a delay compensator.

3. Since there are other, but more limited apps like this out there that
act as effects, instrument hosts, i figured that there might be an audience
outside of the paris community that could use it too. That's the only way
I would really take on something like this.

Chuck

"alex plasko" <alex.plasko@snet.net> wrote:
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>> supported project for this if there was.
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>> Chuck
>>
>>

>>
>>
>
>"chuck duffy" <c@c.com> wrote:
>
>Well, the real question is if this has any use *outside* of paris users.

>It would need to have some sort of audience beyond the paris community to
>get the kind of numbers that make ad revenue possible.

Check, what if you included, as part of the whole package,
a few plugins as well (maybe a basic selection of dynamics,
reverb, EQ, modulation stuff, etc. - just some different flavors
of those types of things)? That way it could give anyone &
everyone a reason to try it beyond just the routing options.
Are any of the plugins you ported over port-able to VST? If so,
you could use some of those.

Neill hear that. I wouldnt want to work for next to nothing either.maybe someday
edmund will have a change of heart and i can dust off the paris rig again.if
i live that long :-)"chuck duffy" <c@c.com> wrote in message
news:4542245c$1@linux...
>
> Hi Alex,
>
> It might be, but I'm looking at a few other angles:
>
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> poorly,
> with weird side effects, or not at all. This would be an opportunity to
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>I co-produced and recorded an album done in Paris with an artist named Sunny
Sweeney (http://sunnysweeney.com/index.html). Sunny has entries on the preliminary
ballot in four categories for the upcoming Grammy Awards:

Best Female Country Vocal Performance
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(83 entries) for
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The top 5 in each category will advance to the final round of voting.

You can hear clips from the record here:

http://profile.myspace.com/index.cfm?fuseaction=user.viewpro file&friendid=47942490

Sunny is a special talent and genuine nice person! Any consideration would be much appreciated.....

Thanks,

Tommy

Tommy Detamore

http://www.cherryridgestudio.comChuck,

There is already an ASIO host application that has unlimited I/O so #2 has been covered. It's not simple though
http://www.plogue.com/index.php?option=content&task=view &id=21&Itemid=35
I haven't tried it yet but will likely get around to it over the weekend or early next week. I was thinking of something that could interface directly with Paris so that the UAD-1 cards could work directly on the Paris DAW without having to interface via ADAT on a second workstation. Old Magma's are cheap these days and having the cards in the Paris workstation running Win XP without having to interface with a second DAW using lightpipe would be ideal. this is why I was thinking of Wires. As far as an ASIO driver, under the "Paris DAW being host" scenario, without an efficient ASIO driver, for Paris, I don't see this happening. To tell you the truth, I haven't used the Paris ASIO driver in years. I wonder if it would work with a VST host like Forte or Chainer? I do remember some latency with this driver, but it's been a long time. Anyway, as far as third party uses for the VST host you are proposing in #1 ....I honestly don't know unless they were

wanting to stream from a DAW with no latency compensation to a digital
mixer. I don't think there are any DAWs, other than Paris left on earth that
don't have latency compensation.

If you decide you want to do this, I will support your efforts 100%.

Thanks,

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>Is the reason for all of this to basically make up for the need of native
inserts on the PARIS AUX busses?

Brandon

"DJ" <notachance@net.net> wrote in message news:45422b7b@linux...
> Chuck,

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where and how much drives me even crazier than the crap I'm dealing with
trying to find a wyay not to do it....plus, it makes using the now line as a
visual reference in the editor a useless endeavor because the latency is so
extreme that the track is practically off the screen by the time you hear
it.

;o)

"Brandon" <a@a.com> wrote in message news:45422e76$1@linux...
> Is the reason for all of this to basically make up for the need of native
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> Brandon
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> "DJ" <notachance@net.net> wrote in message news:45422b7b@linux...
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>"Gary Flanigan" <gary_flanigan@ce9.uscourts.gov> wrote in message
news:454221fd$1@linux...
> I just discovered that two of the three PCI slots on my "new" PC are too
> short
> to accomodate MY UAD-1s. So I am considering getting a (probablly used)
> Magma box. This would use a PCI slot in the PC, not a PCIe slot.
>
> Does this work well? Are there cable length restrictions?

I have three Magma chassis, two 13 slot models and a 7 slot. They all just
work. My only complaint would be that the original fans are quite noisy.
IIRC, there are a couple different cable lengths, the longest being
something like three feet. You may find more about that at
http://www.mobilityelectronics.com/expansion/ ...

Doug

http://www.parisfaqs.comSo if there was an automatic delay compensator and Native inserts on the AUX
busses everyone would be happy?

Brandon

"DJ" <notachance@net.net> wrote in message news:454231d8@linux...
> Not really. The need for this (for me) is to be able to work in a less
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> > > > Chuck
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> > > >
> > >
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> >
> >
>
>Hey Tommy!

Glad to hear it. Good luck.

DC

"Tommy Detamore" <cherryrdg@aol.com> wrote:
>
>I co-produced and recorded an album done in Paris with an artist named Sunny
>Sweeney (http://sunnysweeney.com/index.html). Sunny has entries on the preliminary
>ballot in four categories for the upcoming Grammy Awards:
>
>Best Female Country Vocal Performance
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>Sunny is a special talent and genuine nice person! Any consideration would
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>Thanks,
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>Tommy
>
>-------------------
>Tommy Detamore
>
>http://www.cherryridgestudio.com
>
>
>Dj,

I think you are misunderstanding a little :-) The asio streams in this "new"
asio host would be all virtual and not require any hardware or adats interfaces
at all.

The "new" vst plug when used on a channel in paris would let you select a
route in and back out of the "new" asio host.

The "new" host would accept real vsts and delay them to a specific user entered,
consistent threshold, then feed the output back to the same "new" vst plug
in paris.

It would be hardwareless.

Chuck

Chuck
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>No because there are tons of plugs that just DONT run in paris, let alone
vstis!

Chuck

"Brandon" <a@a.com> wrote:
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>> > > "chuck duffy" <c@c.com> wrote in message news:454206a9$1@linux...
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>> > > > Chuck
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>> > > >
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>
>Dj,

The only thing wires can do is re-route signals from one place in the EDS
to another. Believe it or not, I don't have access to the actual audio!

ChuckAhhh.....OK......so Paris would open this as a VST plugin? Can't we already
do that with chainer, etc?

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> > >> Chuck
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> > >>

> > >
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> >
>
>That's like asking my wife if I finish painting the Kitichen and cut the grass
can I watch all the games every Sunday for the rest of the year....

"Brandon" <a@a.com> wrote:
>So if there was an automatic delay compensator and Native inserts on the
AUX
>busses everyone would be happy?
>
>Brandon
>
>
>
>
>"DJ" <notachance@net.net> wrote in message news:454231d8@linux...
>> Not really. The need for this (for me) is to be able to work in a less
>> chaotic way when using UAD-1 plugins with Paris. To be nudging tracks
>around
>> all over the *^%$%$^&* place and trying to keep up with what whas nudged
>> where and how much drives me even crazier than the crap I'm dealing with
>> trying to find a wyay not to do it....plus, it makes using the now line
as
>a
>> visual reference in the editor a useless endeavor because the latency
is
>so
>> extreme that the track is practically off the screen by the time you hear
>> it.
>>
>> ;o)
>>
>> "Brandon" <a@a.com> wrote in message news:45422e76$1@linux...
>> > Is the reason for all of this to basically make up for the need of
>native
>> > inserts on the PARIS AUX busses?
>> >
>> > Brandon
>> >
>> >
>> >
>> > "DJ" <notachance@net.net> wrote in message news:45422b7b@linux...
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>> > > "chuck duffy" <c@c.com> wrote in message news:454206a9$1@linux...

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>>
>
>Aaron,

So do you know which Dream Theater albums were done on Paris?

Tony

"Aaron Allen" <know-spam@not_here.dude> wrote in message
news:45414827$2@linux...
> Doug O and Dream Theatre.
> AA
>
> "DJ" <notachance@net.net> wrote in message news:4540e3a6@linux...
>>A Love Affair....the music of Ivan Lins...the song was She Walks This
>>Earth.
>> Sting on lead vocal. Excellent CD. I think this won a Grammy in 2000-01.
>> He
>> may have won other Grammys as well. I think he still uses Paris. Any hits
>> by
>> Markus Miller are likely to involve a Paris system in the production as
>> well.
>>
>>
>>
>>
>> "Don Nafe" <dnafe@magma.ca> wrote in message news:4540d2c3@linux...
>>> Didn't Jason Miles win a grammy not long ago?
>>>
>>>
>>> "alex plasko" <alex.plasko@snet.net> wrote in message
>> news:4540cd94@linux...
>>> > Just a little curious. I dont recall seeing a count on our beloved
>>> > news
>>> > group here as to how many hit records, or at least ones that charted
>> ,were
>>> > recorded with paris.
>>> > we all know about BT and the Lonestar track.
>>> > 1)How many of us worldclass engineers have actually hit paydirt using
>>> > paris?
>>> > 2)Has anyone researched the top system(s) used for said hit records?
>>> > I dont want to hear the hype. just the facts, if anyone knows.
>>> > just curious guys,(and girls) no need to start a flame fest here.:-)
>>> >
>>>
>>>
>>
>>
>
>Well, there were three things going on.

1. I thought chainer didn't allow enough channels, or enough instances.

2. I thought the other VST hosts you were using required physical audio connections (ie were not virtual) .

3. I thought the other hosts didn't have enough asio channels

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>>
>
>Hey Deej

Please excuse my pestering but Reaper has two relatively new things
happening now...one is the their version of rewire (routing between apps)

and the second is "Remote" which is Wormhole-like (passing audio via
networking) and between these two things it might be what you're looking
for.

DOn

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> Chuck
>
>
>
>Very good work Tommy-

Checked out your website and listened to samples. Very nice.

Hope you get some praise from the industry.

Ted

"Tommy Detamore" <cherryrdg@aol.com> wrote:
>
>I co-produced and recorded an album done in Paris with an artist named Sunny
>Sweeney (http://sunnysweeney.com/index.html). Sunny has entries on the preliminary
>ballot in four categories for the upcoming Grammy Awards:

>
>Best Female Country Vocal Performance
>(52 entries involving 43 artists) for
>Heartbreakers Hall of Fame/Sunny Sweeney
>
>Best Country Collaboration with Vocals
>(38 entries) for
>Lavender Blue/Sunny Sweeney & Jim Lauderdale
>
>Best Country Song
>(129 entries) for
>Heartbreakers Hall of Fame/Sunny Sweeney
>
>Best Country Album
>(83 entries) for
>Heartbreakes Hall of Fame/Sunny Sweeney
>
>The top 5 in each category will advance to the final round of voting.
>
>You can hear clips from the record here:
>
> http://profile.myspace.com/index.cfm?fuseaction=user.viewpro file&friendid=47942490
>
>Sunny is a special talent and genuine nice person! Any consideration would
>be much appreciated.....
>
>Thanks,
>
>Tommy
>
>------------------
>Tommy Detamore
>
>http://www.cherryridgestudio.com
>
>
>Just to clear up things regarding Chuck's suggestion and ASIO.
Well
1)you will be needing another audio card that supports asio
i.e. Pulsar card with scope environement (which can acommodate 16adat channels,spdif,2
analog) Pulsar can give you 64 asio routing channels.
Now if Chuck's vst2asio plugin can "see" these channels (or less than 64
maybe 24 whatever) then if pulsar runs at 3ms asio the latency beetween pulsar
and Paris back and forth will be 6ms.

2) If you are gonna use Cubase on same computer with Paris you will be needing
either pulsar card or rme card on same computer with paris.
So vst2asio will see the cubase asio outputs and so audio can transfer back

and forth.

3) If you are gonna use also UAD1 cards there will be a big pci stress on
the machine.

I see only true benefit with a dsp card like Pulsar which has asio or any
other dsp card with asio like Emu or maybe the Nuende or Focusrite ...

If you don't need to use UAD1 on same computer I am sure with one asio audio
card cubase can be this way intergrated with Paris on same computer with
very small latency as so to bring in VSTI and other.

Just some thoughts...
Regards,
Dimitrios

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>>> "chuck duffy" <c@c.com> wrote in message news:4542449d$1@linux...
>>> >
>>> > Dj,
>>> >
>>> > I think you are misunderstanding a little :-) The asio streams in this
>>> "new"
>>> > asio host would be all virtual and not require any hardware or adats
>>> interfaces

>>> > at all.
>>> >
>>> > The "new" vst plug when used on a channel in paris would let you select
>>a
>>> > route in and back out of the "new" asio host.
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>>> > The "new" host would accept real vsts and delay them to a specific
user
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>>plug
>>> > in paris.
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>>> > It would be hardwareless.
>>> >
>>> > Chuck
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>>> >
>>> > Chuck
>>> > "DJ" <notachance@net.net> wrote:
>>> > >Chuck,
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>>> > >There is already an ASIO host application that has unlimited I/O so
>#2
>>> has
>>> > >been covered. It's not simple though
>>> >
>>> http://www.plogue.com/index.php?option=content&task=view &id=21&Itemid=35
>>> > >I haven't tried it yet but will likely get around to it over the
>>weekend
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>>> > >with Paris so that the UAD-1 cards could work directly on the Paris
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>>> > >without having to interface via ADAT on a second workstation. Old
>>Magma's
>>> > >are cheap these days and having the cards in the Paris workstation
>>> running
>>> > >Win XP without having to interface with a second DAW using lightpipe
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>>> > >be ideal. this is why I was thinking of Wires. As far as an ASIO
>>driver,
>>> > >under the "Paris DAW being host" scenario, without an efficient ASIO
>>> driver,
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>>> > used

>>> > >the Paris ASIO driver in years. I wonder if it would work with a VST
>>host
>>> > >like Forte or Chainer? I do remember some latency with this driver,
>but
>>> > it's
>>> > >been a long time. Anyway, as far as third party uses for the VST
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>>> you
>>> > >are proposing in #1 ....I honestly don't know unless they
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>>> > >wanting to stream from a DAW with no latency compensation to a digital
>>> > >mixer. I don't think there are any DAWs, other than Paris left on
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>>> > that
>>> > >don't have latency compensation.
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>>> > >If you decide you want to do this, I will support your efforts 100%.
>>> > >
>>> > >Thanks,
>>> > >
>>> > >DJ
>>> > >
>>> > >"chuck duffy" <c@c.com> wrote in message news:454206a9$1@linux...
>>> > >>
>>> > >> Hi DJ,
>>> > >>
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>>> > >> 1. An ASIO host application with let's say 64 ins and 64 outs.
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>>> > >> latency of it's own.
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>>> > >> I would be willing to get involved in an open source freeware,
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>>> > >>
>>> > >> Chuck
>>> > >>
>>> > >>
>>> > >>
>>> > >>
>>> > >
>>> > >
>>> >
>>>
>>>
>>
>>
>i installed a new mobo in the paris comp and now i'm getting Error 40/32
when it starts, then quits.

any ideas?

jefsorry that's 40/28. I just took out the adat card...that wasn't it.

could be the new eds card?....I installed the software with an older eds
installed, then got a "hardware not connected" failure so I swapped it
out for a newer one i had kickin' around, which is when I got the message.

jef

jef knight wrote:

> i installed a new mobo in the paris comp and now i'm getting Error
> 40/32 when it starts, then quits.
>
> any ideas?
>
>
> jefhmmm...I moved the eds card up a slot and the error when away.

jk

jef knight wrote:

> i installed a new mobo in the paris comp and now i'm getting Error
> 40/32 when it starts, then quits.
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>
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> jefwhose crazy idea was that? ricks ? jeez

jef knight wrote:
> hmmm...I moved the eds card up a slot and the error when away.
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> jk
>
> jef knight wrote:
>
>> i installed a new mobo in the paris comp and now i'm getting Error
>> 40/32 when it starts, then quits.
>>
>> any ideas?
>>
>>
>> jeffunny you should say that cuz it's a card i bought from rick....lol

John wrote:

> whose crazy idea was that? ricks ? jeez
>
> jef knight wrote:
>
>> hmmm...I moved the eds card up a slot and the error when away.
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>> jk
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>> jef knight wrote:
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>>> any ideas?
>>>
>>>
>>> jef
>>Thanks Neil. I now have the following left. I've had a lot of

inquiries but no MONEY yet, so first come gets it.

eds   150
c16   75
8in   200
8out   200

FIRST PAYPALS GET IT. $10 shipping for all or $30 to Germany
my paypal is john@kfocus.comGlad to see you're alive and kickin Jef

Now where the fuck are my sex tapes...oops I mean sax tracks

;-)

heeehehehehehehe

"jef knight" <thestudio@allknightmusic.com> wrote in message
news:4542612a$1@linux...
>i installed a new mobo in the paris comp and now i'm getting Error 40/32
>when it starts, then quits.
>
> any ideas?
>
>
> jefVERY cool, Tommy... hope you get it!

Neil

"Tommy Detamore" <cherryrdg@aol.com> wrote:
>
>I co-produced and recorded an album done in Paris with an artist named Sunny
>Sweeney (http://sunnysweeney.com/index.html). Sunny has entries on the preliminary
>ballot in four categories for the upcoming Grammy Awards:
>
>Best Female Country Vocal Performance
>(52 entries involving 43 artists) for
>Heartbreakers Hall of Fame/Sunny Sweeney
>
>Best Country Collaboration with Vocals
>(38 entries) for
>Lavender Blue/Sunny Sweeney & Jim Lauderdale
>
>Best Country Song
>(129 entries) for
>Heartbreakers Hall of Fame/Sunny Sweeney
>
>Best Country Album

>(83 entries) for
>Heartbreakes Hall of Fame/Sunny Sweeney
>
>The top 5 in each category will advance to the final round of voting.
>
>You can hear clips from the record here:
>
> http://profile.myspace.com/index.cfm?fuseaction=user.viewpro file&friendid=47942490
>
>Sunny is a special talent and genuine nice person! Any consideration would
>be much appreciated.....
>
>Thanks,
>
>Tommy
>
>------------------
>Tommy Detamore
>
>http://www.cherryridgestudio.com
>
>
>how much for your crickets?

"John" <no@no.com> wrote in message news:45426d95@linux...
> Thanks Neil. I now have the following left. I've had a lot of inquiries
> but no MONEY yet, so first come gets it.
>
> eds   150
> c16   75
> 8in   200
> 8out   200
>
> FIRST PAYPALS GET IT. $10 shipping for all or $30 to Germany
> my paypal is john@kfocus.comImages and Words on SSC's pre release PARIS

"Tony Benson" <tony@standinghampton.com> wrote in message
news:45424bf5@linux...
> Aaron,
>
> So do you know which Dream Theater albums were done on Paris?
>
> Tony
>
>
> "Aaron Allen" <know-spam@not_here.dude> wrote in message
> news:45414827$2@linux...
>> Doug O and Dream Theatre.

>> AA
>>
>> "DJ" <notachance@net.net> wrote in message news:4540e3a6@linux...
>>>A Love Affair....the music of Ivan Lins...the song was She Walks This
>>>Earth.
>>> Sting on lead vocal. Excellent CD. I think this won a Grammy in 2000-01.
>>> He
>>> may have won other Grammys as well. I think he still uses Paris. Any
>>> hits by
>>> Markus Miller are likely to involve a Paris system in the production as
>>> well.
>>>
>>>
>>>
>>>
>>> "Don Nafe" <dnafe@magma.ca> wrote in message news:4540d2c3@linux...
>>>> Didn't Jason Miles win a grammy not long ago?
>>>>
>>>>
>>>> "alex plasko" <alex.plasko@snet.net> wrote in message
>>> news:4540cd94@linux...
>>>> > Just a little curious. I dont recall seeing a count on our beloved
>>>> > news
>>>> > group here as to how many hit records, or at least ones that charted
>>> ,were
>>>> > recorded with paris.
>>>> > we all know about BT and the Lonestar track.
>>>> > 1)How many of us worldclass engineers have actually hit paydirt using
>>>> > paris?
>>>> > 2)Has anyone researched the top system(s) used for said hit records?
>>>> > I dont want to hear the hype. just the facts, if anyone knows.
>>>> > just curious guys,(and girls) no need to start a flame fest here.:-)
>>>> >
>>>>
>>>>
>>>
>>>
>>
>>
>
>There's a prize inside every Paris nugget. :-)

j-cron wrote:
> how much for your crickets?
>
> "John" <no@no.com> wrote in message news:45426d95@linux...
>> Thanks Neil. I now have the following left. I've had a lot of inquiries
>> but no MONEY yet, so first come gets it.

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>> eds   150
>> c16   75
>> 8in   200
>> 8out   200
>>
>> FIRST PAYPALS GET IT. $10 shipping for all or $30 to Germany
>> my paypal is john@kfocus.com
>
>I'm in.
AA

"chuck duffy" <c@c.com> wrote in message news:4542245c$1@linux...
>
> Hi Alex,
>
> It might be, but I'm looking at a few other angles:
>
> 1. The plugin code in paris is ancient and it runs lots of newer plugs
> poorly,
> with weird side effects, or not at all. This would be an opportunity to
> run
> the plugs in a modern environment, and get latency compensation
>
> 2. We don't have access to the VST/DX plug-in code, or any other paris
> application
> code, so I can't really produce a delay compensator.
>
> 3. Since there are other, but more limited apps like this out there that
> act as effects, instrument hosts, i figured that there might be an
> audience
> outside of the paris community that could use it too. That's the only way
> I would really take on something like this.
>
> Chuck
>
> "alex plasko" <alex.plasko@snet.net> wrote:
>>Chuck
>>would an auto latency compensation plug in be easier to build?Im no
>>software
>
>>engineer but isnt that just a ping?
>>
>>"chuck duffy" <c@c.com> wrote in message news:454206a9$1@linux...
>>>
>>> Hi DJ,
>>>
>>> I guess what we are talking about is two things:

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>>> 1. An ASIO host application with let's say 64 ins and 64 outs. This app
>>> would also be a VST host application that would let you insert plugs on
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>>> I would be willing to get involved in an open source freeware,
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>>>
>>> Chuck
>>>
>>>
>>>
>>>
>>
>>
>would work over a network/firewire??
AA

"chuck duffy" <c@c.com> wrote in message news:4542449d$1@linux...
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> I think you are misunderstanding a little :-) The asio streams in this
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>>I haven't tried it yet but will likely get around to it over the weekend
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>>with Paris so that the UAD-1 cards could work directly on the Paris DAW
>>without having to interface via ADAT on a second workstation. Old Magma's
>>are cheap these days and having the cards in the Paris workstation running
>>Win XP without having to interface with a second DAW using lightpipe would
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>>under the "Paris DAW being host" scenario, without an efficient ASIO
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>>the Paris ASIO driver in years. I wonder if it would work with a VST host
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>>been a long time. Anyway, as far as third party uses for the VST host you
>>are proposing in #1 ....I honestly don't know unless they were
>>wanting to stream from a DAW with no latency compensation to a digital
>>mixer. I don't think there are any DAWs, other than Paris left on earth
> that
>>don't have latency compensation.
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>>If you decide you want to do this, I will support your efforts 100%.
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>>Thanks,
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>>DJ
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>>> I would be willing to get involved in an open source freeware,
>>ad/donation
>>> supported project for this if there was.
>>>
>>> Chuck
>>>
>>>
>>>
>>>
>>
>>
>On Thu, 26 Oct 2006 23:25:14 -0600, Jamie K <Meta@Dimensional.com>
wrote:

>
>Let us know how you like it.
>
>Cheers,
> -Jamie
> www.JamieKrutz.com

Will do. MacMall just sent me a shipping notification....

Of everything BUT the MBP.

So, I can stare at the battery and the software boxes. Whee.

pabThat's not true, and dont let anyone tell you it is.
You still need to get all levels as optimised as possible, as we all did
with tape.
Otherwise you are not using all the bits available to you, and noise will be
the end result.
This is why the good/great engineers are what they are...they make sure the
levels are hot....just not to the stage of distortion.
it's a balancing act, but, hey, who said anything done properly is easy.

--
Martin Harrington
www.lendanear-sound.com

"John" <no@no.com> wrote in message news:453ff9d7@linux...
> If you are recording @ 24 bit you really don't need to get that high.
> Peaks of -25 to -15 are more than enough. Terry Manning of Compass Point
> Studios (AC/DC, ZZ Top etc) turned me on to this on a different forum and
> is a big advocate of it. I tried it. I have to agree with him that it made
> a significant improvement in the resulting sound of the recording.
>
> http://www.cubase.net/phpbb2/viewtopic.php?t=55258&highl ight=clippingTo me it's going to be all about whether I think the Pulsar FX are
equivalent to UAD FX...not exactly the same, I wouldn't expect that, but
equivalent. I would really like to be able to use the UAD-1 cards with Paris
in a low latency environment though.These FX just wsound great and I'm used
to working with them so mixing with them is comfortable.

;o)

"Dimitrios" <musurgio@otenet.gr> wrote in message news:4542603b$1@linux...
>
> Just to clear up things regarding Chuck's suggestion and ASIO.
> Well
> 1)you will be needing another audio card that supports asio
> i.e. Pulsar card with scope environement (which can acommodate 16adat
channels,spdif,2
> analog) Pulsar can give you 64 asio routing channels.
> Now if Chuck's vst2asio plugin can "see" these channels (or less than 64
> maybe 24 whatever) then if pulsar runs at 3ms asio the latency beetween
pulsar
> and Paris back and forth will be 6ms.
>

> 2) If you are gonna use Cubase on same computer with Paris you will be needing
> either pulsar card or rme card on same computer with paris.
> So vst2asio will see the cubase asio outputs and so audio can transfer back
> and forth.
>
> 3) If you are gonna use also UAD1 cards there will be a big pci stress on
> the machine.
>
> I see only true benefit with a dsp card like Pulsar which has asio or any
> other dsp card with asio like Emu or maybe the Nuende or Focusrite ...
>
> If you don't need to use UAD1 on same computer I am sure with one asio audio
> card cubase can be this way intergrated with Paris on same computer with
> very small latency as so to bring in VSTI and other.
>
> Just some thoughts...
> Regards,
> Dimitrios
>
> "chuck duffy" <c@c.com> wrote:
> >
> >Well, there were three things going on.
> >
> >1. I thought chainer didn't allow enough channels, or enough instances.
> >
> >2. I thought the other VST hosts you were using required physical audio
> connections
> >(ie were not virtual) .
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> >>> > >
> >>> > >
> >>> >
> >>>
> >>>
> >>
> >>
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"chuck duffy" <c@c.com> wrote in message news:45424d16$1@linux...

>
> Well, there were three things going on.
>
> 1. I thought chainer didn't allow enough channels, or enough instances.

I've never used Chainer so I don't know.
>
> 2. I thought the other VST hosts you were using required physical audio
connections
> (ie were not virtual) .

Again, I'm not sure.

>
> 3. I thought the other hosts didn't have enough asio channels

The only one that I know of that does is the one by Plogue.

I'm definitely not an authority on these things. I have used Forte as a
standalone VST host. It worked great streaming over ADAT but I ran out of
channels. The one with unlimited channels looks promising, but I don't know
what it requires, other than it will recognize *all* physical I/O and
allow it to be configured as busses of various types.

DJ

>
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> >> > Chuck
> >> >
> >> >
> >> >
> >> > Chuck
> >> > "DJ" <notachance@net.net> wrote:
> >> > >Chuck,
> >> > >
> >> > >There is already an ASIO host application that has unlimited I/O so
> #2
> >> has
> >> > >been covered. It's not simple though
> >> >
> >> http://www.plogue.com/index.php?option=content&task=view &id=21&Itemid=35
> >> > >I haven't tried it yet but will likely get around to it over the
> >weekend
> >> > or
> >> > >early next week. I was thinking of something that could interface
> >> directly
> >> > >with Paris so that the UAD-1 cards could work directly on the Paris
> DAW
> >> > >without having to interface via ADAT on a second workstation. Old
> >Magma's
> >> > >are cheap these days and having the cards in the Paris workstation
> >> running
> >> > >Win XP without having to interface with a second DAW using lightpipe
> >> would
> >> > >be ideal. this is why I was thinking of Wires. As far as an ASIO
> >driver,
> >> > >under the "Paris DAW being host" scenario, without an efficient ASIO
> >> driver,

> >> > >for Paris, I don't see this happening. To tell you the truth, I
haven't
> >> > used
> >> > >the Paris ASIO driver in years. I wonder if it would work with a VST
> >host
> >> > >like Forte or Chainer? I do remember some latency with this driver,
> but
> >> > it's
> >> > >been a long time. Anyway, as far as third party uses for the VST
host
> >> you
> >> > >are proposing in #1 ....I honestly don't know unless they
> >were
> >> > >wanting to stream from a DAW with no latency compensation to a
digital
> >> > >mixer. I don't think there are any DAWs, other than Paris left on
earth
> >> > that
> >> > >don't have latency compensation.
> >> > >
> >> > >If you decide you want to do this, I will support your efforts 100%.
> >> > >
> >> > >Thanks,
> >> > >
> >> > >DJ
> >> > >
> >> > >"chuck duffy" <c@c.com> wrote in message news:454206a9$1@linux...
> >> > >>
> >> > >> Hi DJ,
> >> > >>
> >> > >> I guess what we are talking about is two things:
> >> > >>
> >> > >> 1. An ASIO host application with let's say 64 ins and 64 outs.
This
> >> app
> >> > >> would also be a VST host application that would let you insert
plugs
> >on
> >> > >each
> >> > >> of the 64 ins. It would add up the total latency on each input,
> >buffer
> >> > >the
> >> > >> output to some consistent user entered amount, and send it out the
> >> output.
> >> > >> The latency for every channel would end up being exactly the same
> >user
> >> > >entered
> >> > >> amount.

> >> > >>
> >> > >> 2. A simple VST plugin that would allow you to select an input and
> >> output
> >> > >> ASIO channel. That's all that one would do. This plug wouldn't
> have
> >> > any
> >> > >> latency of it's own.
> >> > >>
> >> > >> So my question is..... Is there any other possible use for such a
> >> setup?
> >> > >> I would be willing to get involved in an open source freeware,
> >> > >ad/donation
> >> > >> supported project for this if there was.
> >> > >>
> >> > >> Chuck
> >> > >>
> >> > >>
> >> > >>
> >> > >>
> >> > >
> >> > >
> >> >
> >>
> >>
> >
> >
>Rich,
My Emu 0404 card is a delight, I've never had a problem.
--
Martin Harrington
www.lendanear-sound.com

"Rich" <studiodog_99@yahoo.com> wrote in message news:453e7140$1@linux...
>
> Anyone useing a 1616 or 1616M as a portable recording solution? If so
> what
> do you think, and what speed laptop are you using? Units have received
> great
> reviews (EM etc) but it's E-mu so something just does not sit right.
>
> Spec's state the card has DSP and I think someone was raving about the
> converters
> at one point here awhile back.
> I'm looking for an inexpensive portable solution to add to my Paris
> system
> not replace.
> Or
> Is the Hamerfall RME DSP mutliface a better solution or??   Thanks again

> for any and all help!!
>Ok, we're down to this. Thanks Craig! Crazy Johnny won't sleep till everything is sold! hehe

>>> c16   75
>>> 8in   200
>>> 8out   200
>>>
>>> FIRST PAYPALS GET IT. $10 shipping for all or $30 to Germany
>>> my paypal is john@kfocus.com
>>
>>Martin, so do you know anything about the Cubase mix bus? Do they maybe mean that on mixdown you pull the faders way back but still record hot?
  Just wondering how the Cubase mix bus behaves.

Thanks

Martin Harrington wrote:
> That's not true, and dont let anyone tell you it is.
> You still need to get all levels as optimised as possible, as we all did
> with tape.
> Otherwise you are not using all the bits available to you, and noise will be
> the end result.
> This is why the good/great engineers are what they are...they make sure the
> levels are hot....just not to the stage of distortion.
> it's a balancing act, but, hey, who said anything done properly is easy.
>OK, if you haven't read my other posts then here is my situation:
I am installing Paris into a new AMD dual core cpu, I had encountered problems with the video card's IRQ setting and I think I have resolved that issue, however I am having a new problem...
I can open and run Paris successfully after initial CPU boot, but once I close it I can't open it again without a rebooting the cpu or I get an error.
Also, once in a while, when I close a song and try to open another song I get a fatal exception. I installed Paris according to the specific instructions and I am using the XP drivers and the subsystem is installed correctly.
What could be causing this?

Thanks,
MikeIs this somehow the equivalent to the getting out of your ya ya's?

"John" <no@no.com> wrote in message news:45426d95@linux...
> Thanks Neil. I now have the following left. I've had a lot of
> inquiries but no MONEY yet, so first come gets it.
>
> eds   150
> c16   75
> 8in   200
> 8out   200

>
> FIRST PAYPALS GET IT. $10 shipping for all or $30 to Germany
> my paypal is john@kfocus.comI know ! It's only Rock and Roll but I like it, like it yes I do.
Ya know I like it !!

DJ wrote:
> Is this somehow the equivalent to the getting out of your ya ya's?
>
> "John" <no@no.com> wrote in message news:45426d95@linux...
>> Thanks Neil. I now have the following left. I've had a lot of
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>> c16   75
>> 8in   200
>> 8out   200
>>
>> FIRST PAYPALS GET IT. $10 shipping for all or $30 to Germany
>> my paypal is john@kfocus.com
>
>I have audio files that were created on a Mac and that are sd2 files that
I want to convert to .wav files to use on a pc audio program.Can you help
me. Thanks.Quietly he emerges with a scant handful of salient posts and then silently
slips back into the black waters. Where ya been man?

W.

"Doug Wellington" <doug@parisfaqs.com> wrote in message
news:454232a1@linux...
> "Gary Flanigan" <gary_flanigan@ce9.uscourts.gov> wrote in message
> news:454221fd$1@linux...
>> I just discovered that two of the three PCI slots on my "new" PC are too
>> short
>> to accomodate MY UAD-1s. So I am considering getting a (probablly used)
>> Magma box. This would use a PCI slot in the PC, not a PCIe slot.
>>
>> Does this work well? Are there cable length restrictions?
>
> I have three Magma chassis, two 13 slot models and a 7 slot. They all
> just work. My only complaint would be that the original fans are quite
> noisy. IIRC, there are a couple different cable lengths, the longest being
> something like three feet. You may find more about that at
> http://www.mobilityelectronics.com/expansion/ ...
>
> Doug
>
> http://www.parisfaqs.com
>Good luck Tommy

Deej

"Tommy Detamore" <cherryrdg@aol.com> wrote in message
news:454228f2$1@linux...
>
> I co-produced and recorded an album done in Paris with an artist named
Sunny
> Sweeney (http://sunnysweeney.com/index.html). Sunny has entries on the
preliminary
> ballot in four categories for the upcoming Grammy Awards:
>
> Best Female Country Vocal Performance
> (52 entries involving 43 artists) for
> Heartbreakers Hall of Fame/Sunny Sweeney
>
> Best Country Collaboration with Vocals
> (38 entries) for
> Lavender Blue/Sunny Sweeney & Jim Lauderdale
>
> Best Country Song
> (129 entries) for
> Heartbreakers Hall of Fame/Sunny Sweeney
>
> Best Country Album
> (83 entries) for
> Heartbreakes Hall of Fame/Sunny Sweeney
>
> The top 5 in each category will advance to the final round of voting.
>
> You can hear clips from the record here:
>
>
http://profile.myspace.com/index.cfm?fuseaction=user.viewpro file&friendid=47
942490
>
> Sunny is a special talent and genuine nice person! Any consideration would
> be much appreciated.....
>
> Thanks,
>
> Tommy
>
> ------------------
> Tommy Detamore
>
> http://www.cherryridgestudio.com
>

>
>Can't tell you anything technically about the Cubase mix bus, (I use
Nuendo),but I think it's basically the same, but it's a fallacy that if you
record at lower levels you are protecting the file from clipping.
What you are doing is not using all the "bits" available to you, and
therefore start introducing unwanted artifacts into the mix.
If the "bits" aren't there on the original recording, and the levelis
cosewuently low at the mix bus, no matter what you do, you can't get those
bits back and the resolution and "size" of your mix has to suffer.
I record as hot as I can, and use the channel faders to mix, usually never
moving the master fader, although having said that, my mixes for TV/ Doco
work are not quite as complicated as most decent size music mixes would be.
--
Martin Harrington
www.lendanear-sound.com
"John" <no@no.com> wrote in message news:45429eda@linux...
> Martin, so do you know anything about the Cubase mix bus? Do they maybe
> mean that on mixdown you pull the faders way back but still record hot?
> Just wondering how the Cubase mix bus behaves.
>
> Thanks
>
> Martin Harrington wrote:
>> That's not true, and dont let anyone tell you it is.
>> You still need to get all levels as optimised as possible, as we all did
>> with tape.
>> Otherwise you are not using all the bits available to you, and noise will
>> be the end result.
>> This is why the good/great engineers are what they are...they make sure
>> the levels are hot....just not to the stage of distortion.
>> it's a balancing act, but, hey, who said anything done properly is easy.
>>hi martin
 I think what john is referring to is what started this thread. We are
trying to emulate the way paris handles files at mixdown, not at recording
files, at mixdown.
 chuck said that paris automatically and transparently cuts channel levels
by -22db, and then adds it back automatically when it hits the submix bus
much the way analog consoles do.
what we were toying with is if was possible to emulate that *effect* with
other daws by cutting channel levels 22db and making it back up at the
output bus.
what we dont know is how cubase/nuendo mix bus handles the files.or exactly
how paris does it for that matter.
If we can duplicate the way paris handles the mix bus DJ can sleep nights
again .
"Martin Harrington" <lendan@bigpond.net.au> wrote in message
news:4542c966$1@linux...
> Can't tell you anything technically about the Cubase mix bus, (I use

> Nuendo),but I think it's basically the same, but it's a fallacy that if
> you record at lower levels you are protecting the file from clipping.
> What you are doing is not using all the "bits" available to you, and
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> moving the master fader, although having said that, my mixes for TV/ Doco
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> --
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> www.lendanear-sound.com
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>>
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>>> This is why the good/great engineers are what they are...they make sure
>>> the levels are hot....just not to the stage of distortion.
>>> it's a balancing act, but, hey, who said anything done properly is easy.
>>>
>
>"Dubya Mark Wilson" <mark.xspam@avidrecording.com> wrote:
> Quietly he emerges with a scant handful of salient posts and then silently
> slips back into the black waters. Where ya been man?

[looks around nervously] Umm... Are you talking to me?

Doug

http://www.parisfaqs.comThanks 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 h