
Subject: Paris Skins - alternatives (3)

Posted by [Yanoska](#) on Fri, 06 Oct 2006 17:43:58 GMT

[View Forum Message](#) <> [Reply to Message](#)

>
> 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.

BrandonThanks, Brandon.

Wrapped means it doesn't 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
>
>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.

"Tony Benson" <tony@standinghampton.com> wrote:

>Neil,

>

>Thanks for posting this. Last night I recorded a test song, drums, guitar,

>and bass in DP. I dropped the individual channel faders to -6.0 and added
a

>limiter (for "make-up" gain and almost no limiting) to the DP main out.

I

>didn't raise any channel fader above -6.0. Only lowered channels to balance

>levels. Even though I only had about 20 channels going, I could already tell
>it was one of the better sounding mixes I've been able to get out of DP.
Can
>it really be this simple? I was so used to maximizing the levels in PARIS

>that I took that methodology over to DP and my mixes in DP always sounded

>"smaller". Now I'm jazzed about doing some more experimentation in DP.
>Thanks again.
>
>Tony
>
>"Neil" <IOUIU@OIU.com> wrote in message news:453d8006\$1@linux...
>>
>> "chuck duffy" <c@c.com> wrote:
>>
>>>If we can't get decent mixes out of a native daw then something is wrong.
>>> Let's find the thing that's wrong, and make it right.
>>
>> (Long, but thought-provoking, and hopefully helpful, rant
>> follows):
>>
>> I think the thing that's wrong is that some people just can't
>> get their heads around the differences between analog & digital.
>> With analog, "big" = hotter, and so hotter is better. When you
>> overbias your tape machines & smack the hell out of the tape,
>> you're getting compression right off the bat on every track you
>> do that with, so one gets used to hearing most tracks with some
>> degree of tape compression already... and we all know that
>> compression can make things sound "bigger". Or, you use a
>> compressor on the way in to the tape so that you get a better
>> SNR, but since that's not an issue with digital (unless you're
>> recording at levels so low that you just simply get poor
>> resolution, but that's a slightly different scenario), people
>> quit using compressors on the way in to digital since SNR isn't
>> an issue there.... you also can't smack an AD convertor hard &
>> expect it to like it - unlike tape. So right off the bat we've
>> got a whole different set of dynamics action going on from one
>> world to the other - then, when you've already got that
>> compressed kick or bassline on tape, you compress it more, and
>> you're compressing an already-compressed signal, so when you
>> apply compression to your uncompressed kick on your DAW you're
>> thinking "nah, that CAN'T be right, it can't need THAT much
>> compression! I'd better back that off a bit!" (because you're
>> looking at the ratios & the threshold, 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 on a processor on it's
>> own card or off your CPU; the difference being how well a
>> particular VST or Direct-X compressor or reverb is written (and
>> what it's designed to do in terms of treating the sound) vs.
>> whatever DSP compressor or reverb plugin you're talking about.
>> Can I get an "Amen, brutha!" on that?
>>
>> Chuck's nailed the Paris mix buss thing, it seems, with that
>> -22db at the channel & +22db at the mix buss, but WHY does that
>> make a difference? Well, here's why gang... it's just as I said
>> earlier in another thread - you've got to give yourself some
>> headroom, dammit! Paris apparently does this for you. Want to
>> prove me wrong? Open up a Paris mix and drag the mix buss
>> master fader down 22db from wherever you have it, then insert

>> any plugin that has an output level control on each individual
>> channel of that mix - if the plugin is a compressor, for
>> example, don't use any compression, just use the output
>> control - now boost every channel by 22db using that output
>> control... if it only goes up 10 db, then insert that plugin
>> twice in a row & max out the output on each insertion...
>> that'll be close enough... how's that sound? I'll bet it won't
>> sound all that good! Are you hearing that "overstuffed" mix
>> buss sound? Is it smaller, with less dimension? I'd be curious
>> to see what you guys think if you try this. Now that we know
>> what Chuck told us he discovered, this is the best way to see
>> if that makes a difference or not (my guess - it DOES make a
>> difference, otherwise, they wouldn't have written the code that
>> way!).
>>
>>
>> So how can you get "big" in Native? Give yourself what Paris
>> apparently already gives you... some headroom - think "clean",
>> then dirty it up if you have to later... hell, just mash the
>> mix with a comp & limiter or an L2 or something equivalent -
>> you'll get all the harmonic distortion you want. I wasn't
>> kidding the other day when I said: "Think zen when mixing in
>> Cubase" it's all gotta flow without clips, gang... think about
>> it... if you have one channel getting "overs" in a 32-bit float-
>> point system, you may not notice it... heck you can't notice
>> each sample in a given sound file can you? Of course not. But
>> if you start adding more channels, and each of those channels
>> is running hot... let's say 32 channels - as a comparison
>> for you guys running two-card paris systems & no native mixes.
>> and let's say you're running hot (over zero) about 25% of the
>> time on each channel - that's 352,000 errors PER SECOND across
>> the 32 tracks. That's a lot of floating-point math going on
>> there, isn't it? And in this scenario, I want you to think of
>> each error as a mistake, because that's what it is... in this
>> style of mixing, it's a mistake. How can you expect something
>> that's got 352,000 mistakes per second going on, to sound good?
>>
>> Are you still not convinced? Then you should also definitely
>> investigate running stems (submixes) & reimporting. When I've
>> done this I definitely can hear a difference, and I suspect you
>> most likely will be able to as well.. it is NOT a huge
>> difference, but it's audible. In fact, some months ago I posted
>> a stems mix vs. a non-stems mix & a number of you said you
>> could hear a difference. Now, if you think "aww, this is just
>> another pain-in-the-ass procedure I have to go through if I mix
>> in Native", keep in mind that you can run 90 Million stems
>> mixes in the time it will take DeeJ to set up his first Pulsar
>> card, and another 900 million in the time that it takes Chuck

>> to research & write that plugin (OK, just giving hell to DeeJ
>> there, and no really no offense intended to Chucks coding
>> capability, but I'm just saying this is something you can do
>> RIGHT NOW, TONIGHT if you want to if you have a Native system,
>> without having to wait for anything new). Now, if you have a
>> small project - one acoustic guitar, piano, & a vocal - with
>> just a few tracks, running stems won't make a difference, but
>> if you have a large project, give it a shot... you may not hear
>> enough of a difference to make it worth doing in any given
>> instance, but then again, you might.
>>
>> So, now that I hope I've made my case, here's my own personal
>> guidelines for Native mixing - try it out & see wat you think:
>>
>> 1.) Do NOT bring down your Master Fader. It stays at zero
>> (unless you're doing a fade).
>>
>> 2.) On your Master inserts, use a peakstop/brickwall limiter
>> set anywhere from -.03 to -3db, depending on how much headroom
>> you want to give your mastering engineer. Settings for volume
>> maximization & other parameters will, of course, depend on the
>> program material.
>>
>> 3.) Record at 24-bit 88.2k or higher (Dan Lavry has a white
>> paper that makes a good case for a 60k sample rate - in order
>> to get the ringing from the convertors' FIR filters out of the
>> top range of our hearing - but since there is no standard 60k
>> sample rate, 88.2 is the next one up). Also, 16-bit may have
>> worked with Paris for whatever reason (maybe it just enhanced
>> the harmonic distortion you're hearing?), but let's face it,
>> everybody knows that more bits = greater "truth", especially
>> when combined with higher resolutions.
>>
>> 4.) Default your individual channel settings to -6db or lower...
>> I find that -6 is a good place to start because you can load up
>> a decent amount of tracks without overloading the mix buss &
>> hitting your limiter too hard at that level. Consider setting
>> it lower as a starting point if you plan on getting into the
>> range of 40+ tracks. HERE'S THE KEY... if you've got your mix
>> roughed out & you can pull out that peakstop limiter I
>> mentioned in #2 & NOT go over zero on the Master - you're
>> golden. Fuck it, set 'em all at -15 as a starting point if you
>> want, Paris is already setting them for you at -22, right? If
>> you're getting a few scant overs without the limiter, you're
>> still ok, really... the idea is not to overstuff the mix buss
>> so heavily that if you pull the limiter off you're going into
>> the +5, +6 range without it.
>>

>> Think "clean" people = think "no clips" (or as few as
>> possible), you get 30-40 channels of "overs" constantly (like
>> the 352,000 of 'em per second in the example I gave earlier),
>> and it's going to get harsh & thin.... it's a cumulative effect.
>>
>> That's it, really... it's just like any other tool - you can't
>> use an allen wrench to properly drive a nail, and you can't use
>> a hammer to trim your nose hair.
>>
>> Happy Native mixing!
>>
>> (think "zen"!)
>>
>> Neil
>
>I believe it works but it cuts off part of the GUI if you don't wrap it. If
my
recollection is recolating properly. It might be a bit more stable wrapped as
well.

Brandon

news:453fba4e@linux...

> Thanks, Brandon.
> Wrapped means it doesn't 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

> >

> >

>

>"Mic Cross" <crzymnmchl@comcast.net> wrote in news:453fc211\$1@linux:

>

> 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.
>

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,
Mikel 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

"Mic Cross" <crzymnmchl@comcast.net> wrote in message
news:453fc211\$1@linux...

>
> 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.

>
>
> "Tony Benson" <tony@standinghampton.com> wrote:

>>Neil,

>>
>>Thanks for posting this. Last night I recorded a test song, drums, guitar,

>
>>and bass in DP. I dropped the individual channel faders to -6.0 and added
> a

>>limiter (for "make-up" gain and almost no limiting) to the DP main out.

> I
>>didn't raise any channel fader above -6.0. Only lowered channels to
>>balance

>
>>levels. Even though I only had about 20 channels going, I could already

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

>>> cutting something by 3, 4, 6db & getting an audible
>>> difference... with digital/plugin EQ's, sometimes you gotta
>>> boost or cut HUGE swaths of that frequency to really make a
>>> difference... why? I think it's a phase thing... you get more
>>> phase shift with analog filters, and so the change is more
>>> apparent at smaller degrees of boost & cut. That also helps to
>>> isolate things to have their own place in the mix at the same
>>> time... considering that phase is the reason we have two ears -
>>> it's the thing that makes it possible for us to tell which
>>> direction a sound is coming from - this makes perfect sense.
>>>
>>> So, those of us (and I think that's "most of us here") who cut
>>> our teeth in the analog world first, and are used to all the
>>> things mentioned above - and who have not changed that style of
>>> mixing - could be disappointed in Native systems - not because
>>> they fall short of analog or Paris, but because they are
>>> actually much more accurate (assuming good quality convertors)
>>> & as a result do not impart certain types of coloration that we
>>> might interpret as "pleasing". If you could go back to a great
>>> mix you did on analog & a console & take out half of the amount
>>> of dynamics processing & half of the amount of EQ'ing you did,
>>> what would you get? A mix that sounded flatter & more colorless
>>> & with less dimension than the one you ended up with. Want
>>> proof? Here it is: If you didn't need the amount of EQ &
>>> dynamics you applied, you wouldn't have done so! If half the
>>> amounts/degrees of those things would have sufficed, that's
>>> what you would have used! So Paris sounds & acts kinda like
>>> analog, and people who like Paris like that aspect of it... how
>>> do we know there's not a few lines of code in there somewhere
>>> that adds graduated degrees of even-harmonic distortion when
>>> you push the faders or saturate the mix buss to whatever
>>> degree? I personally don't think it's strictly a DSP thing,
>>> because let's face it.. a plugin is basically doing the same
>>> thing to your mix whether it's running of a processor on it's
>>> own card or off your CPU; the difference being how well a
>>> particular VST or Direct-X compressor or reverb is written (and
>>> what it's designed to do in terms of treating the sound) vs.
>>> whatever DSP compressor or reverb plugin you're talking about.
>>> Can I get an "Amen, brutha!" on that?
>>>
>>> Chuck's nailed the Paris mix buss thing, it seems, with that
>>> -22db at the channel & +22db at the mix buss, but WHY does that
>>> make a difference? Well, here's why gang... it's just as I said
>>> earlier in another thread - you've got to give yourself some
>>> headroom, dammit! Paris apparently does this for you. Want to
>>> prove me wrong? Open up a Paris mix and drag the mix buss
>>> master fader down 22db from wherever you have it, then insert
>>> any plugin that has an output level control on each individual

>>> channel of that mix - if the plugin is a compressor, for
>>> example, don't use any compression, just use the output
>>> control - now boost every channel by 22db using that output
>>> control... if it only goes up 10 db, then insert that plugin
>>> twice in a row & max out the output on each insertion...
>>> that'll be close enough... how's that sound? I'll bet it won't
>>> sound all that good! Are you hearing that "overstuffed" mix
>>> buss sound? Is it smaller, with less dimension? I'd be curious
>>> to see what you guys think if you try this. Now that we know
>>> what Chuck told us he discovered, this is the best way to see
>>> if that makes a difference or not (my guess - it DOES make a
>>> difference, otherwise, they wouldn't have written the code that
>>> way!).
>>>
>>>
>>> So how can you get "big" in Native? Give yourself what Paris
>>> apparently already gives you... some headroom - think "clean",
>>> then dirty it up if you have to later... hell, just mash the
>>> mix with a comp & limiter or an L2 or something equivalent -
>>> you'll get all the harmonic distortion you want. I wasn't
>>> kidding the other day when I said: "Think zen when mixing in
>>> Cubase" it's all gotta flow without clips, gang... think about
>>> it... if you have one channel getting "overs" in a 32-bit float-
>>> point system, you may not notice it... heck you can't notice
>>> each sample in a given sound file can you? Of course not. But
>>> if you start adding more channels, and each of those channels
>>> is running hot... let's say 32 channels - as a comparison
>>> for you guys running two-card paris systems & no native mixes.
>>> and let's say you're running hot (over zero) about 25% of the
>>> time on each channel - that's 352,000 errors PER SECOND across
>>> the 32 tracks. That's a lot of floating-point math going on
>>> there, isn't it? And in this scenario, I want you to think of
>>> each error as a mistake, because that's what it is... in this
>>> style of mixing, it's a mistake. How can you expect something
>>> that's got 352,000 mistakes per second going on, to sound good?
>>>
>>> Are you still not convinced? Then you should also definitely
>>> investigate running stems (submixes) & reimporting. When I've
>>> done this I definitely can hear a difference, and I suspect you
>>> most likely will be able to as well.. it is NOT a huge
>>> difference, but it's audible. In fact, some months ago I posted
>>> a stems mix vs. a non-stems mix & a number of you said you
>>> could hear a difference. Now, if you think "aww, this is just
>>> another pain-in-the-ass procedure I have to go through if I mix
>>> in Native", keep in mind that you can run 90 Million stems
>>> mixes in the time it will take DeeJ to set up his first Pulsar
>>> card, and another 900 million in the time that it takes Chuck
>>> to research & write that plugin (OK, just giving hell to DeeJ

>>> there, and no really no offense intended to Chucks coding
>>> capability, but I'm just saying this is something you can do
>>> RIGHT NOW, TONIGHT if you want to if you have a Native system,
>>> without having to wait for anything new). Now, if you have a
>>> small project - one acoustic guitar, piano, & a vocal - with
>>> just a few tracks, running stems won't make a difference, but
>>> if you have a large project, give it a shot... you may not hear
>>> enough of a difference to make it worth doing in any given
>>> instance, but then again, you might.
>>>
>>> So, now that I hope I've made my case, here's my own personal
>>> guidelines for Native mixing - try it out & see wat you think:
>>>
>>> 1.) Do NOT bring down your Master Fader. It stays at zero
>>> (unless you're doing a fade).
>>>
>>> 2.) On your Master inserts, use a peakstop/brickwall limiter
>>> set anywhere from -.03 to -3db, depending on how much headroom
>>> you want to give your mastering engineer. Settings for volume
>>> maximization & other parameters will, of course, depend on the
>>> program material.
>>>
>>> 3.) Record at 24-bit 88.2k or higher (Dan Lavry has a white
>>> paper that makes a good case for a 60k sample rate - in order
>>> to get the ringing from the convertors' FIR filters out of the
>>> top range of our hearing - but since there is no standard 60k
>>> sample rate, 88.2 is the next one up). Also, 16-bit may have
>>> worked with Paris for whatever reason (maybe it just enhanced
>>> the harmonic distortion you're hearing?), but let's face it,
>>> everybody knows that more bits = greater "truth", especially
>>> when combined with higher resolutions.
>>>
>>> 4.) Default your individual channel settings to -6db or lower...
>>> I find that -6 is a good place to start because you can load up
>>> a decent amount of tracks without overloading the mix buss &
>>> hitting your limiter too hard at that level. Consider setting
>>> it lower as a starting point if you plan on getting into the
>>> range of 40+ tracks. HERE'S THE KEY... if you've got your mix
>>> roughed out & you can pull out that peakstop limiter I
>>> mentioned in #2 & NOT go over zero on the Master - you're
>>> golden. Fuck it, set 'em all at -15 as a starting point if you
>>> want, Paris is already setting them for you at -22, right? If
>>> you're getting a few scant overs without the limiter, you're
>>> still ok, really... the idea is not to overstuff the mix buss
>>> so heavily that if you pull the limiter off you're going into
>>> the +5, +6 range without it.
>>>
>>> Think "clean" people = think "no clips" (or as few as

>>> possible), you get 30-40 channels of "overs" constantly (like
>>> the 352,000 of 'em per second in the example I gave earlier),
>>> and it's going to get harsh & thin.... it's a cumulative effect.
>>>
>>> That's it, really... it's just like any other tool - you can't
>>> use an allen wrench to properly drive a nail, and you can't use
>>> a hammer to trim your nose hair.
>>>
>>> Happy Native mixing!
>>>
>>> (think "zen"!)
>>>
>>> Neil
>>
>>
>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
>recollection is recolating properly. It might be a bit more stable wrapped
as
>well.
>
>
>Brandon
>
>
>news:453fba4e@linux...
>> Thanks, Brandon.
>> Wrapped means it doesn't 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
>> >
>> >
>>

>>
>
>If it's a stool sample.....well.....never mind.
Rod
"Tom Bruhl" <arpeggio@comcast.net> wrote:
>
>
>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:
> >
> >yes it works. Thank god Waves still code their plugins in Direct-x. =
>Surprisely,
> >the plug added no latency. However, the mix I was working on only had
=>
>10
> >tracks.=20
> >
> >"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><BR>Hi,<=
>BR>The=20
> SSLcompressor has 0 latency.<BR>Thee SSL channel has 1 sample=20
> latency.<BR>Regards,<BR>Dimitrios<BR><BR>"LaMont" <<A=20
> href=3D"mailto:jjdpro@ameritech.net">jjdpro@ameritech.net</A>>=20
> wrote:<BR>><BR>>yes it works. Thank god Waves still code their =
>plugins=20
> in Direct-x. Surprisely,<BR>>the plug added no latency. However, =
>the mix I=20
> was working on only had 10<BR>>tracks. <BR>><BR>>"Goran =
>Stojiljkovic"=20
> <<A=20
> =
>href=3D"mailto:goran.stojiljkovic@os.t-com.hr">goran.stojiljkovic@os.t-co=
>m.hr</A>>=20
> wrote:<BR>>>does it work?<BR>>>latency=20
> ?<BR>>><BR>>>please answer.....=20
> <BR>>><BR>>><BR>><BR></BLOCKQUOTE>
><DIV><FONT size=3D2><BR><BR>I choose Polesoft Lockspam to fight spam, =
>and=20
>you?<BR><A=20
>href=3D"http://www.polesoft.com/refer.html">http://www.polesoft.com/refer=
>
>
>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.

"Tony Benson" <tony@standinghampton.com> wrote:

>I assume it does Mic, but going by ears, things sounded good. I guess I

>don't know for sure how lowering the fader level affects the bit depth in

>DP. It is a question I was wondering about also.

>

>Tony

>

>

>"Mic Cross" <crzymnmchl@comcast.net> wrote in message

>news:453fc211\$1@linux...

>>

>> 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.

>>

>>

>> "Tony Benson" <tony@standinghampton.com> wrote:

>>>Neil,

>>>

>>>Thanks for posting this. Last night I recorded a test song, drums, guitar,

>>

>>>and bass in DP. I dropped the individual channel faders to -6.0 and added

>> a

>>>limiter (for "make-up" gain and almost no limiting) to the DP main out.

>> I

>>>didn't raise any channel fader above -6.0. Only lowered channels to

>>>balance

>>

>>>levels. Even though I only had about 20 channels going, I could already

>> tell

>>>it was one of the better sounding mixes I've been able to get out of DP.

>> Can

>>>it really be this simple? I was so used to maximizing the levels in PARIS

>>

>>>that I took that methodology over to DP and my mixes in DP always sounded

>>

>>>"smaller". Now I'm jazzed about doing some more experimentation in DP.

>>>Thanks again.
>>>
>>>Tony
>>>
>>>"Neil" <IOUIU@OIU.com> wrote in message news:453d8006\$1@linux...
>>>>
>>>>"chuck duffy" <c@c.com> wrote:
>>>>
>>>>>If we can't get decent mixes out of a native daw then something is
>>>>>wrong.
>>>>> Let's find the thing that's wrong, and make it right.
>>>>
>>>> (Long, but thought-provoking, and hopefully helpful, rant
>>>> follows):
>>>>
>>>> I think the thing that's wrong is that some people just can't
>>>> get their heads around the differences between analog & digital.
>>>> With analog, "big" = hotter, and so hotter is better. When you
>>>> overbias your tape machines & smack the hell out of the tape,
>>>> you're getting compression right off the bat on every track you
>>>> do that with, so one gets used to hearing most tracks with some
>>>> degree of tape compression already... and we all know that
>>>> compression can make things sound "bigger". Or, you use a
>>>> compressor on the way in to the tape so that you get a better
>>>> SNR, but since that's not an issue with digital (unless you're
>>>> recording at levels so low that you just simply get poor
>>>> resolution, but that's a slightly different scenario), people
>>>> quit using compressors on the way in to digital since SNR isn't
>>>> an issue there.... you also can't smack an AD convertor hard &
>>>> expect it to like it - unlike tape. So right off the bat we've
>>>> got a whole different set of dynamics action going on from one
>>>> world to the other - then, when you've already got that
>>>> compressed kick or bassline on tape, you compress it more, and
>>>> you're compressing an already-compressed signal, so when you
>>>> apply compression to your uncompressed kick on your DAW you're
>>>> thinking "nah, that CAN'T be right, it can't need THAT much
>>>> compression! I'd better back that off a bit!" (because you're
>>>> looking at the ratios & the threshold, 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 on a processor on it's
>>>> own card or off your CPU; the difference being how well a
>>>> particular VST or Direct-X compressor or reverb is written (and
>>>> what it's designed to do in terms of treating the sound) vs.
>>>> whatever DSP compressor or reverb plugin you're talking about.
>>>> Can I get an "Amen, brutha!" on that?
>>>>
>>>> Chuck's nailed the Paris mix buss thing, it seems, with that
>>>> -22db at the channel & +22db at the mix buss, but WHY does that
>>>> make a difference? Well, here's why gang... it's just as I said
>>>> earlier in another thread - you've got to give yourself some
>>>> headroom, dammit! Paris apparently does this for you. Want to
>>>> prove me wrong? Open up a Paris mix and drag the mix buss
>>>> master fader down 22db from wherever you have it, then insert
>>>> any plugin that has an output level control on each individual
>>>> channel of that mix - if the plugin is a compressor, for
>>>> example, don't use any compression, just use the output
>>>> control - now boost every channel by 22db using that output
>>>> control... if it only goes up 10 db, then insert that plugin
>>>> twice in a row & max out the output on each insertion...
>>>> that'll be close enough... how's that sound? I'll bet it won't
>>>> sound all that good! Are you hearing that "overstuffed" mix
>>>> buss sound? Is it smaller, with less dimension? I'd be curious

>>>> to see what you guys think if you try this. Now that we know
>>>> what Chuck told us he discovered, this is the best way to see
>>>> if that makes a difference or not (my guess - it DOES make a
>>>> difference, otherwise, they wouldn't have written the code that
>>>> way!).
>>>>
>>>>
>>>> So how can you get "big" in Native? Give yourself what Paris
>>>> apparently already gives you... some headroom - think "clean",
>>>> then dirty it up if you have to later... hell, just mash the
>>>> mix with a comp & limiter or an L2 or something equivalent -
>>>> you'll get all the harmonic distortion you want. I wasn't
>>>> kidding the other day when I said: "Think zen when mixing in
>>>> Cubase" it's all gotta flow without clips, gang... think about
>>>> it... if you have one channel getting "overs" in a 32-bit float-
>>>> point system, you may not notice it... heck you can't notice
>>>> each sample in a given sound file can you? Of course not. But
>>>> if you start adding more channels, and each of those channels
>>>> is running hot... let's say 32 channels - as a comparison
>>>> for you guys running two-card paris systems & no native mixes.
>>>> and let's say you're running hot (over zero) about 25% of the
>>>> time on each channel - that's 352,000 errors PER SECOND across
>>>> the 32 tracks. That's a lot of floating-point math going on
>>>> there, isn't it? And in this scenario, I want you to think of
>>>> each error as a mistake, because that's what it is... in this
>>>> style of mixing, it's a mistake. How can you expect something
>>>> that's got 352,000 mistakes per second going on, to sound good?
>>>>
>>>> Are you still not convinced? Then you should also definitely
>>>> investigate running stems (submixes) & reimporting. When I've
>>>> done this I definitely can hear a difference, and I suspect you
>>>> most likely will be able to as well.. it is NOT a huge
>>>> difference, but it's audible. In fact, some months ago I posted
>>>> a stems mix vs. a non-stems mix & a number of you said you
>>>> could hear a difference. Now, if you think "aww, this is just
>>>> another pain-in-the-ass procedure I have to go through if I mix
>>>> in Native", keep in mind that you can run 90 Million stems
>>>> mixes in the time it will take DeeJ to set up his first Pulsar
>>>> card, and another 900 million in the time that it takes Chuck
>>>> to research & write that plugin (OK, just giving hell to DeeJ
>>>> there, and no really no offense intended to Chucks coding
>>>> capability, but I'm just saying this is something you can do
>>>> RIGHT NOW, TONIGHT if you want to if you have a Native system,
>>>> without having to wait for anything new). Now, if you have a
>>>> small project - one acoustic guitar, piano, & a vocal - with
>>>> just a few tracks, running stems won't make a difference, but
>>>> if you have a large project, give it a shot... you may not hear
>>>> enough of a difference to make it worth doing in any given

>>>> instance, but then again, you might.
>>>>
>>>> So, now that I hope I've made my case, here's my own personal
>>>> guidelines for Native mixing - try it out & see wat you think:
>>>>
>>>> 1.) Do NOT bring down your Master Fader. It stays at zero
>>>> (unless you're doing a fade).
>>>>
>>>> 2.) On your Master inserts, use a peakstop/brickwall limiter
>>>> set anywhere from -.03 to -3db, depending on how much headroom
>>>> you want to give your mastering engineer. Settings for volume
>>>> maximization & other parameters will, of course, depend on the
>>>> program material.
>>>>
>>>> 3.) Record at 24-bit 88.2k or higher (Dan Lavry has a white
>>>> paper that makes a good case for a 60k sample rate - in order
>>>> to get the ringing from the convertors' FIR filters out of the
>>>> top range of our hearing - but since there is no standard 60k
>>>> sample rate, 88.2 is the next one up). Also, 16-bit may have
>>>> worked with Paris for whatever reason (maybe it just enhanced
>>>> the harmonic distortion you're hearing?), but let's face it,
>>>> everybody knows that more bits = greater "truth", especially
>>>> when combined with higher resolutions.
>>>>
>>>> 4.) Default your individual channel settings to -6db or lower...
>>>> I find that -6 is a good place to start because you can load up
>>>> a decent amount of tracks without overloading the mix buss &
>>>> hitting your limiter too hard at that level. Consider setting
>>>> it lower as a starting point if you plan on getting into the
>>>> range of 40+ tracks. HERE'S THE KEY... if you've got your mix
>>>> roughed out & you can pull out that peakstop limiter I
>>>> mentioned in #2 & NOT go over zero on the Master - you're
>>>> golden. Fuck it, set 'em all at -15 as a starting point if you
>>>> want, Paris is already setting them for you at -22, right? If
>>>> you're getting a few scant overs without the limiter, you're
>>>> still ok, really... the idea is not to overstuff the mix buss
>>>> so heavily that if you pull the limiter off you're going into
>>>> the +5, +6 range without it.
>>>>
>>>> Think "clean" people = think "no clips" (or as few as
>>>> possible), you get 30-40 channels of "overs" constantly (like
>>>> the 352,000 of 'em per second in the example I gave earlier),
>>>> and it's going to get harsh & thin.... it's a cumulative effect.
>>>>
>>>> That's it, really... it's just like any other tool - you can't
>>>> use an allen wrench to properly drive a nail, and you can't use
>>>> a hammer to trim your nose hair.
>>>>

>>>> Happy Native mixing!

>>>>

>>>> (think "zen"!)

>>>>

>>>> Neil

>>>

>>>

>>

>

>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.

>

>
>
> "Tony Benson" <tony@standinghampton.com> wrote:
>>I assume it does Mic, but going by ears, things sounded good. I guess I
>
>>don't know for sure how lowering the fader level affects the bit depth in
>
>>DP. It is a question I was wondering about also.
>>
>>Tony
>>
>>
>>"Mic Cross" <crzymnmchl@comcast.net> wrote in message
>>news:453fc211\$1@linux...
>>>
>>> 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.
>>>
>>>
>>> "Tony Benson" <tony@standinghampton.com> wrote:
>>>>Neil,
>>>>
>>>>Thanks for posting this. Last night I recorded a test song, drums,
>>>>guitar,
>>>
>>>>and bass in DP. I dropped the individual channel faders to -6.0 and
>>>>added
>>> a
>>>>limiter (for "make-up" gain and almost no limiting) to the DP main out.
>>> I
>>>>didn't raise any channel fader above -6.0. Only lowered channels to
>>>>balance
>>>
>>>>levels. Even though I only had about 20 channels going, I could already
>>> tell
>>>>it was one of the better sounding mixes I've been able to get out of DP.
>>> Can
>>>>it really be this simple? I was so used to maximizing the levels in
>>>>PARIS
>>>
>>>>that I took that methodology over to DP and my mixes in DP always
>>>>sounded
>>>
>>>>"smaller". Now I'm jazzed about doing some more experimentation in DP.
>>>>Thanks again.

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

>>>>> direction a sound is coming from - this makes perfect sense.
>>>>>
>>>>> So, those of us (and I think that's "most of us here") who cut
>>>>> our teeth in the analog world first, and are used to all the
>>>>> things mentioned above - and who have not changed that style of
>>>>> mixing - could be disappointed in Native systems - not because
>>>>> they fall short of analog or Paris, but because they are
>>>>> actually much more accurate (assuming good quality convertors)
>>>>> & as a result do not impart certain types of coloration that we
>>>>> might interpret as "pleasing". If you could go back to a great
>>>>> mix you did on analog & a console & take out half of the amount
>>>>> of dynamics processing & half of the amount of EQ'ing you did,
>>>>> what would you get? A mix that sounded flatter & more colorless
>>>>> & with less dimension than the one you ended up with. Want
>>>>> proof? Here it is: If you didn't need the amount of EQ &
>>>>> dynamics you applied, you wouldn't have done so! If half the
>>>>> amounts/degrees of those things would have sufficed, that's
>>>>> what you would have used! So Paris sounds & acts kinda like
>>>>> analog, and people who like Paris like that aspect of it... how
>>>>> do we know there's not a few lines of code in there somewhere
>>>>> that adds graduated degrees of even-harmonic distortion when
>>>>> you push the faders or saturate the mix buss to whatever
>>>>> degree? I personally don't think it's strictly a DSP thing,
>>>>> because let's face it.. a plugin is basically doing the same
>>>>> thing to your mix whether it's running of a processor on it's
>>>>> own card or off your CPU; the difference being how well a
>>>>> particular VST or Direct-X compressor or reverb is written (and
>>>>> what it's designed to do in terms of treating the sound) vs.
>>>>> whatever DSP compressor or reverb plugin you're talking about.
>>>>> Can I get an "Amen, brutha!" on that?
>>>>>
>>>>> Chuck's nailed the Paris mix buss thing, it seems, with that
>>>>> -22db at the channel & +22db at the mix buss, but WHY does that
>>>>> make a difference? Well, here's why gang... it's just as I said
>>>>> earlier in another thread - you've got to give yourself some
>>>>> headroom, dammit! Paris apparently does this for you. Want to
>>>>> prove me wrong? Open up a Paris mix and drag the mix buss
>>>>> master fader down 22db from wherever you have it, then insert
>>>>> any plugin that has an output level control on each individual
>>>>> channel of that mix - if the plugin is a compressor, for
>>>>> example, don't use any compression, just use the output
>>>>> control - now boost every channel by 22db using that output
>>>>> control... if it only goes up 10 db, then insert that plugin
>>>>> twice in a row & max out the output on each insertion...
>>>>> that'll be close enough... how's that sound? I'll bet it won't
>>>>> sound all that good! Are you hearing that "overstuffed" mix
>>>>> buss sound? Is it smaller, with less dimension? I'd be curious
>>>>> to see what you guys think if you try this. Now that we know

>>>>> what Chuck told us he discovered, this is the best way to see
>>>>> if that makes a difference or not (my guess - it DOES make a
>>>>> difference, otherwise, they wouldn't have written the code that
>>>>> way!).
>>>>>
>>>>>
>>>>> So how can you get "big" in Native? Give yourself what Paris
>>>>> apparently already gives you... some headroom - think "clean",
>>>>> then dirty it up if you have to later... hell, just mash the
>>>>> mix with a comp & limiter or an L2 or something equivalent -
>>>>> you'll get all the harmonic distortion you want. I wasn't
>>>>> kidding the other day when I said: "Think zen when mixing in
>>>>> Cubase" it's all gotta flow without clips, gang... think about
>>>>> it... if you have one channel getting "overs" in a 32-bit float-
>>>>> point system, you may not notice it... heck you can't notice
>>>>> each sample in a given sound file can you? Of course not. But
>>>>> if you start adding more channels, and each of those channels
>>>>> is running hot... let's say 32 channels - as a comparison
>>>>> for you guys running two-card paris systems & no native mixes.
>>>>> and let's say you're running hot (over zero) about 25% of the
>>>>> time on each channel - that's 352,000 errors PER SECOND across
>>>>> the 32 tracks. That's a lot of floating-point math going on
>>>>> there, isn't it? And in this scenario, I want you to think of
>>>>> each error as a mistake, because that's what it is... in this
>>>>> style of mixing, it's a mistake. How can you expect something
>>>>> that's got 352,000 mistakes per second going on, to sound good?
>>>>>
>>>>> Are you still not convinced? Then you should also definitely
>>>>> investigate running stems (submixes) & reimporting. When I've
>>>>> done this I definitely can hear a difference, and I suspect you
>>>>> most likely will be able to as well.. it is NOT a huge
>>>>> difference, but it's audible. In fact, some months ago I posted
>>>>> a stems mix vs. a non-stems mix & a number of you said you
>>>>> could hear a difference. Now, if you think "aww, this is just
>>>>> another pain-in-the-ass procedure I have to go through if I mix
>>>>> in Native", keep in mind that you can run 90 Million stems
>>>>> mixes in the time it will take DeeJ to set up his first Pulsar
>>>>> card, and another 900 million in the time that it takes Chuck
>>>>> to research & write that plugin (OK, just giving hell to DeeJ
>>>>> there, and no really no offense intended to Chucks coding
>>>>> capability, but I'm just saying this is something you can do
>>>>> RIGHT NOW, TONIGHT if you want to if you have a Native system,
>>>>> without having to wait for anything new). Now, if you have a
>>>>> small project - one acoustic guitar, piano, & a vocal - with
>>>>> just a few tracks, running stems won't make a difference, but
>>>>> if you have a large project, give it a shot... you may not hear
>>>>> enough of a difference to make it worth doing in any given
>>>>> instance, but then again, you might.

>>>>>
>>>>> So, now that I hope I've made my case, here's my own personal
>>>>> guidelines for Native mixing - try it out & see wat you think:
>>>>>
>>>>> 1.) Do NOT bring down your Master Fader. It stays at zero
>>>>> (unless you're doing a fade).
>>>>>
>>>>> 2.) On your Master inserts, use a peakstop/brickwall limiter
>>>>> set anywhere from -.03 to -3db, depending on how much headroom
>>>>> you want to give your mastering engineer. Settings for volume
>>>>> maximization & other parameters will, of course, depend on the
>>>>> program material.
>>>>>
>>>>> 3.) Record at 24-bit 88.2k or higher (Dan Lavry has a white
>>>>> paper that makes a good case for a 60k sample rate - in order
>>>>> to get the ringing from the convertors' FIR filters out of the
>>>>> top range of our hearing - but since there is no standard 60k
>>>>> sample rate, 88.2 is the next one up). Also, 16-bit may have
>>>>> worked with Paris for whatever reason (maybe it just enhanced
>>>>> the harmonic distortion you're hearing?), but let's face it,
>>>>> everybody knows that more bits = greater "truth", especially
>>>>> when combined with higher resolutions.
>>>>>
>>>>> 4.) Default your individual channel settings to -6db or lower...
>>>>> I find that -6 is a good place to start because you can load up
>>>>> a decent amount of tracks without overloading the mix buss &
>>>>> hitting your limiter too hard at that level. Consider setting
>>>>> it lower as a starting point if you plan on getting into the
>>>>> range of 40+ tracks. HERE'S THE KEY... if you've got your mix
>>>>> roughed out & you can pull out that peakstop limiter I
>>>>> mentioned in #2 & NOT go over zero on the Master - you're
>>>>> golden. Fuck it, set 'em all at -15 as a starting point if you
>>>>> want, Paris is already setting them for you at -22, right? If
>>>>> you're getting a few scant overs without the limiter, you're
>>>>> still ok, really... the idea is not to overstuff the mix buss
>>>>> so heavily that if you pull the limiter off you're going into
>>>>> the +5, +6 range without it.
>>>>>
>>>>> Think "clean" people = think "no clips" (or as few as
>>>>> possible), you get 30-40 channels of "overs" constantly (like
>>>>> the 352,000 of 'em per second in the example I gave earlier),
>>>>> and it's going to get harsh & thin.... it's a cumulative effect.
>>>>>
>>>>> That's it, really... it's just like any other tool - you can't
>>>>> use an allen wrench to properly drive a nail, and you can't use
>>>>> a hammer to trim your nose hair.
>>>>>
>>>>> Happy Native mixing!

>>>>>

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

>>>>>

>>>>> Neil

>>>>

>>>>

>>>

>>

>>

>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 notably 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 doesn't 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,
how did you install paris on xp, why is the sky blue, how many DJ's does
it take to fill a mix bus, ponder.....ponder.....

Mike P wrote:

> 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,

> 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" <arpeggio@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
>Surprisingly,
> >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>What's a sample among =3D
>friends?</DIV>
><DIV>T.</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> wrote =
in =3D
>message=3D20
> <A=3D20
> =3D
=

>href=3D3D"news:453f8b69\$1@linux">news:453f8b69\$1@linux...</DIV>
H=
i,<=3D

```

>BR>The=3D20
> SSLcompressor has 0 latency.<BR>Thee SSL channel has 1 sample=3D20
> latency.<BR>Regards,<BR>Dimitrios<BR><BR>"LaMont" <<A=3D20
> =
href=3D3D"mailto:jjdpro@ameritech.net">jjdpro@ameritech.net</A>>=3D20
> wrote:<BR>><BR>>yes it works. Thank god Waves still code their =3D
>plugins=3D20
> in Direct-x. Surprisely,<BR>>the plug added no latency. However, =
=3D
>the mix I=3D20
> was working on only had 10<BR>>tracks. <BR>><BR>>"Goran =3D
>Stojiljkovic"=3D20
> <<A=3D20
> =3D
=
>href=3D3D"mailto:goran.stojiljkovic@os.t-com.hr">goran.stojiljkovic@os.t=
-co=3D
>m.hr</A>>=3D20
> wrote:<BR>>>does it work?<BR>>>latency=3D20
> ?<BR>>><BR>>>please answer.....=3D20
> <BR>>><BR>>><BR>><BR></BLOCKQUOTE>
><DIV><FONT size=3D3D2><BR><BR>I choose Polesoft Lockspam to fight =
spam, =3D
>and=3D20
>you?<BR><A=3D20
=
>href=3D3D"http://www.polesoft.com/refer.html">http://www.polesoft.com/re=
fer=3D
>.html</A> </FONT></DIV></BODY></HTML>
>
>

```

-----=_NextPart_000_0036_01C6F862.01B217D0

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>oh.</FONT></DIV>
<BLOCKQUOTE=20
style=3D"PADDING-RIGHT: 0px; PADDING-LEFT: 5px; MARGIN-LEFT: 5px; =

```

BORDER-LEFT: #000000 2px solid; MARGIN-RIGHT: 0px">
 <DIV>"Rod Lincoln" <<A=20
 =
 href=3D"mailto:rlincoln@nospam.kc.rr.com">rlincoln@nospam.kc.rr.com&g=
 t;=20
 wrote in message <A=20
 =
 href=3D"news:453fd252\$1@linux">news:453fd252\$1@linux...</DIV>
If =
 it's a=20
 stool sample.....well.....never mind.
Rod
"Tom =
 Bruhl" <<A=20
 href=3D"mailto:arpeggio@comcast.net">arpeggio@comcast.net>=20
 wrote:
>
>
>What's a sample among=20
 friends?
>T.
>
> "Dimitrios" <<A=20
 href=3D"mailto:musurgio@otenet.gr">musurgio@otenet.gr> wrote in =
 message=20
 =3D
>news:453f8b69\$1@linux...
>
> =
 Hi,
> The=20
 SSLcompressor has 0 latency.
> Thee SSL channel has 1 =
 sample=20
 latency.
> Regards,
> =
 Dimitrios
>
> =20
 "LaMont" <<A=20
 href=3D"mailto:jjdpro@ameritech.net">jjdpro@ameritech.net>=20
 wrote:
> >
> >yes it works. Thank god =
 Waves=20
 still code their plugins in Direct-x. =
 =3D
>Surprisely,
> =20
 >the plug added no latency. However, the mix I was working on only=20
 had
=3D
>10
> >tracks.=3D20
> =20
 >
> >"Goran Stojiljkovic" <<A=20
 =
 href=3D"mailto:goran.stojiljkovic@os.t-com.hr">goran.stojiljkovic@os.t-co=
 m.hr>=20
 wrote:
> >>does it work?
> =
 >>latency=20
 ?
> >>
> >>please=20
 answer.....=3D20
> >>
> =
 >>
> =20
 >
>
>
>I choose Polesoft Lockspam to =
 fight spam,=20
 and you?
>http://www.polesoft.com/refer.html =20

>
><!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


```

>=3D<BR>&gt;m.hr&lt;/A&gt;&gt;=3D20<BR>&gt;&nbsp;=20
wrote:&lt;BR&gt;&gt;&gt;&gt;does it=20
work?&lt;BR&gt;&gt;&gt;&gt;latency=3D20<BR>&gt;&nbsp;=20
?&lt;BR&gt;&gt;&gt;&gt;&lt;BR&gt;&gt; ;&gt;please =
answer.....=3D20<BR>&gt;&nbsp;=20
=
&lt;BR&gt;&gt;&gt;&gt;&lt;BR&gt;&gt; &gt;&lt;BR&gt;&gt;&lt;BR&gt;&lt; /BLOCKQUO=
TE&gt;<BR>&gt;&lt;DIV&gt;&lt;FONT=20
size=3D3D2&gt;&lt;BR&gt;&lt;BR&gt;I choose Polesoft Lockspam to fight =
spam,=20
=
=3D<BR>&gt;and=3D20<BR>&gt;you?&lt;BR&gt;&lt;A=3D20 <BR>&gt;href=3D3D"<A=20
=
href=3D'http://www.polesoft.com/refer.html">http://www.polesoft.com/refer=
'>http://www.polesoft.com/refer.html"&gt;http://www.polesoft.com/refer</A=
>=3D<BR>&gt;.html&lt;/A&gt;=20
=
&lt;/FONT&gt;&lt;/DIV&gt;&lt;/BODY&gt;&lt;/HTML&gt; <BR>&gt;<BR>&gt;<BR></=
BLOCKQUOTE></BODY></HTML>

```

-----=_NextPart_000_0036_01C6F862.01B217D0--2EDS filling 2 of three PCI's, the other has a second video card.

Amd 2.2 X2 proc, 2gig ram, running XP home.

I cant get to my Paris cfg file at the moment that PC is not online. What should I be looking for there?

I followed the "installing paris on XP" guidelines. I have the pc "optimized" for audio apps.

I can run sessions using eds plugs fine, as soon as I add native plugs it crashes.

John <no@no.com> wrote:

>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,

>how did you install paris on xp, why is the sky blue, how many DJ's does

>it take to fill a mix bus, ponder.....ponder.....

>

>Mike P wrote:

>> 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,
>> MikeHere is a typical Paris.cfg with more here (www.kfocus.com/paris). My money is on the irq conflicts on the motherboard between video cards and EDS. What motherboard is it and what slots share irqs?
John

Mike P wrote:

> 2EDS filling 2 of three PCI's, the other has a second video card.
> Amd 2.2 X2 proc, 2gig ram, running XP home.
> I cant get to my Paris cfg file at the moment that PC is not online. What
> should I be looking for there?
> I followed the "installing paris on XP" guidelines. I have the pc "optimized"
> for audio apps.
> I can run sessions using eds plugs fine, as soon as I add native plugs it
> crashes.
>
>
>
>
> John <no@no.com> wrote:
>> 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,
>
>> how did you install paris on xp, why is the sky blue, how many DJ's does
>
>> it take to fill a mix bus, ponder.....ponder.....
>>
>> Mike P wrote:
>>> 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,
>>> 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

VSTDDirectory=C:\vsts\

MIDIPlayDisabled=1

Mike P wrote:

> 2EDS filling 2 of three PCI's, the other has a second video card.

> Amd 2.2 X2 proc, 2gig ram, running XP home.

> I cant get to my Paris cfg file at the moment that PC is not online. What

> should I be looking for there?

> I followed the "installing paris on XP" guidelines. I have the pc "optimized"

> for audio apps.

> I can run sessions using eds plugs fine, as soon as I add native plugs it

> crashes.

>
>
>
>
> John <no@no.com> wrote:
>> 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,
>
>> how did you install paris on xp, why is the sky blue, how many DJ's does
>
>> it take to fill a mix bus, ponder.....ponder.....
>>
>> Mike P wrote:
>>> 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,
>>> 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

John <no@no.com> wrote:
>Here is a typical Paris.cfg with more here (www.kfocus.com/paris). My
>money is on the irq conflicts on the motherboard between video cards and

>EDS. What motherboard is it and what slots share irqs?
>John
>
>Mike P wrote:
>> 2EDS filling 2 of three PCI's, the other has a second video card.
>> Amd 2.2 X2 proc, 2gig ram, running XP home.
>> I cant get to my Paris cfg file at the moment that PC is not online. What
>> should I be looking for there?
>> I followed the "installing paris on XP" guidelines. I have the pc "optimized"
>> for audio apps.

>> I can run sessions using eds plugs fine, as soon as I add native plugs
it
>> crashes.
>>
>>
>>
>>
>> John <no@no.com> wrote:
>>> 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,
>>
>>> how did you install paris on xp, why is the sky blue, how many DJ's does
>>
>>> it take to fill a mix bus, ponder.....ponder.....
>>>
>>> Mike P wrote:
>>>> 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,
>>>> Mike
>>Were you previously running Paris on 98 or ME with no problems? If so, its
probably not an IRQ issue.
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
location as the Paris exe file.
Rod
"Mike P" <mikep@4hometown.com> wrote:
>
>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,
>Mike If I installed it in the wrong location would I still see all the plugins?

"Rod Lincoln" <rlincoln@nosopam.kc.rr.com> wrote:

>Wou

>Were you previously running Paris on 98 or ME with no problems? If so, its
>probably not an IRQ issue.

>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
>location as the Paris exe file.

>Rod

>"Mike P" <mikep@4hometown.com> wrote:

>>

>>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,

>>Mike

>Here Don

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

hit the links page for impulses

Rod

"Don Nafe" <dnafe@magma.ca> wrote:

>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 doesn't 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
 >>>> >
 >>>> >
 >>>>
 >>>>
 >>>
 >>>
 >>
 >
 >

>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&highlight=clipping>also 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:

> 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&highlight=clipping>"alex plasko"
<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.

Genewere you running win98 or ME just fine with this same config? Also, yes you'd everythin would work normally up to the crash.

mike P wrote:

> If I installed it in the wrong location would I still see all the plugins?

>

>

>

>

>

> "Rod Lincoln" <rlincoln@nosopam.kc.rr.com> wrote:

>> Wou

>> Were you previously running Paris on 98 or ME with no problems? If so, its
>> probably not an IRQ issue.

>> 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
>> location as the Paris exe file.

>> Rod

>> "Mike P" <mikep@4hometown.com> wrote:

>>> 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,

>>> Mike

>I was running 98se on a different machine and it worked well.

The only changes I made in the config on this machine was to run
256 I/O and 256 cashe

I ran 96/96 on the 98se machine.

All Paris functions are working fine, and I can run native plugins fine,
but when I try to exit that song and open another Paris will not play and
creates an error. Then I have not only reboot Paris, but I have to reboot
the computer too.

John <no@no.com> wrote:

>were you running win98 or ME just fine with this same config? Also, yes

>you'd everythin would work normally up to the crash.

>

>mike P wrote:

>> If I installed it in the wrong location would I still see all the plugins?

>>
>>
>>
>>
>>
>> "Rod Lincoln" <rlincoln@nosopam.kc.rr.com> wrote:
>>> Wou
>>> Were you previously running Paris on 98 or ME with no problems? If so,
its
>>> probably not an IRQ issue.
>>> 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
>>> location as the Paris exe file.
>>> Rod
>>> "Mike P" <mikep@4hometown.com> wrote:
>>>> 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,
>>>> Mike
>>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

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
> "Don Nafe" <dnafe@magma.ca> wrote:
>>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
>>>>recollection is recolating properly. It might be a bit more stable
>>>>wrapped
>>> as
>>>>well.
>>>>
>>>>
>>>>Brandon
>>>>
>>>>
>>>>news:453fba4e@linux...
>>>>> Thanks, Brandon.
>>>>> Wrapped means it doesn't work as a 'regular' VSt plug ? i need a
>>>>> wrapper,
>>>>> correct ?
>>>>>
>>>>> 453fb902\$1@linux...
>>>>> > WORKS GREAT WRAPPED.


```

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-virus-free"
e"> http://free.grisoft.com/doc/5390/lng/us/tpl/v5#avg-anti-virus-free</A>=
</FONT></DIV>
<DIV><FONT face=3DArial size=3D2></FONT><FONT face=3DArial =
size=3D2></FONT>&nbsp;</DIV>
<DIV><FONT face=3DArial size=3D2></FONT>&nbsp;</DIV>
<DIV><FONT face=3DArial size=3D2>It's always worked perfectly for me, so =
I have no=20
reason to switch my machines off from it. </FONT></DIV>
<DIV><FONT face=3DArial size=3D2>AA</FONT></DIV>
<DIV><FONT face=3DArial size=3D2></FONT>&nbsp;</DIV>
<DIV><FONT face=3DArial size=3D2></FONT>&nbsp;</DIV>
<DIV><FONT face=3DArial size=3D2>"John" &lt;</FONT><A =
href=3D"mailto:no@no.com"><FONT=20
face=3DArial size=3D2>no@no.com</FONT></A><FONT face=3DArial =
size=3D2>&gt; wrote in=20
message </FONT><A href=3D"news:453f7707@linux"><FONT face=3DArial=20
size=3D2>news:453f7707@linux</FONT></A><FONT face=3DArial=20
size=3D2>...</FONT></DIV><FONT face=3DArial size=3D2>&gt;I meant =
antivirus.&nbsp;<=20
sorry<BR>&gt; <BR>&gt; John wrote:<BR>&gt;&gt; Whatcha all=20
think?<BR>&gt;&gt; <BR>&gt;&gt; </FONT><A=20
href=3D" http://www.microsoft.com/athome/security/spyware/software/default=
..mspx"><FONT=20
face=3DArial=20
size=3D2> http://www.microsoft.com/athome/security/spyware/software/default=
t.mspx</FONT></A>=20
</BODY></HTML>

```

-----=_NextPart_000_004F_01C6F872.9CF2DD00--C.O.E. is a typical symptom of not installing the subsystem correctly. I'd try re-installing your subsystem, making sure you direct it to the folder that contains the Paris exe

Rod

"Mike P" <mikep@4hometown.com> wrote:

>

>I was running 98se on a different machine and it worked well.

>The only changes I made in the config on this machine was to run

>256 I/O and 256 cache

>I ran 96/96 on the 98se machine.

>All Paris functions are working fine, and I can run native plugins fine,

>but when I try to exit that song and open another Paris will not play and
>creates an error. Then I have not only reboot Paris, but I have to reboot
>the computer too.

>

>John <no@no.com> wrote:

>>were you running win98 or ME just fine with this same config? Also, yes

>

>>you'd everythin would work normally up to the crash.

>>

>>mike P wrote:

>>> If I installed it in the wrong location would I still see all the plugins?

>>>

>>>

>>>

>>>

>>>

>>> "Rod Lincoln" <rlincoln@nosopam.kc.rr.com> wrote:

>>>> Wou

>>>> Were you previously running Paris on 98 or ME with no problems? If so,

>>>> its

>>>> probably not an IRQ issue.

>>>> 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

>>>> location as the Paris exe file.

>>>> Rod

>>>> "Mike P" <mikep@4hometown.com> wrote:

>>>>> 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,

>>>>> 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...

>

> 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

>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.

"Rod Lincoln" <rlincoln@nospam.kc.rr.com> wrote:

>

>C.O.E. is a typical symptom of not installing the subsystem correctly. I'd

>try re-installing your subsystem, making sure you direct it to the folder

>that contains the Paris exe

>Rod

>"Mike P" <mikep@4hometown.com> wrote:

>>

>>I was running 98se on a different machine and it worked well.

>>The only changes I made in the config on this machine was to run

>>256 I/O and 256 cashe

>>I ran 96/96 on the 98se machine.

>>All Paris functions are working fine, and I can run native plugins fine,

>>but when I try to exit that song and open another Paris will not play and

>>creates an error. Then I have not only reboot Paris, but I have to reboot

>>the computer too.

>>

>>John <no@no.com> wrote:

>>>were you running win98 or ME just fine with this same config? Also, yes

>>

>>>you'd everythin would work normally up to the crash.

>>>
>>>mike P wrote:
>>>> If I installed it in the wrong location would I still see all the plugins?
>>>>
>>>>
>>>>
>>>>
>>>>
>>>> "Rod Lincoln" <rlincoln@nosopam.kc.rr.com> wrote:
>>>>> Wou
>>>>> Were you previously running Paris on 98 or ME with no problems? If
so,
>>its
>>>>> probably not an IRQ issue.
>>>>> 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
>>>>> location as the Paris exe file.
>>>>> Rod
>>>>> "Mike P" <mikep@4hometown.com> wrote:
>>>>>> 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,
>>>>>> Mike
>>>>>
>>
>"Gene Lennon" <glennon@NOSPmyrealbox.com> wrote:
>
>"alex plasko" <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.
>Gene

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

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.

Genelt'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}How do you wrap your VST plugs ?
(newbee style question)

4540087e@linux...

> Thanks 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

>> "Don Nafe" <dnafe@magma.ca> wrote:

>>>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.

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

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

Doug

<http://www.parisfaqs.com>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}
>
>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}
>>
>>
>This is a multi-part message in MIME format.

-----=_NextPart_000_00C2_01C6F8B1.A16B6E30
Content-Type: text/plain;
charset="iso-8859-1"
Content-Transfer-Encoding: quoted-printable

Kim,
Hopefully more than just your brain.
Tom
"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}

>>

>>

>

I choose Polesoft Lockspam to fight spam, and you?

<http://www.polesoft.com/refer.html>

-----=_NextPart_000_00C2_01C6F8B1.A16B6E30

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>Kim,</DIV>
 <DIV>Hopefully more than just your =
 brain.</DIV>
 <DIV>Tom</DIV>
 <BLOCKQUOTE=20
 style=3D"PADDING-RIGHT: 0px; PADDING-LEFT: 5px; MARGIN-LEFT: 5px; =
 BORDER-LEFT: #000000 2px solid; MARGIN-RIGHT: 0px">
 <DIV>"Kim" <<A=20
 =
 href=3D"mailto:hiddensounds@hotmail.com">hiddensounds@hotmail.com>=
 wrote=20
 in message <A=20
 =
 href=3D"news:45405600\$1@linux">news:45405600\$1@linux...</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>=
 =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>=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 =
 about 3=20
 days before things started going south. CRAP!!!!....I
>>hope =

the=20
video cards and the mobo aren't all damaged. I need to take=20
an
>>electrical engineering course.....I'm gonna get a gimme =
hat with=20
"dumbass"
>>in big block=20
=
letters.
>>
>>(sigh)
>>
>>;o}
>=
>
>>
>
</BLOCKQUOTE>
<DIV>

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 </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)

> >>

> >>;0}
> >>
> >>
> >>
> >

>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,
> but 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)
>
> "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}

>> >>

>> >>

>> >

>>

>

>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 significant
>> 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 Apogeess 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 involved 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 whilst 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 latencywell.....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 !!

---=_linux4540d2c9

Content-Type: image/jpeg; name="millingmachines1_02.jpg"

Content-Transfer-Encoding: base64

Content-Disposition: attachment; filename="millingmachines1_02.jpg"

/9j/4AAQSkZJRgABAQAAZABkAAD/7AARRHVja3kAAQAEAAAAPAAA/+4ADkFk
b2JlAGTAAAAAAf/bAlQABgQEBAUEBgUFBGkGBQYJCwgGBggLDAoKCwoKDBAM
DAwMDAwQDA4PEA8ODBMTFBQTExwbGxscHx8fHx8fHx8fHwEHBwcnDA0YEBAY
GhURFRofHx8fHx8fHx8fHx8fHx8fHx8fHx8fHx8fHx8fHx8fHx8fHx8fHx8f
Hx8fHx8fHx8f/8AAEQgA2QCmAwERAAIRAQMRAf/EALIAAQACAwEBAQAAAAAA
AAAAAAAFBgMEBwIIAQEBAQEBAQEBAQAAAAAAAAAAAECAwQFBhAAAQMDAgIE
CAsGAwcfAQAAAQIDBAARBRIGIQcxQSITUXEy0hSUVhdhoSOTtNQVdRY2CIGR
QlJygrFiJJKywjNDg8PRonOzNEQRAQABAgIGCAMIAgMAAAAAAAABEQISAYEx
klMEFEFRYbFyotlZgRMF8KHB0SlyQmJxkVIVBv/aAAwDAQACEQMRAD8A+qaB
QKBQKBQKBQKBQKBQKBQKBQKBQKBQKBQKDnOzJHJPESpSMBhsZIXCCDIQvHNNKAc
vpIC203HZPR0VMMPrcVHGZFMd18V/tP5tjZ0fk/vCC/OwOCxsiNHd7h1S8c0
0QvSFWsttJ6FCmGGeKv4vlu2++6Jn+0/m0N6ZXkZs6Q1FzWHxaJrye8REZx
zLrmgm2pQS3Zl4cLnj1Uww6cJZxufFbLr6deKfzb2zWuTW8YLSzA4XFSEMKC
JDSsey242pQukLSpsdNuBHCmGHPiruMyLqX33xX+0/msHu+2F7NYr1KP5IMM
PLz2fvL9qUbuTb/LDbuElZrJbcxqYMNIU8puBHWqyIBAskI48VCmGHbl4jic
2+LLb7qz/aXrAba5Z57DRMxj9uYxUKa2HWCuBHSopJtxTo4dFMMJncTxOXfN
l199Y/tKQ932wvZrFepR/Mphhy57P3l+1J7vthezWK9Sj+ZTDBz2fvL9qT3f
bC9msV6IH8ymGDns/eX7Unu+2F7NYr1KP5IMMHPZ+8v2pPd9sL2axXqUfzKY
YOez95ftSe77YXs1ivUo/mUwwc9n7y/ak932wvZrFepR/Mphg57P3l+1J7vt
hezWK9Sj+ZTDBz2fvL9qT3fbC9msV6IH8ymGDns/eX7Unu+2F7NYr1KP5IMM
HPZ+8v2pPd9sL2axXqUfzKYYOez95ftSe77YXs1ivUo/mUwwc9n7y/ak932w
vZrFepR/Mphg57P3l+1J7vthezWK9Sj+ZTDBz2fvL9qXzByrmSdmSMDvgrlw
uQmv4XMXPZbQQ2tKzb+rX/ZVfsPqNscRF+T/ADtti637fd8XUP0o/k3L/eJ/
+huj4/8A6P3bfD+Mo3mvtbdTfM87o2mIWanIjoEvDOFh+Q1obCCTFcOooUgp
IKO1c+DpOv03iMqEG+Xm4rlrou0xE/FCYPm1DwO1NzT8ftuPgd6B9iPLS2l1
LCIOqcusx3FHU1NFK+x0XI8VHozvps5mbZbdfN+VSZjr6Onpr1tzbXNreTOR
MVzPOZ5idi5T633MelvoU5qM48gIJQEuo1NgXPZP8o6zHEfTcqba4MEXfEfu
ritrEfBr43e3OTL8ucpu2bllqsRABcYSlyPHLrz5eaAcLXcKaUIGvTxt4j00
azOE4SziLcqlZxXds6lpPTWrLk+cG50QNoYNnKpwonY5M3LZhmEh9wFanQID
cZtISB8l/AkdPSADRMv6Xl4sy+bcdLqRbip1dPxZlc3eZjnLjJO48rm5PGZJ
MZecTE0qVAcQ4pD5YUjSkITdiop4Ai/a40Z/6zh44i2LtFt1tcNf5aNFVz5l
7xzOelZJiXueNuGE22hyMFsGJkWVEgK71kIDZRC21JdX1ceNqPn/AFbhbMuL
ZiybJ/zW2fjr+6HWApiFAoFAoFAoFAoFAoFAoFBzHF8j4UXIIP2NKyXpYlvqIM5

Page 68 of 205 ---- Generated from The PARIS Forums

Mz+mej7drv8APyJyJtm2Zz6/vrop1UfQ1beQoFAoFB+LQlaFIULpUCCPgNBw
+XyY3fNM5j7QegwV5ASYLDMx1xKi7fZS1dZ12Wkdu6umtRLNOI2HHYHEQYTE
Vil2IDDaWkavIVBKegFxxUtXjUalWnEOcMDaWz9/wN3TFhlp6Op5GMjt2L09
jS226dA49hSQL8Bpv0mt2amLtah57bTuVkbTzuNSqL6bMfXlsc2pTLZ7kl5Y
cSjSCbCxrVImaM3TMRV1TI1hWsftLUWi27kVvS3kqUpfB1xXdpBWSdKW9ISP
BUnWts1iJbW5MJHyG3J8RMxMSS/FWwytfkkqlsFWBNiRaosuY7ExW9vxGxlz
eVbx0aG2Ftd6nvlvuC6U8QIJAKSdQK6kJSVx2Jy/yE7dL+Wj5kM4192ROage
jNlxLq1+SqRwWtGpRNUFK0aonJvJbJ/iHJZZWYb+y5y1SX4rUFpycFFACm2J
LhXYHTwug1mqzDpe2MMMLgYWM7xbpjNhK3HFFaionUq6iBfifBUVJ0CgEAgg
i4PAg0Ggzg8awyGWW1tNJSElbQ66IKUpFkpSArsgDwUGeNj4cZ115lvS8/pD
zhKIKUECyQVKJNkjooNigUCggOX35C2191Qvo6Klup6uO9/M8d3en6rylAoF
ByfnFt4ZjMwFB5cdyLHJbe06klxZ42UFJPK9YrrlzoYuiqs4nameiJcyWbz
b2TbixnEQYq0NNNoa7xISVnukl1K0DTxrUa2b/wBsprCZaJidjQHstkwmlqia
0OSFts3FiUpSDovpBA8NZu1rbqhzndPMvbq/RgzmyyloBJLboVoWDxBCb+Cs
rKpuc2sLLdPzfQ3oULkhSgog+UnSFdfGIYJq61y6598qIWNjx5eY7qY2hYdT
6O+RxXcdollqSsRLs21914HdOKGVwcoS4KlraDoSpHbQbKBSsJV8VZaS1AoF
AoFAoFAoFBACvvYFtr7qhFR0VLdT1cd7+Z47u9P1XIKBQCQOk2oKPvZCHMs3
psT3KAojwa12rdupJV3NT4gw8plDqVPKbKUNpuVFRtw4Vq2dLnmTWJhyLmvh
39zbl2vDxJRlyEFvRLZbfZCm7tlSpKwpQ/iTWbtMtW3Q5S3yh3gwbtw1pURZ
VnEk2I420g1lccNmFyn3ggyXw20llsR1eHoJ01KwuJL43klmXkl1/P4yFpN
u7fJ1j9+mtRHak5lHe+S8/A8vdsycRlNyQ8guRKVKb7kLGjUhKCnh3l76L8K
kkXwuj/OrZDV/wDWaz1aG3Vf8lqGOEXK/UJs5nyEyXj1BDB/4lJqszmwi5X6
k8C2dKMbMN+hRDSB8alUZ+dDVP6iFuh0wsepZaTrWHFIUEpHC6u7sQPhqL8x
q4n9TEiZkocFWLjOGU+2yXGn18A4sJvp0r6L+GhGZPUu28907+CmDtVeHilb
uZgzHfnVxsAITJSEJ+FV61R0qk+VvMRjfO3nch6OlK2FlchT46VhxAeatdT
x5SFX4Hx+OpMELjUUoIdI9+QttfdUL6OipbqerjvzfPHd3p+q8pQfO3O/wDU
huPZnMFja+AiwnGmmmmzOfmhavln+KQNCK6UoQUk9N7/BSEumkLztPeMTdMVy
Y3/+1oA5FhtanG2XF3JSIZCQtF+gjqrqc7L8dsTqTDoaeaP/UbWCCT0W8YN
Gql2c9t9KiZsilniSpL7rXEkdYWerqqFHJeZW7cQxmW28dMhvREMIspuZHDa
F3UDp7SiFAW6qzKUc7I74aUolzJs6R0JEI54/wDsbt8dGcMrJtfb+5t2437T
wsQNY1Z0NvS1Xlik8FKRctHRfo4VmbW6aE6xyc3w6r5SZEZJ6Nluf/JVwsYG
fHcrJcxTqW9yMSHYjgYkNxEdJdcQ7ewbKUoSQq/C1WhhS2S5UxG+6D2TfU672
e6ccLni2kuOX0XJOIntiTerhWYV6dy+5fYZ8rz0zKvRHtTq8khtRhX20qUEN
uuJDqk+Se0Rx+CICKLZieT3LSdHYksRXpER1IW2684sBaCLpKbBBsatFwoPm
nsPljt7BpfddXiFibalMrHrC3kvPEf8AOZcVdaUpQbFKk2JqTBEQkOSewNs8
wdr5LLbpivSMjGnSsUy+iQ80BHZQgNq0tLda3E6z8oU3NuupLURCs8xf0tbz
x0BiDsWQvLxZRP2o5MfQ1KKtWpAJVobLSQB0G+rrq10FNLq/6ceWu4+X+18j
ic8y2mZlilUmQy8HULSppKNATZJTOKOvpvWWnWqBQQHL78hba+6oX0dFS3U9
XHe/meO7vT9V5Sg+H/1XYI3Fc3ZGTVHQ61kl7L6C4FFJu13AvxAulTKilfto
LXyEyfNMbJZENiNKwD0iQW1OvBp86lpDyeKT2SUcDW4YmJXyRyw2pLjOSspj
/wDVLQVuRmpD3ozagnobQISEdXE6eJ41ZWhB5U8uUMajt6ltRUe2tKIKsDa1
yonqqUNagc4+U7czbkSVs7Fx40jHS5Akw4iEMqeZPG4ltqLfd8Ek9B4VJhXD
sG9teNkmUbjjTGmG3Qt1KBqUdJ6CFFBt4asU6WdM9L6p25zf5Yt4iLHYnqip
S2nuozkZ5CtNrJ0hKVJtw8NakrELijY7deKlxI3pUeM6IKTKeaWwCkm/wAn
qKFHgKiRNUdsDEQ8ZEyxS6G33Mo+oOpCQsJYOho2TcG4ur+41ErCYy7OJlss
Ca86TGdD7LjZKF96BYL1JA4/FQxQ9HJ4lcJcdSe8jBbU04OytJFtJvfeih
i7GrtqZg4MBqDNieiRWg6UsQwppnUtRVdIToKeJPAUaiW9vriFhtycvMpAwT
EdjNZRph+PkXytZU606mQnU6rvFpSspINvD0VmZWlaPLrD8xdh4FOGTtWPkG

3H3ZT8mJkWkkuPruRoebasEiw4XvaixV10EkAkWPWKilAoFBACvvYFtr7qhf
R0VLdT1cd7+Z47u9P1XIQ+69tI3Hi/s5eRnYtPeJcMnGPmM+dII0FYCuydXE
WoKJL/TVyunMpRk2chkngoFcqXkJLjqhcnQSV2Cbm/ZAoM0vHY3bmOXj8PGb
iQYALMOMng2hOuWuT47kn9tcuKz5ysq6+Im6bY1RpWyKzRIW42pguDi2Uav7
SL16ImJisMsDCgWezxGpVvEVE1ZZsnQqG/5WSj7QIKxjoYnuT/RmHVJKwUI
ZZKtI8AcuKLLiea2y3jMSw3mZaH8y5FEgRiIS+9Gk2uSmySSLXURW6x0udGL
bGKk7ayTbrjSnFBCE9dLlms6/gaKkEAHh2TVi2hV1eRv6Hk9tT0T87jO8iNK
dahtpdhSi62m6Ud2+5q/i4hKa5y1M6FQ2NuKYp14tzVFjUkhJcJbsGnlW6eA
ukVlysJrT2694zMLttrOelxXUurU0ywgJUts0JCIImwKjYXAtfplqujk8Hn1u
j7VQMmltWOUsJfQ2FJdSkmxKVg9I6bWrNW4h3dO0dxTFGNBeiyHFqQFoMkB9
tghKvSO7cAUU6VXsnia0y+gMVFMtFw4pUFIhhtorT5J0ICbi/UbVh0bVAoFA
oFBACvvYFtr7qhfR0VLdT1cd7+Z47u9P1XIKBQUfMEDKSbfnzh463CI2SbsO
cOGhX+FUYyOWGRrFIdKuriRSUt1NdqHGkMPofbS6IEpbiUqANlp4pUL9Y6qK
gt1cvNuZ5UR2ehSX4rfchxtZQS0O0Em3Dsr7QqpMKg9y8lQZSnMTIXGwggob
kAPJ4f7JrUMYUG5tmHBz+R3DvCCzkY5Q0G0MhViEp0rCbEFKyUpIPVWZida9
CyLZ5YyIOPyOXjJx6VD0Ragg91JkON2KVOtpKNaSLiy+J4UrKxEoXc8ce9DI
4qHDhLx0XGJcYK27Ljd2taVJdSoAKCiq+vVx6KI/WW61U9AROZENlrYVko6a
iQm47YDelCCIClosIKrjtfQLVmlZxaXV0cs1s7fh74O43ZL+MnQzuLGABSo
8cuJ0kOIOq/dIK7EEEGrMUIYpGmX1Lldo4fLQw0HZcRBstp3Hy5EQjhwKe5W
IJHHoII+Cs1bolYMQQ4TEUOuPhhCWw8+ouOr0i2paz5Sj1morPQKBQKCA5ff
kLbX3VC+joqW6nq4738zx3d6fqvKUCg5q5i0wcxmHg646qfNXJX3nHSdCWgl
P+UJaHTW4R7cYdcjuaEKUCIQBANujw1RXc8+s5ODCRIEVwJ9ICIEgr7vSktg
dHEHpN7dXhqufTRIQMjCEfIPqeQllh51bjhIACQgKJPiqOiFm8wMS7Ebew7T
2WW6IDgbZSUaUueTrKwCIR/lterEMTf1aWSUsOyVlabOtaRbUUIKVcergePX
xqtlPeEGMxKhpy0czMFNQly4yS4IPpS76FrLZ1JANjq6qzplm5c8LGcf2nBj
5FkR3vRkplNBQUW1hNidZv2kEXCvDxo10MU/buOmxnO9Sh5lxhSVKWO8CkqT
5RPHV4asSTD5g3Ng8Fjcr3qY0f7Miu909HjqUmRpaV3altLN1a1q6BbTwrMx
Slnrqkx+qkdERX/M6af6p/t3XJws6OTeenYQwnNv5KEkyEyUOx5wDcZpDToc
CnUuLKUpQUKSmx46uNSdbUO6bXbDe2cQ2L2RCjpGrp4NJHGsrKToFAoFAoId
I9+QttfdUL6OipbqerjvfzPHd3p+q8pQKCizUbHuvNd/paWApXck2GhCRa6r
9lQueAVwrUJL583LzwwLG6pmLynp8fGw5JZZygJcbWlry1Lba7aCpadKeBGn
jw6K3qTWxZjimpsxyBhZKkOZLGZeQuLgkOtHUI1paUqcUVkLSkFdrjiamJmbW
7lg4/MbHy2Qi492BkmkykNR/SFuFtccqTq4IKLqDeoXHgrWKWZsiYa20NzbS
mwW/s5pTbiQ09/JUg3sCtVlquo8FLUq/Hp6atGopEUWXbmTx+QfvJgyWpJAC
Sp2M6l7gL31JQrULfCKktLUuOzIZQFjU3wUkHhwtw6ayqOnbgx+GymNxxkzU0
jMuqYgPW+S79KdQYWr+FSx5Hh6OqoJwrShKlqISICSoqPAAAXv8AsoOT7n5Y
8tpW5DlpORXFmLdEp2I04FtqlWFOANo7YLh6r+EgUw1Z0RNXQs7Jxs3ITmdt
4iz+ZXFWhjDpcbRKstQU3pQsp8luygPgt00mFbGE51YaDDxcHd2Jye1p7zbL
I9MhPeiKdICQuQ2ISOJ6lWt11lp0+oFAoFAoIdI9+QttfdUL6Oipbqerjvf
zPHd3p+q8pQKCub6gx9vzVyHvR2VMKZfeA1WQ4QL6euxNasnSzdD54zHK56
d8oibBfjDgqV32qyP6QnvCf8tq7TMMUakvIbKyMnFRliHleEwjakRFIKe8dc
cVrcdU2RoTrX0DVcCs6FpK9RMM43Cm412QtByHfF9xtluDICgSL9kadfCixD
5n3rsff+zpD+G71+ZipykhqRF1KQ+hF9CVpTdSDx4oPX4emsaWtCG2tiMxH3
fi4jyXYMhcplTtd1MugKWB1WUOmlsaUmXc+YG7uY+x9z46JjZbk+FKZSpcNx
oSj2LBZTw7wHxKrcsxoXTB83dpZRpDU+QmJOR2izJaLLjbgB7SUr1gEdRph6
IxLJBzWEz8QtRnmchHRZLhJDhuBwK02TYnxUw0WtWc4WOpvuYyvQW/5omlpX
7bCISjUZ2NFbkqkuKmt4m/eOcVH+q5INJvSLU/FkZKcAGnnmkDoQo627Dq0r
1JH7LViWkzF3hMRYSGUPp61tnu1fuOoH94qUEvE3RipC0NqUqO6shKUOptcn

gAFC6fjqUWqWqBQKCA5ffkLbX3VC+joqW6nq4738zx3d6fqvKUCgh934jlZf
b0uBjpKIs11PyLjqdTRUDfS4B2tKusp4jpFWJHFfxTE2rlkYLecMYTKrQHI8
orEiG82oOpt9I1J7SSLOJHw1utWdSztJTLQHoq7sO8UPlcUpKgetJSSk/so
rL9jsEhS3HFm9xqWo2+OIRFSdqpeUrW6Sk9BUST0+OtrcmFEyuVuDIOoecBT
JQQUSEHStNj1GmNMLOjYbTueenZF5U1BaYTD9JJWEONEKKrHr1JCvHTEUbgU
223lck5AyUNt/HtMIW24ptCluOrKtdlqSpSQkAcAatRHYXlrExWSddwMqXi2
9ASoJWHUKWDFSW3QpJQAF30qYVzx8LLMi0ksyLdDjQLKj40KKk/uNZmVSTkh
hhvXJWI6QLITxCE/7RON46ggsnzK2jBbKTL9IWnpDKbov/8AlrSj46sWyzWG
ZqJvvLqCoGDaxbKv/wCnlOdqx6+6RY00QuIZNubDnQ56J+Zyy8i62CWYjaAz
GQs/xFKfLi6tXR09NZm7qWIXGsNFAoIdI9+QttfdUL6OipbqerjvzPHd3p+
q8pQKBQcm5+7Xg5SBjZshoLLS1x1LtxAWNaf9w11y5YuhwiJA3btZ5UjbGQc
ZaUdS4iu2yr+ptV0/FetzazVcNvc9GtYi7pgnHvjh6XHBWyT4VNm60/2IVYm
1qLnS8dlsdk4qZcCS1MjK6HmFhafEbeSfgPGstNo36vGKD9uDw6fg6aCNYG5
cLi7+I5BlhQ/6SIBSsz/Ym6/ixqEpMqzl+eO24CCiMyuQ6OhThDKP2JOpw/7N
awszco+X567mmG2PT6Mk9AZQEfw7rX+4CtRbDM3OpYDIJjt47EgZTI5We1m
ckwh8zA4h7ulnpShLqFjT8fwiuc3zGpuLYIR4P6Wd3xOYOGnT8s1m9vRZaJE
h5xS23gho6gksL1p7VrdlRrOJqj6grKIAoFAoIdI9+QttfdUL6Oipbqerjvf
zPHd3p+q8pQKBQQW9sC7nNvSILGn0klJGo2GpCr9Pwi4rVs0IJhwPLYfKYq
QY2QirjOnydY4H4UKHA+MGU8TVzog5+HgS0FLyEkn+KwqorSNu7pxWVYOz1P
Lyktzu48VhwNI3ShTik9ohJ7KDwVcVi6KLDdP6hdzYwO4vMQfR8xFOh1EmIt
t1Kh/OgLQm/9orETA1pQ+S5o7rzTSVNvv9y8CbFXcN9NjZpm3g/iUa3HYzMo
dDOVlf8AOelSrpQ32E/Fx/fWqSxVc8NyY3nMjsy4+GfUy+R3bhARcEXCu2Un
Sf5uilbYWky6Rt39NuTXoXmJjMJHAqaZHfOeK/ZQP3mszmx0NRlu54DDRsjh
oeJjKUtiE2Gm1uW1EJ6zYAVxmay6RFG/UUoFAoFAoIdI9+QttfdUL6Oipbqe
rjvzPHd3p+q8pQKBQKDXn46Dkl6o01hEhhXS24kKHj49BpEjmm5+SzTmqRt
9/u1cT6E+SUH4EOdl/uv46625nWxNqp7N2xmYfM3AszoL0cxFyZLhKSUBKYy
2wrWLpsVO2BvVvmKJEaXVN9cr9lb4idzn8eh19AsxPbs3Jb/AKXQL2/yqun4
K4xLcwoGJ/TJhIncsSMo47DjgpbS20IDqklaldpZKk3sq3BNdfmaGPlr9gOV
WxMGUryYtt19PRIk/Lr/AGa7pT/aBWZvmWotiFsAAAAFgOgVhooFAoFAoFAo
FBRdjbgzyWytvtN7ZyT6G8bDSI9tzGhCwGEALSHJiF2V0jUkHwgVi2dD6fGZ
Fk518/Mtj9V3/Pr8Kb/EuZ9IMr87i/r1ar2PNy9m8s8/oPxLmfZTK/O4v69S
vYcvZvLPP6D8S5n2UyvzuL+vUr2HL2byzz+g/EuZ9IMr87i/r1K9hy9m8s8/
oPxLmfZTK/O4v69SvYcvZvLPP6D8S5n2UyvzuL+vUr2HL2byzz+g/EuZ9IMr
87i/r1K9hy9m8s8/oPxLmfZTK/O4v69SvYcvZvLPP6D8S5n2UyvzuL+vUr2H
L2byzz+g/EuZ9IMr87i/r1K9hy9m8s8/oPxLmfZTK/O4v69SvYcvZvLPP6D8
S5n2UyvzuL+vUr2HL2byzz+g/EuZ9IMr87i/r1K9hy9m8s8/oPxLmfZTK/O4
v69SvYcvZvLPP6D8S5n2UyvzuL+vUr2HL2byzz+g/EuZ9IMr87i/r1K9hy9m
8s8/oPxLmfZTK/O4v69SvYcvZvLPP6D8S5n2UyvzuL+vUr2HL2byzz+g/EuZ
9IMr87i/r1K9hy9m8s8/oPxLmfZTK/O4v69SvYcvZvLPP6DI9+QttfdUL6Oi
luo4738zx3d6fqvK1cq641i5jrailxthxSFDpBCCQaCm4vceSdbIOCStao+E
9IstJA9ISCSuyg/LAOFB+RMxl0QcjqmOuqGKYmtrXYqQ643qUUKAcL9VBgbz
2VGKzS25j6vR40NxpTo0ulcdA7wpCglWk9XV4KCdmSJity4mHFIOguoMqc3q
ugNISakabfxqFjQReDzWXdnQVvS3HEzkzi60rToSWCdGgADTa1BpbM3Flspk
mlMqY6q7b/ekGxIskoVe3SLm1Bvs5bJp2WuWJbq8jLkejMOKNyIXfFA0i38o
NBkiZ3JPYrby1PqEh2cl008Lr0lQIV47Cgmt6zZULbMyTFcLL7fdaHE9I1Oo
SfiNBVcnuDNxMDi8oZSy5NILfcbSQUhm90tD4AkUGePnJ7uKwjroX4IITH25
DrdytTaSrSOyCeHioJUzVndjcJcuQlghsuMoTqupwr03cskkXHTqtQeOY2Uy
OOxcV2A+phxb+ISkdJToUbfFQRsDcuRm5/N6JCxEagOuxWx5KVoCAFJ4eG9B

I4fKZB7IYFt19S0Scep19J6FuC3aPw0HtmZnJMHcalbqlzI8pxuEDbspASdK
b8Oi9qDb2hJelRpD6pq5TSIpShl4WeYWINnG3OA46qCfoIDl9+QttfdUL6Oi
pbqerjvfzPHd3p+q8rDOjmTCkRgrSX21thXTbWkpv8dBARNnqaS4hySFIdxg
xqtKbEdN3Bc/D0UHiNs6SiJMaemIW5lhNwGllbKUpQ0nSFKBUq5P7KA9sx1y
NPZ9KSDNjRowOg9kxgAVdPHVag24uBnx869k0yGlpeQ0zoW2rUhpsDUIBCgO
0RfiKDVxmznoctlxctLjERMgRkBspX/qTc61aiDpv1AUGPAbHVicpHnelBzu
mCy4gII1KJPavc9VqDI3sw+gY+C/IS7HhylynkFHBy5UUp6eFtRoPCdmSGAk
RZTaENZE5CO2psIKQRbu+ChwFBLZ3EP5fAvY5byW3ngjU6lJKQULSs2STfjp
8NBEObGU5CagmUPRGH33GUaDdLT6SA30/wAJUTeg9RNnS4mPxcdmW2Xsa+4+
FrbJSvXfhYKB4X8NBIDD5FOdGVQ+ZqXHRHfbLaiDpVqUUEL4X6r3oP3cu3/t
pmM13oaSy73iwU6tQKCKjpH81BE4bYjmOLxMtLinoLsNRCCO064V6+nqHC1B
uM7ZnRnsY/Gktd7j4pinvG1FK727XZUm3RQPw3IEJyJceelhGSeW+HUIWHm1
KtbSoLH8vGg3MFhX8e9NkyH0vyJy0Lc0IKEDQnSLAQubnpPGglqCA932wvZr
FepR/MqYYerns/eX7Unu+2F7NYr1KP5IMMHPZ+8v2pPd9sL2axXqUfzKYYOe
z95ftSe77YXs1ivUo/mUwwc9n7y/ak932wvZrFepR/Mphg57P3l+1J7vthez
WK9Sj+ZTDBz2fvL9qT3fbC9msV6IH8ymGDns/eX7Unu+2F7NYr1KP5IMMHPZ
+8v2pPd9sL2axXqUfzKYYOez95ftSe77YXs1ivUo/mUwwc9n7y/ak932wvZr
FepR/Mphg57P3l+1J7vthezWK9Sj+ZTDBz2fvL9qT3fbC9msV6IH8ymGDns/
eX7Unu+2F7NYr1KP5IMMHPZ+8v2pPd9sL2axXqUfzKYYOez95ftSe77YXs1i
vUo/mUwwc9n7y/ak932wvZrFepR/Mphg57P3l+1J7vthezWK9Sj+ZTDBz2fv
L9qT3fbC9msV6IH8ymGDns/eX7Unu+2F7NYr1KP5IMMHPZ+8v2pPd9sL2axX
qUfzKYYOez95ftSn6rylAoFAoFAoFAoFAoFAoFAoFAoFAoFAoFAoP/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 involved 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)

>

>

>"Tom Bruhl" <arpeggio@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...

>> Basically this involved 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.....soooooo.....
>> 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 whilst 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)
>>
>>
>>
>
>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...
>
> "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...
> >> Basically this involved 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)
> >>

> >>
> >>
> >
> >
> "DJ" <notachance@net.net> wrote:
> 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.

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

James

>
> "James McCloskey" <excelsm@hotmail.com> wrote in message
> news:4540e6d7\$1@linux...
>>
>> "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...
>> >> Basically this involved 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)
>> >>
>> >>
>> >>
>> >
>> >
>>
>
>I'm sure I'll hear back from him soooooonnnnn.....

"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
> >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.
>
> But did you tell him that the DAW is called Paris, so it should work.
>
> James
>
> >
> >"James McCloskey" <excelsm@hotmail.com> wrote in message
> >news:4540e6d7\$1@linux...
> >>
> >> "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...
> >> >> Basically this involved 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

"Tom Bruhl" <arpeggio@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" <arpeggio@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

>Surprisingly,

> >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
> =3D
> =
>href=3D3D"news:453f8b69$1@linux">news:453f8b69$1@linux</A>...</DIV><BR>H=
i,<=3D
>BR>The=3D20
> SSLcompressor has 0 latency.<BR>Thee SSL channel has 1 =
sample=3D20
> latency.<BR>Regards,<BR>Dimitrios<BR><BR>"LaMont" <<A=3D20
> =
href=3D3D"mailto:jjdpro@ameritech.net">jjdpro@ameritech.net</A>>=3D20
> wrote:<BR>><BR>>yes it works. Thank god Waves still code their =
=3D
>plugins=3D20
> in Direct-x. Surprisely,<BR>>the plug added no latency. However, =
=3D
>the mix l=3D20
> was working on only had 10<BR>>tracks. <BR>><BR>>"Goran =3D
>Stojiljkovic"=3D20
> <<A=3D20
> =3D
> =
>href=3D3D"mailto:goran.stojiljkovic@os.t-com.hr">goran.stojiljkovic@os.t=
-co=3D
>m.hr</A>>=3D20
> wrote:<BR>>>does it work?<BR>>>latency=3D20
> ?<BR>>><BR>>>please answer.....=3D20
> <BR>>><BR>>><BR>><BR></BLOCKQUOTE>

```

```
><DIV><FONT size=3D3D2><BR><BR>I choose Polesoft Lockspam to fight =
spam, =3D
>and=3D20
>you?<BR><A=3D20
=
>href=3D3D"http://www.polesoft.com/refer.html">http://www.polesoft.com/re=
fer=3D
>.html</A> </FONT></DIV></BODY></HTML>
>
>
```

-----=_NextPart_000_0035_01C6F90A.D2CBD240

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 6.00.2900.2873" name=3DGENERATOR>
```

```
<STYLE></STYLE>
```

```
</HEAD>
```

```
<BODY bgColor=3D#ffffff>
```

```
<DIV><FONT size=3D2>Speaking of SSL, have you guys seen this =
new&nbsp;piece , l=20
```

```
wonder how this would work with Paris latency speeking..??</FONT></DIV>
```

```
<DIV><FONT size=3D2><A=20
```

```
href=3D"http://www.solid-state-logic.com/music/duende_home.html">http://w=
ww.solid-state-logic.com/music/duende_home.html</A></FONT> </DIV>
```

```
<DIV><FONT size=3D2></FONT>&nbsp;</DIV>
```

```
<DIV><FONT size=3D2>Rob</FONT></DIV>
```

```
<DIV><FONT size=3D2></FONT>&nbsp;</DIV>
```

```
<DIV><FONT size=3D2></FONT>&nbsp;</DIV>
```

```
<BLOCKQUOTE dir=3Dltr=20
```

```
style=3D"PADDING-RIGHT: 0px; PADDING-LEFT: 5px; MARGIN-LEFT: 5px; =
BORDER-LEFT: #000000 2px solid; MARGIN-RIGHT: 0px">
```

```
<DIV>"Tom Bruhl" &lt;<A=20
```

```
href=3D"mailto:arpeggio@comcast.net">arpeggio@comcast.net</A>&gt; wrote =
in message=20
```

```
<A href=3D"news:453fe0a4@linux">news:453fe0a4@linux</A>...</DIV>
```

```
<DIV><FONT face=3DArial size=3D2>oh.</FONT></DIV>
```

```
<BLOCKQUOTE=20
```

```
style=3D"PADDING-RIGHT: 0px; PADDING-LEFT: 5px; MARGIN-LEFT: 5px; =
BORDER-LEFT: #000000 2px solid; MARGIN-RIGHT: 0px">
```

```
<DIV>"Rod Lincoln" &lt;<A=20
```

```
=
```

```
href=3D"mailto:rlincoln@nospam.kc.rr.com">rlincoln@nospam.kc.rr.com</A>&g=
```

t;=20
wrote in message <A=20
=
href=3D"news:453fd252\$1@linux">news:453fd252\$1@linux...</DIV>
If =
it's a=20
stool sample.....well.....never mind.
Rod
"Tom =
Bruhl"=20
<A =
href=3D"mailto:arpeggio@comcast.net">arpeggio@comcast.net>=20
wrote:
>
>
>What's a sample among=20
friends?
>T.
>
> "Dimitrios" <<A=20
href=3D"mailto:musurgio@otenet.gr">musurgio@otenet.gr> wrote =
in message=20
=3D
>news:453f8b69\$1@linux...
>
> =
Hi,
> =20
The SSLcompressor has 0 latency.
> Thee SSL channel has =
1 sample=20
latency.
> Regards,
> =20
Dimitrios
>
> "LaMont" <<A=20
href=3D"mailto:jjdpro@ameritech.net">jjdpro@ameritech.net>=20
wrote:
> >
> >yes it works. Thank god =
Waves=20
still code their plugins in Direct-x. =
=3D
>Surprisely,
> =20
>the plug added no latency. However, the mix I was working on =
only=20
had
=3D
>10
> >tracks.=3D20
> =20
>
> >"Goran Stojiljkovic" <<A=20
=
href=3D"mailto:goran.stojiljkovic@os.t-com.hr">goran.stojiljkovic@os.t-co=
m.hr>=20
wrote:
> >>does it work?
> =
>>latency=20
?
> >>
> >>please=20
answer.....=3D20
> >>
> =
>>
> =20
>
>
>
>
>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?</DIV>
><DIV><FONT=20
face=3D3DArial =
size=3D3D2>T.</DIV> ;
><DIV>=20
=
</DIV>
><BLOCKQUOTE=3D20
>style=3D3D"PADDING-RIGHT: =
0px;=20
PADDING-LEFT: 5px; MARGIN-LEFT: 5px; =3D
>BORDER-LEFT: #000000 =
2px=20
solid; MARGIN-RIGHT: 0px">
> <DIV>"Dimitrios"=20
<<A=3D20
> href=3D3D"<A=20
=
href=3D'mailto:musurgio@otenet.gr">musurgio@otenet.grmailto:musurgio=
@otenet.gr">musurgio@otenet.gr</A>=20
wrote in =3D
>message=3D20
> =
<A=3D20
> =20
=3D
>href=3D3D"<A=20
=
href=3D'news:453f8b69\$1@linux">news:453f8b69\$1@linux...</DIV>
Hi'>=
news:453f8b69\$1@linux">news:453f8b69\$1@linux...</DIV>&=
lt;BR>Hi,<=3D
>BR>The=3D20
> =20
SSLcompressor has 0 latency.
Thee SSL channel has 1=20
sample=3D20
> =20
=
latency.
Regards,
Dimitrio s

 "LaMont"=
=20
<<A=3D20
> href=3D3D"<A=20
=
href=3D'mailto:jjdpro@ameritech.net">jjdpro@ameritech.net>=3D20'>mail=
to:jjdpro@ameritech.net">jjdpro@ameritech.net>=3D20<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 l=3D20
> was working on only had =
10
>tracks.=20

>
>"Goran =
=3D
>Stojiljkovic"=3D20
> =20
<<A=3D20
> =3D
>href=3D3D"<A=20
=
href=3D'mailto:goran.stojiljkovic@os.t-com.hr">goran.stojiljkovic@os.t-co=
'>mailto:goran.stojiljkovic@os.t-com.hr">goran.stojiljkovic@os.t-co</A=

```

>=3D<BR>&gt;m.hr&lt;/A&gt;&gt;=3D20<BR>&gt;&nbsp;=20
  wrote:&lt;BR&gt;&gt;&gt;does it=20
  work?&lt;BR&gt;&gt;&gt;latency=3D20<BR>&gt;&nbsp;=20
  ?&lt;BR&gt;&gt;&gt;&lt;BR&gt;&gt; ;&gt;please =
answer.....=3D20<BR>&gt;&nbsp;=20
=
&lt;BR&gt;&gt;&gt;&lt;BR&gt;&gt; &gt;&lt;BR&gt;&gt;&lt;BR&gt;&lt; /BLOCKQUO=
TE&gt;<BR>&gt;&lt;DIV&gt;&lt;FONT=20
  size=3D3D2&gt;&lt;BR&gt;&lt;BR&gt;I choose Polesoft Lockspam to =
fight spam,=20
=
=3D<BR>&gt;and=3D20<BR>&gt;you?&lt;BR&gt;&lt;A=3D20 <BR>&gt;href=3D3D"<A=20
=
href=3D'http://www.polesoft.com/refer.html">http://www.polesoft.com/refer=
'>http://www.polesoft.com/refer.html"&gt;http://www.polesoft.com/refer</A=
>=3D<BR>&gt;.html&lt;/A&gt;=20
=
&lt;/FONT&gt;&lt;/DIV&gt;&lt;/BODY&gt;&lt;/HTML&gt; <BR>&gt;<BR>&gt;<BR></=
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 soooooonnnnn.....

>

> "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

>> >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.

>>

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

>>

>> James

>>

>> >

>> >"James McCloskey" <excelsm@hotmail.com> wrote in message

>> >news:4540e6d7\$1@linux...

>> >>

>> >> "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...
>> >> >> Basically this involved 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 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...
> > I'm sure I'll hear back from him soooooonnnnn.....
> >
> > "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
> >> >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.
> >>
> >> But did you tell him that the DAW is called Paris, so it should work.
> >>
> >> James
> >>
> >> >
> >> >"James McCloskey" <excelsm@hotmail.com> wrote in message
> >> >news:4540e6d7\$1@linux...
> >> >>
> >> >> "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...
> >> >> >> Basically this involved 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.....soooooo.....
 > > > > 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)
 > > > >

> > > >
> > > >
> > > >
> > > >
> > > >
> > > >
> > >
> > >
> > >
> >
> >
> >
>
>

> 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
> news:4540f566\$1@linux...
> > 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 soooooonnnnn.....
> > >
> > > "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
> > > > > 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.

> > >>
> > >> But did you tell him that the DAW is called Paris, so it should work.
> > >>
> > >> James
> > >>
> > >> >
> > >> >"James McCloskey" <excelsm@hotmail.com> wrote in message
> > >> >news:4540e6d7\$1@linux...
> > >> >>
> > >> >> "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...
> > >> >> >> Basically this involved 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

> >
> >
>

>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>Scott,</DIV>

```

<DIV><FONT face=3DArial size=3D2>Yes the wrapper will fix the GUI of =
most=20
plugs.&nbsp;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" &lt;<A=20
    =
href=3D"mailto:volthause-nospam-@soldrocks-nospam-.com">volthause-nospam-=
@soldrocks-nospam-.com</A>&gt;=20
    wrote in message <A=20
      =
href=3D"news:Xns986875884AFEBvolthause@202.63.37.102">news:Xns986875884AF=
EBvolthause@202.63.37.102</A>...</DIV>So,=20
      if I've got VST effects working on my Paris system, should I bother =
<BR>trying=20
      to wrap them? I've noticed that things like meters on VST effects =
<BR>don't=20
      work, but the effect still seems to. Does the wrapper make the dancy=20
      <BR>lights and stuff work on VST effects?<BR><BR>What are the =
preferred=20
      options for general wrapping of VSTs?<BR><BR>Sorry for the dumb =
questions, I'm=20
      just trying to get my head around it.<BR><BR>-scott v.</BLOCKQUOTE>
<DIV><FONT size=3D2><BR><BR>I choose Polesoft Lockspam to fight spam, =
and=20
you?<BR><A=20
href=3D"http://www.polesoft.com/refer.html">http://www.polesoft.com/refer=
..html</A>&nbsp;&nbsp;&nbsp;&nbsp;</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 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...
> 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 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
>
>
>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

>

> 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...

> > 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

> >

> >

> >

>

>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 involved 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

"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 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
>
>
> 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,
> 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:
> >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.....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:
> >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
> >
> >
> >
> >
>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.
>
>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:
>> >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
>> >
>> >
>> >
>> >
>>
>

>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 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:

>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,

>> 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:
 >> >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
 >> >
 >> >
 >> >
 >> >
 >>
 >
 >"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.

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 am sure 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 show on its routing when selecting Scope asio these exact ins and outs.

That's 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 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

>

>

>consider them reserved

Neil wrote:

> 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

>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

> >

> >

> >

>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:
>> >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
>> >
>> >
>> >
>> >
>>
>
>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:
 > >> >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 if
 > >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 am sure 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 O2) 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
> , 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 am sure 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 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 don't 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 don't 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 tutorial 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

>

>

> "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

>>>>>
>>>>
>>>>
>>>>
>>>
>>>
>>
>>
>>
>

>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 tutorial 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

>>

>>

>> "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
 >>>>>>
 >>>>>
 >>>>>
 >>>>>
 >>>>>
 >>>>
 >>>>
 >>>
 >>>
 >>>
 >>
 >>
 >
 >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@kfocust.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.
 Genel'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
>
>
> "Don Nafe" <dnafe@magma.ca> wrote in message news:45414828@linux...
>> would need a tutorial 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
>>>
>>>
>>> "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
>>>>>>

[illegible]

>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;?)

 $\vee$

>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.

pablInquiring 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:

>> >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
>> >
>> >
>> >
>> >
>>
>
>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](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 :-)

> 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:
> >> >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
> >> >
> >> >

> >> >
> >>
> >
> >
>god bless you DJ. "DJ" <notachance@net.net> wrote in message
news:45417186\$1@linux...
>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 :-)
>> 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:
>> >> >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
>> >> >
>> >> >
>> >> >
>> >> >
>> >>
>> >
>> >
>>
>
>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
>> 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.
>
> 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 accomodate
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:
>> >> >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

>> >> >

>> >> >

>> >> >

>> >>

>> >

>> >

>>

>

>DJ,

Cubase 5 is so damn easy and light I just opened 64 asio ins and outs and
cpu goes to a 13 % on a 2 Ghz cpu Intel.

So you can have 64 !! vst chaining possibilities.

Cubase runs on Pulsar computer of course.

Your UAD1 set must be on another computer I guess or same ?

Note that Pulsar cards eat a lot of pci bandwidth.

So if you will use on card you can carry away with UAD1 cards.

Beater though use 3 cards and another pc for uad1

Regards,

Dimitrios

"DJ" <notachance@net.net> wrote:

>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 with 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](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.

>

>;0)

>

>

>"Mic Cross" <crzymnmchl@comcast.net> wrote in message
>news:45416b6e\$1@linux...
>>
>> Inquiring 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:
>> >> >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
>> >> >
>> >> >
>> >> >
>> >> >
>> >>
>> >
>> >
>>
>
>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:

> DJ,
> Cubase 5 is so damn easy and light I just opened 64 asio ins and outs and
> cpu goes to a 13 % on a 2 Ghz cpu Intel.
> So you can have 64 !! vst chaining possibilities.
> Cubase runs on Pulsar computer of course.
> Your UAD1 set must be on another computer I guess or same ?
> Note that Pulsar cards eat a lot of pci bandwidth.
> So if you will use on card you can carry away with UAD1 cards.
> Better though use 3 cards and another pc for uad1
> Regards,
> Dimitrios
>
> "DJ" <notachance@net.net> wrote:
>> 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 with 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 :-)
>>> 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:
>>>>>> 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
 >>>>>
 >>>>>
 >>>>>
 >>>>
 >>
 >Hehehehehhehee
 That was funny,,, thanks.....)
 I was reffering to the old Cubase VST 5.1 !!
 Regards,
 Dimitrios

John <no@no.com> wrote:
 >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:
 >> DJ,
 >> Cubase 5 is so damn easy and light I just opened 64 asio ins andouts and
 >> cpu goes to a 13 % on a 2 Ghz cpu Intel.
 >> So you can have 64 !! vst chaining possibilities.
 >> Cuabse runs on Pulsar computer of course.
 >> Your UAD1 set must be on another computer I guess or same ?

>> Note that Pulsar cards eat a lot of pci bandwidth.
>> So if you will use on card you can carry away with UAD1 cards.
>> Beeter though use 3 cards and another pc for uad1
>> Regards,
>> Dimitrios
>>
>> "DJ" <notachance@net.net> wrote:
>>> 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](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 :-)
>>>> 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:

>>>>>>> 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
>>>>>>>
>>>>>>>
>>>>>>>
>>>>>
>>>
>>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 among Paris users...
Can you imagine all Parisians have the ability to send their Cubase (sorry DJ) tracks via asio destination channels to Paris for mixing ?
So one could install a basic simple audio card with low latency like rme or other like pULSAR which can have up to 64 asio destinations, well 16 would be great 8 would be enough.
Thus you can open a vst effect on Paris audio track 1 choose asio destination 1 and then on Pulsar environment asio 1 will receive the output of that channel.
So if a Paris user installs a Pulsar II card (3ms) then a paris audio track can go out to Pulsar have a great digital effect and come back on vst again with asio to complete the route.

That could be done with rme/Cubase on same computer with Paris or other combination.
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
asio card will use. I guess that this may be obvious but anyway I would
like to point.
Other daws that luck asio could benefit also.
How much would you define as a payment for the time and skills that would
involve such a task ?
Regards and thanks !
Dimitrios

"chuck duffy" <c@c.com> wrote:

>
>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
>
>
>
>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,
>
> 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
>
>
>
>"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" <arpeggio@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 am sure 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 O2) and then an adat out card to paris or pulsar.

>What do you think ??

>Regards,

>Dimitrios 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.

Chuck

"Dimitrios" <musurgio@otenet.gr> wrote:

>

>Chuck,

>You have hit gold here.

>Well at least amoung Paris users...

>Can you imagine all Parisians have the ability to send their Cubase (sorry

>DJ) tracks via asio destination channels to Paris for mixing ?

>So one could instal a basic simple audio card with low latency like rme

or

>other like pULSAR which can have up to 64 asio destinations, well 16 would
>be great 8 would be enough.

>Thus you can open a vst effect on Paris audio track 1 choose asio destination
>1 and then on Pulsar environnement asio 1 will receive the output of that
>channel.

>So if a Paris user installs a Pulsar II card (3ms) then a paris audio track
>can go out to Pulsar have a great digital eeffect and come back on vst again
>with asio to complete the route.

>That could be done with rme/Cubase on same computer with Paris or other
combination.

>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
>asio card will use. I guess that this may be obvious but anyway I wouldd
>like to point.

>Other daws that luck asio could benefit also.

>How much would you define as a payment for the time and skills that would
>involve such a task ?

>Regards and thanks !

>Dimitrios

>

>

>"chuck duffy" <c@c.com> wrote:

>>

>>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

>>

>>
>>
>>

>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 (probably 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,

>

> 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

>

>

>

>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:

>>

>> 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

>>

>>

>>
>>
>
>"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:
>
> 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:
>>>
>>> 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
>>>
>>>
>>>
>>>
>>
>>
>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
Heartbreakers 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.viewprofile&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>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 #1I 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

>

>

>

>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,

>
> 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](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 #1I 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
> >
> >
> >
> >
> >
>
> 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...
> > 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 #1I 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

> > >

> > >

> > >

> > >

> >

> >

>

>"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 (probably 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.parisfaq.com> 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...
> > > 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 #1I 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.
> > > >

>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.viewprofile&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>
>
>
>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

"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](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 #1I 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
>>
>>
>>
>>
>
>No because there are tons of plugs that just DONT run in paris, let alone
vstis!

Chuck

"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...
>> > > 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](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 #1I 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
>> > > >
>> > > >
>> > > >

>> > > >
>> > >
>> > >
>> >
>> >
>>
>>
>
>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?

"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.
>
> 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
> "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 #1I 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
> >>
> >>
> >>
> >>
> >
> >
>DOH!!!!.....OK, the difference being that with this plug we ould compensate
latency?

"DJ" <notachance@net.net> wrote in message news:454246bc\$1@linux...
> Ahhh.....OK.....so Paris would open this as a VST plugin? Can't we
already
> do that with chainer, etc?
>
>
>
> "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.
> >

> > 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
 > > "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](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 #1I 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
> > >>
> > >>
> > >>
> > >>
> > >>

> > >
> > >
> >
>

> That's like asking my wife if I finish painting the kitchen 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 way 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...

>> > > 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](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 #1I 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
>> > > >
>> > > >
>> > > >
>> > > >
>> > >
>> > >
>> >
>> >
>>
>>
>
>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

"DJ" <notachance@net.net> wrote:

>DOH!!!!.....OK, the difference being that with this plug we ould compensate
>latency?

>

>

>"DJ" <notachance@net.net> wrote in message news:454246bc\$1@linux...

>> Ahhh.....OK.....so Paris would open this as a VST plugin? Can't we

>already

>> do that with chainer, etc?

>>

>>

>>

>> "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.

>> >

>> > 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

>> > "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](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 #1I 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
>> > >>
>> > >>
>> > >>
>> > >>
>> > >
>> > >
>> >
>>
>>
>
>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

"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

>

>

>

>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
>Heartbreakers 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.viewprofile&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 environment (which can accommodate 16adats, 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 between 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) .

>

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

>

>"DJ" <notachance@net.net> wrote:

>>DOH!!!!.....OK, the difference being that with this plug we ould compensate

>>latency?

>>

>>

>>"DJ" <notachance@net.net> wrote in message news:454246bc\$1@linux...

>>> Ahhh.....OK.....so Paris would open this as a VST plugin? Can't we

>>already

>>> do that with chainer, etc?

>>>

>>>

>>>

>>> "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.
>>> >
>>> > 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
>>> > "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](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 #1I 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
>>> > >>
>>> > >>
>>> > >>
>>> > >>
>>> > >>
>>> > >
>>> > >
>>> >
>>>
>>>
>>
>>
>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.
>
> any ideas?
>
>
> jefwhose crazy idea was that? ricks ? jeez

jef knight wrote:

> hmmm...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.
>>
>> 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.
>>
>> 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?
>>>
>>>
>>> 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.com Glad 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.viewprofile&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@kfocust.com Images 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.

```

>>
>> 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:

```

>>>
>>> 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
>>>
>>>
>>>
>>>
>>
>>
>would work over a network/firewire??
AA

"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.
>
> 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
> "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](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 #1I 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
>>>
>>>
>>>
>>>
>>
>>
>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.lendaneer-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&highlight=clipping>To 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 sound 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...](mailto: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 environment (which can accommodate 16adac 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 between 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) .
> >
> >3. I thought the other hosts didn't have enough asio channels
> >
> >"DJ" <notachance@net.net> wrote:
> >>DOH!!!!.....OK, the difference being that with this plug we ould
compensate
> >>latency?
> >>
> >>
> >>"DJ" <notachance@net.net> wrote in message news:454246bc\$1@linux...
> >>> Ahhh.....OK.....so Paris would open this as a VST plugin? Can't we
> >>>already
> >>> do that with chainer, etc?
> >>>
> >>>
> >>>
> >>> "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.
 > >>> >
 > >>> > 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
 > >>> > "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](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 #1I 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

>
> 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

>
> "DJ" <notachance@net.net> wrote:
> >DOH!!!!.....OK, the difference being that with this plug we could
compensate
> >latency?
> >
> >
> >"DJ" <notachance@net.net> wrote in message news:454246bc\$1@linux...
> >> Ahhh.....OK.....so Paris would open this as a VST plugin? Can't we
> >already
> >> do that with chainer, etc?
> >>
> >>
> >>
> >> "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.
 > >> >
 > >> > 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
 > >> > "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](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 #1I 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.

> 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,

Mikels 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.com I 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

>> 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.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 (probably 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=47942490
>
> 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 level is consistently 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.lendaneer-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 don't 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 it 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 don't 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
> therefore start introducing unwanted artifacts into the mix.
> If the "bits" aren't there on the original recording, and the level is
> consequently 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.lendaneer-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 don't 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.
>>>
>
> "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.com> Thanks for that well thought out "rant", Neil.
That's pretty much the way I feel with Nuendo, although I don't put an
arbitrary 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.lendaneer-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 threshold, 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 on 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
