Subject: Oh ..... HELL YES!!!!!!!!!!!!!
Posted by animix on Sun, 01 Oct 2006 23:54:02 GMT
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Mixer: External insert in EDS inserts, Sampleslide on
>
>>>>> >> >> > native
>>>>> >> >> > insert set to 48 samples to achieve sample accuracy with
>>>>> >> >> > buffer
>>>>> >> >> > latency.
>>>>> >> >> > (this will ned to be done for every track in the mix)
>>>>> >> >> >
>>>>> >> >> snip
>>>>> >> >
>>>>> >> >
>>>>> >>
>>>>> >>
>>>>> >
>>>>> >
>>>>>
>>>>>
>>>>
>>>>
>>>
>>>
>>
>>
>Hey Deej! I'm just throwing stuff up against the wall to see if anything
sticks. Did you ever try the Steinberg distributed processing thing? It's
were you can hook up multiple ma

Subject: Re: Oh ..... HELL NO!!!!!!!!!!!!
Posted by animix on Mon, 02 Oct 2006 00:13:15 GMT
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chines with light pipe, for more processing
power.

What about using forte in the same machine? Did you try the Solid Sound
Studio's Aux Bus Set software for forte?

Here it is again:
http://www.solidsoundstudio.net/index.html

Mono with pan???
http://www.solidsoundstudio.net/html/basic_strip.html

Forte extras page:
http://www.brainspawn.com/products/forte/?op=extras

Thinking out side the brain;)

James

"DJ" <notachance@net.net> wrote:
>
>"DJ" <notachance@net.net> wrote in message news:45318cbc@linux...
>> I'm doing a 36 track mix in Cubase SX right now with 8 x stereo stems
>routed
>> to Paris submix 1. Very stable. Sounding better and better as I go. I
feel
>> sorta' like I just stepped in poo, but as I get the hang of it I'm
>thinking
>> I just might be able to achieve something *very nice*.
>> Neil/Martin/LaMont/Dedric..et al, are correct about the mix bus being
much
>> improved these days. It's just a different journey to get there.
>>
>>

Subject: Re: Oh ..... HELL YES!!!!!!!!!!!!!
Posted by Rob Arsenault on Mon, 02 Oct 2006 00:23:45 GMT
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I'd *really* like to get this thing happening with Forte but unless we
can
>> get enough folks together to justify it for the developer, We're getting
>> close to finishing some tracking on a project here and the mixes are
>coming
>> up pretty soon so it's getting to be close to time to take off the lab
>coat,
>> put Igor back in the dungeon and think in terms of getting these up and
>> ready to go.
>>
>> I've thought through using 3 x computers running Forte. I would need to
>get
>> another Multiface and another UAD-1 to get this happening the way I need
>to
>> and the thought of simultaneously digitally bussing multiple outboard
FX
>> processors through three computers is making me sorta queasy. Without
each
>> interface receiving a BNC feed directly off the house clock (which they
>> can't do), this is a trainwreck in the making and then I'm back to square

>1.
>> I would have to do this in order to accomplish the aux routing across
>> submixes that I would want to do and I'm already fried just thinking about
>> this one.
>>
>> (sigh)
>>
>Well, after getting a stem mix scenario going in this way, I discovered
that
>the bussing of my outboard FX to any track I want in the mix is compromised
>by stem mixing. I have to send the external processor to either one stem
bus
>or another. The way I've got this configured in original track-to-track
>Paris mix template, I can access any external processo

Subject: Re: Oh ..... HELL YES!!!!!!!!!!!!!
Posted by Don Nafe on Mon, 02 Oct 2006 01:05:53 GMT
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; hence the need for such a large I/O interface?
>>
>> Don
>>
>> "DJ" <notachance@net.net> wrote in message news:45314e37@linux...
>>> Dave,
>>>
>>> All tracks are being played back through Paris. The point of this is to
>>> have
>>> all tracks crossing the editor timeline simultaneously whether they are
>>> being processed by a plugin, or not. In order to achieve this, each
> track
>>> has to be sent to a Forte bus, processed by the UAD-1 Delaycomp (and
>>> whatever else) then returned to Paris. the "nudge" and then the
>>> Sampleslide
>>> instance in Paris is in order to cover the latency between the two
>>> computers
>>> caused by the 512k buffer settings of the audio interface on the native
>>> platforn. without this, there will be flamming of all tracks (I have
>>> tested
>>> it). Another solution would be to process only certain tracks through
>>> Forte
>>> and leave others to play back in Paris. this is possible to do, but then
>>> you
>>> get into nudging "all" tracks in Paris that are *not* being processed in
>>> Forte by a measured increment which would be the total latency of 5 x
>>> UAD-1
>>> plugins plus the 512K sample buffer. I haven't yet measured it, but I'm

>>> going to do so today. I'm thinking this may be well over the latency
>>> threshold that would allow a visual reference to the Paris timeline
> while
>>> mixing, plus, those tracks that are being processed in Forte would be
>>> nudged
>>> a different number of samples from the ones that weren't. This creates a
>>> nightmare of a messy mix scenario in my mind. If I wanted to work this
>>> way,
>>> I'd just be using the UAD-1 plugs in Paris with

Subject: Re: Oh ..... HELL YES!!!!!!!!!!!!!
Posted by animix on Mon, 02 Oct 2006 02:28:03 GMT
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> >>
> >
> >Actually John once you get things setup and create a template in both apps
you're good to go. Deej walked me through his lastest setup (configured to
my modules) and we were pretty well done in 15 minutes with an additional
1/2 hour on my part to make sure all components were playing nicely with
each other.

Funny thing was I was able to use this basic template in two other apps -
Saw studio (demo) and Reaper (demo) with no real problems at all.

Don

"John" <no@no.com> wrote in message news:45321abd@linux...
> wow, what a pain in the ass !
>
> J wrote:
>> That's correct. this is what I'm doing in Cubase SX right now. If I track
>> in
>> Paris (which I love for tracking) then render the files, fly them overmy
>> network, batch convert them to .wav format in Wavelab, then import them
>> into
>> a Cubase project to be processed and then set the SX tracks on mono
>> output
>> busses to stream the tracks back over lightpipe to Paris for further
>> processing, panning and summing, then the system is very stable. It takes
>> 30
>> minutes to an hour per song (24 tracks average) to get this going. It's
>> almost second nature to me now actually, but I'd like to avoid having to
>> jump through the initial hoops of rendering, batch conversion in WL and
>> importing into Cubase. With my Paris tracking template set up with the

>> EDS
>> external inserts and the Sampleslide native insert ready to go and
>> bypassed
>> until mix time, I could just enable the inserts, highlight all Paris
>> tracks
>> and nudge them 10 + 1, click on my mix patchbay setup in Paris, open
>> Forte
>> to the default rack processor template and I'm mixing a 40 track project
>> in
>> 5 minutes with delay compensated UAD-1 plugins and external hardware on
>> inserts and auxes in Paris.
>>
>> Deej
>>
>>
>> "Don Nafe" <dnafe@magma.ca> wrote in message news:453152c8$1@linux...
>>> hence the need for such a large I/O interface?
>>>
>>> Don
>>>
>>> "DJ" <notachance@net.net> wrote in message news:45314e37@linux...
>>>> Dave,
>>>>
>>>> All tracks are being played back through Paris. The point of this is to
>>>> have
>>>> all tracks crossing the editor timeline simultaneously whether they are
>>>> being processed by a plugin, or not. In order to achieve this, each
>> track
>>>> has to be sent to a Forte bus, processed by the UAD-1 Delaycomp (and
>>>> whatever else) then returned to Paris. the "nudge" and then the
>>>> Sampleslide
>>>> instance in Paris is in order to cover the latency between the two
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>>>> caused by the 512k buffer settings of the audio interface on the native
>>>> platforn. without this, there will be flamming of all tracks (I have
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>>>> it). Another solution would be to process only certain tracks through
>>>> Forte
>>>> and leave others to play back in Paris. this is possible to do, but
>>>> then
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>>>> get into nudging "all" tracks in Paris that are *not* being processed
>>>> in
>>>> Forte by a measured increment which would be the total latency of 5 x
>>>> UAD-1
>>>> plugins plus the 512K sample buffer. I haven't yet measured it, but I'm
>>>> going to do so today. I'm thinking this may be well over the latency
>>>> threshold that would allow a visual reference to the Paris timeline

>> while
>>>> mixing, plus, those tracks that are being processed in Forte would be
>>>> nudged
>>>> a different number of samples from the ones that weren't. This creates
>>>> a
>>>> nightmare of a messy mix scenario in my mind. If I wanted to work this
>>>> way,
>>>> I'd just be using the UAD-1 plugs in Paris with the FXPansion 3.3
>> wrapper.
>>>> I
>>>> want all lateny to be consistent so that it can be consistently
>>>> compensated
>>>> rather than doing it differently *per track*, otherwise it defeats the
>>>> purpose to my way of thinking.
>>>>
>>>> Yeah....I'm wayyy to picky, I know.
>>>>
>>&gt

Subject: Re: Oh ..... HELL YES!!!!!!!!!!!!!
Posted by LaMont on Mon, 02 Oct 2006 03:03:07 GMT
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going on here now, the applications themselves are
> very
> stable on their own but there is all sorts of potential for clocking
> errors.
> One of the main reasons for this (I'm almost certain) is because of the
> sample delay between EDS cards. Signals that are routed between the
> digital
> I/O of the RME cards from ADAT modules on MECs attached to Cards B/C/D are
> almost certainly receiving this sample latency. It's a wonder this even
> works at all. Add to it the fact that the RME cards are all receiving
> their
> clock signal from MEC ADAT modules and the modules are on cards A & B,
> it's
> almost certain that the RME card that is attached to the MEC on Card B is
> receiving a clock signal that is not sample accurate with the signal that
> the other two RME cards are receiving from the ADAT modules on card A.
> Now,
> let's throw into this whole equatiion the fact that I've got three
> outboard
> modules patched into the three S/Pdif I/O of the RME cards and these
> devices
> are receiving their clock signals from the RME cards which are clocked to
> the Paris ADAT sync and the signals that are routed/being processed
> through

> them are routed through auxes on MECs attached to Card C and D (now
> there's
> the sam

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Posted by animix on Mon, 02 Oct 2006 03:07:17 GMT
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almost certain) is because of the
> sample delay between EDS cards. Signals that are routed between the
> digital
> I/O of the RME cards from ADAT modules on MECs attached to Cards B/C/D are
> almost certainly receiving this sample latency. It's a wonder this even
> works at all. Add to it the fact that the RME cards are all receiving
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> clock signal from MEC ADAT modules and the modules are on cards A & B,
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> almost certain that the RME card that is attached to the MEC on Card B is
> receiving a clock signal that is not sample accurate with the signal that
> the other two RME cards are receiving from the ADAT modules on card A.
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> let's throw into this whole equatiion the fact that I've got three
> outboard
> modules patched into the three S/Pdif I/O of the RME cards and these
> devices
> are receiving their clock signals from the RME cards which are clocked to
> the Paris ADAT sync and the signals that are routed/being processed
> through
> them are routed through auxes on MECs attached to Card C and D (now
> there's
> the sample delay between signals routed through these procesors and the 4
> x
> EDS cards and the three RME cards.) OK...we're not done yet.....let's
> throw into the equation that one of these outboard processors, a Quantec
> Yardstick has AES I/O only and is routed through two format converters.
> Now
> let's move on to the fact that the SPdif I/O of Paris cards A, B, C, a
> Sony
> DPS, V77, a POD XP Pro, two Mytek converters and the ADAT I/O of one of
> the
> HDSP 9652 cards, the Multiface, ADAT #2 of MEC on card C and ADAT #1 of
> Card
> D and the Optical input of my Benchmark DAC-1 are all being interpatched
> through two MAudio Digipatch units which*do not* reclock the signals
> (although the Benchmark DAC-1 does).....and finally, due to the fact that
> these outboard devices are spread out in racks in various parts of
> theroom,

> many of the S/Pdif cables and lightpipe cables which are carrying (likely
> inaccurate) clock signals between these different devices are of varying
> lengths.....some of them over 20' long. It's a miracle that thiseven
> works,
> much works and sounds good to boot. I do hold my breath every time I fire
> it
> up though and I'm getting kinda tired of that.
>
> I've been told that I'm on

Subject: Re: Oh ..... HELL YES!!!!!!!!!!!!!
Posted by damien.gelee on Mon, 02 Oct 2006 20:11:05 GMT
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..com</A>>=20
  wrote in message <A=20
  =
href=3D"news:4532d7e6$1@linux">news:4532d7e6$1@linux</A>...</DIV>

=

It's=20
  really hard to describe, it has to be the room, but somewhere in=20
  the
balance of bass and treble it seems that it is pretty dang =
close in my=20
  room
to the reference CD's but I get to the car and I have to turn =
the bass=20
  down
1 and trebl up 2 clciks to get it where it felt in the house. =
but it=20
  isn't
only that, but in my control room, the mises sound clear, 3=20
  dimentional and
"airy" but the CD burns sound almost hazy, like the =

presence area has been
compressed and almost glassy in bad way, =
Dirty=20
Glassy, and somehow not as
full range. So, the EQ thing bass =
and=20
treble would be understandable,, but
this mushieness freaks me out, =
I=20
really wonder if it my conversion to 16
bit, or somehting else, =
I'll try=20
the Wavelab dither tomorrow.

my stuff still gets some nice nods =

critically, but it is not how I want it
to sound!

hey is =
anyone=20
running mixes through any mic pres? Anyone here have a=20
1968ME
compressor?

"Don Nafe" <<A =

href=3D"mailto:dnafe@magma.ca">dnafe@magma.ca</A>> wrote:
>My =
first=20
thought was "it's the room"
>
>May I ask the room =
dimensions, the=20
approximate location of your speakers
and
>acoustic=20
treatment?
>
>DOn
>
>
>"Cujo" <<A=20
=
href=3D"mailto:chris@nospamapplemanstudio.com">chris@nospamapplemanstudio=
..com</A>>=20
wrote in message=20

>news:4532b8eb$1@linux...
>>
>>
>> =
Ok, It is=20
finally start to get to me.
&a