
Subject: Re: Got the Paris stuff from John...
Posted by [DJ](#) on Thu, 09 Nov 2006 23:32:55 GMT
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>> >What next?

>>

>

>Excellent !

"lucio" <yo@kingtone.com> wrote:

>

>i hit the contacts w/ a D5 pen and let it sit for a week whilst i was travelling.

>upon return and reboot, i'm in business. Thanks!

>

>-lucio

>www.kingtone.com

>

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

>>

>>first try a simple pencil eraser to clean up the contacts.

>>

>>"lucio" <yo@kingtone.com> wrote:

>>>

>>>thanks - i'll try the silvering thing on the 1000 card.

>>>

>>>"steve the artguy" <artguy@somethingorother.net> wrote:

>>>>

>>>>another thing to try:

>>>>

>>>>My original paris card developed a case of worn contacts, and got a similar

>>>>(though I can't say if it was exactly the same) message.

>>>>

>>>>I resilvered the worn contacts with a silver contact pen from Radio Shack

>>>>and it has worked fine ever since.

>>>>

>>>>-steve

>>>>

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

>>>>>

>>>>>Reseating is a one possibility. You can also reseat the daughterboard

>>>>but

>>>>>if it was the daughterboard I think you'd get a different message.

>>>>>

>>>>>

>>>>>"lucio" <yo@kingtone.com> wrote:

>>>>>>

>>>>>>the power is on and i also checked all cables. i tried it w/ the MEC

>>off
>>>>as
>>>>>a test and the

Subject: Got the Paris stuff from John...
Posted by [neil\[1\]](#) on Sat, 11 Nov 2006 21:48:17 GMT
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/>
> ADK
> chrisl@adkproaudio.com <<mailto:chrisl@adkproaudio.com>>
> www.adkproaudio.com <<http://www.adkproaudio.com/>>
> (859) 635-5762 Mine's a 550W. No problems.

;o)

"J.S." <sonicartproductions@hotmail.com> wrote in message
news:457c8997@linux...
> Hey DJ,
> how beefy was your Power supply. I think I was actually wrong --- I think
> I actually have a 550watt powersupply. You think that is still enough?
> Jan
>
>
>
> "DJ" <

Subject: Re: Got the Paris stuff from John...
Posted by [JeffH](#) on Sun, 12 Nov 2006 07:28:00 GMT
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>
>
> Changing to a different destination (output channel) is done in any routing
> field, or by a click on the desired output pair in the bottom row.
>
>
>
> It is very easy to set up a specific submix for whatever output: select output

>
> For advanced users sometimes it makes sense to work without Submix View.
> Example: you want to see and set up some channels of different submixes simultaneously,
> without the need to change between them all the time. Switch off the Submix
> View by a click on the green but-ton. Now the black routing fields below

> the faders no longer show the same entry (A1 1+2), but completely different
> ones. The fader and pan position is the one of the individually shown routing
> destination.
>
>
>
> default setup
>
>
>
> When executing the application for the first time, a default file is loaded,
> sending all playback tracks 1:1 to the corresponding hardware outputs with
> 0 dB gain.
>
> Faders in the Hardware Inputs are set to maximum attenuation (called m.a.
> in the following), so there is no monitoring of the input channels.
>
> All faders of the middle row are set to 0 dB, so no matter on which channels
> a
> playback happens, the audio will be audible via the SPDIF output. Just try
> it!
>
>
>
> direct monitoring
>
>
>
> With ASIO direct monitoring (ADM), moving faders in Cubase will move them
> in TotalMix
>
>
>
> faders / post
>
>
>
> When you pull the fader down to the bottom the routing goes away.
>
> Think of the drop down channel list as being a rotary switch which lets one
> fader be used as multiple faders, the selection depending on where you've
> set the rotary switch.
>
> The faders can also be moved pair-wise, corresponding to the stereo-routing
> settings. This can be achieved by pressing the Alt-key and is especially
> comfortable when setting the SPDIF and analogue output level. At the same
> time.
>

> TotalMix also supports combinations of these keys. If you press Ctrl and
> Alt at the same time, clicking with the mouse makes the faders jump to 0
> dB pair-wise, and they can be set pair-wise by Shift-Alt in fine-mode.
>
>
>
> What I now realise is the input fader (and the playback faders too) are in
> essence multi-function faders i.e. you select what channel you want the
> fader to be adjusting, and the other virtual channels will not be altered.
> So, to adjust the bass guitar level at the phones output, I have to change
> the input fader (with the drop down list at the bottom of it) to "analog".
> If I have also routed the bass to a number of outputs as well, then their
> levels will remain unaffected. If I wish to alter those too, then I have
> to change the input fader to one of the other channels of the drop down
> list.
>
>
>
> grouping
>
>
>
> Click on the fader name label to turn it orange and select multiple faders.
> They are now grouped. It only works in one mixer at a time.
>
>
>
> matrix
>
>
>
> The Matrix provides true mono and is very easy to use.
>
> If you don't want to use the Matrix then use this workaround: use only odd
> or even channels as effect send. You got lots of them, so this is no limitation
> at all!
>
>
> Change gain Ctrl-drag up / down
>
> Horizontal labels: All hardware outputs
>
> Vertical labels: All hardware inputs. Below are all play back channels (software
> playback channels)
>
> Green 0.0 dB field: Standard 1:1 routing
>
> Black gain field: Shows the current gain value as dB

>
> Orange gain field: This routing is muted.
>
>
>
> menu
>
>
>
>
> Always on Top: When active (checked) the TotalMix window will always be on
> top of the Windows desktop. Note: This function may result in problems with
> windows containing help text, as the TotalMix window will even be on top
> of those windows, so the help text isn't readable.
>
> Deactivate Screensaver: When active (checked) any activated Windows screensaver
> will be disabled temporarily.
>
> Ignore Position: When active, the windows size and position stored in a file
> or preset will not be used. The routing will be activated, but the window
> will not change.
>
> ASIO Direct Monitoring (Windows only): When de-activated any ADM commands
> will be ignored by TotalMix. In other words, ASIO Direct Monitoring is globally
> de-activated.
>
> Link Faders: Selecting this option all faders will be treated as stereo pairs
> and moved pair-wise. Hotkey L.
>
> Level Meter Setup: Configuration of the Level Meters. Hotkey F2. See chapter
> 26.14.
>
> Preferences: Opens a dialog box to configure several functions, like Pan
> Law, Dim, Talkback Dim, Listenback Dim. See chapter 26.10.
>
> Enable MIDI Control: Turns MIDI control on. The channels which are currently
> under MIDI control are indicated by a colour change of the info field below
> the faders, black turns to yellow.
>
> Deactivate MIDI in Background: Disables the MIDI control as soon as another
> application is in the focus, or in case TotalMix has been minimized.
>
>
>
> meters
>
>
>
> The input meters are pre fader.

>
> The output meters are post fader.
>
>
>
> mixers
>
>
>
> Upper row: hardware inputs. The level shown is that of the input signal and
> is fader independent.
>
> Using the fader and routing windo

Subject: Re: Got the Paris stuff from John...
Posted by [Neil](#) on Sun, 12 Nov 2006 07:38:58 GMT
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). Never had problems before, but today PARIS won't boot
>>>up.
>>>>>>>
>>>>>>>I get a 'Error initializing PARIS Engine Error code -43/fffffd5
>>>>>>>
>>>>>>>Tried replacing config file, reinstalled PARIS on another drive,
cleaned
>>>>>>>things up, etc all with the same problem. Help!
>>>>>
>>>>
>>>>
>>>
>>
>>
>Greetings :)

Check out this new UAD / Euphonix piece :
<http://www.uaudio.com/webzine/2006/october/index9.html>

Interview about the design etc...
<http://www.uaudio.com/webzine/2006/october/index3.html>

We will certainly have a group buy on this for sure :)

Check out the UAD holiday specials ending Dec 31
<http://www.uaudio.com/webzine/2006/october/index9.html>
* Group buy is still going on -

Morgan

Eastcoast Music Mall
800-901-2001
morganp@ntplx.netHi John,

Nice job on this - We are going to have Tom Sailor < RME USA >
here very soon.

Total mix confuses the heck out of everyone !
You have done a fine job of laying this out :)

Please email me - I have something to run by you

Thanks
Morgan
morganp@ntplx.net

John wrote:

> For those of you using Totalmix with RME hardware I submit my notes to try
> to make it easy to use this cool thingy.
>
>
> totalmix
>
>
>
> Totalmix is a basically a mixer with two input banks and one output bank.
> The two input banks are the Hardware Inputs and the Software Inputs.
>
> #I/O
> Hardware Inputs represent the signal coming in from your external hardware
> where Software Inputs represent the sound coming from your application like
> Cubase or Winamp. You can route any input to any output.
>
> #Normal and Submix mode
> The main mixer screen has two display modes, normal and submix. When the
> Submix button on the right is off you are in normal mode. When it is on,
> you are in Submix mode.
>
> In Submix mode, you click on a hardware output pair (in submix view), and
> all the faders show their routing for that output. Each output pair can have
> its own mix.
>
> #Matrix mode
> In Normal mode there is a drop down on the third row under each fader that
> allows you to select multiple output channels to assign the input to. Pressing
> the "x" key on the keyboard takes you to a third view called Matrix view.
> Matrix view allows you to see every connection at one time.

>
>
>
> _modes
>
>
>
> normal view
>
>
>
> To enable inputs, you do have to go back to normal mode and turn up those
> inputs (top row)
>
> The non-submix view lets you view routings for specific channels to specific
> outputs, but i need find it necessary to use it.
>
> The normal view show's the submixlevels for the 1:1 routing. So if you made
> a submix for hardware outputs 5&6, the normal view will show the same level
> for software outputs 5&6. Btw. I always use the matrix to see what's going
> on.
>
> Clicking in the routing box under the fader allows you to pick multiple routing
> outputs. The fader/pan will change to reflect the value being sent to that
> route destination. I think !
>
>
>
> submix view
>
>
>
> You click on a hardware output pair (in submix view), and all the faders
> show their routing for that output. Each output pair can have its own mix.
>
> When you select submix view, you can select a pair of outputs on row 3, and
> ONLY signals routed to those outputs are shown on the upper rows. Personally,
> I find this simplifies things considerably.
>
> If you are getting something at an output that shouldnt be there, or not
> getting something that should be there, submix view will show you why.
>
>
> Submix sets all routing windows to the same selection. Deactivating Submix
> automatically recalls the previous view.
>
>
> In this mode, all routing fileds jump to the routing pair just being selected.

- > You can then see immediately, which channels, which fader and pan settings
 - > make a submix (for example 'A1 7+8'). At the same time the Submix View simplifies
 - > setting up the mixer, as all channels can be set simultaneously to the same
 - > routing destination with just one click.
-

Subject: Re: Got the Paris stuff from John...

Posted by [Neil](#) on Sun, 12 Nov 2006 17:58:23 GMT

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- >
- >
- > presets
- >
- >
- > Presets are stored in /documents and settings/"your user name"/local settings/application
- > data/rme totalmix/
- >
- > The preset buttons can get meaningful names in the same way. Move the mouse
- > above a preset button, a right mouse click will bring up the dialog box.
- > Note that the name shows up as tool tip only, as soon as the mouse stays
- > above the preset button.
- >
- > The preset button names are not stored in the preset files, but globally
- > in the registry, so won't change when loading any file or saving any state
- > as preset. But loading a preset bank (see chapter 26.8) the names will be
- > updated.
- >
- > TotalMix includes eight factory presets, stored within the program. The user
- > presets can be changed at any time, because TotalMix stores and reads the
- > changed presets from the files preset11.mix to preset81.mix, located in Windows'
- > hidden directory >Documents and Settings, <Username>, Local Settings, Application
- > Data, RME TotalMix<. On the Mac the location is in the folder >User, <Username>,
- > Library / Preferences / Hammerfall DSP<. The first number indicates the current
- > preset, the second number the current unit.
- >
- > This method offers two major advantages:
- > Presets modified by the user will not be overwritten when reinstalling
- > or updating the driver The factory presets remain unchanged,
- > and can be reloaded any time.
- >
- >
- > Restoring Defaults
- > Mouse: The original factory presets can be reloaded by holding down the Ctrl-
- > key and clicking on any preset button. Alternatively the files described
- > above can
- > be renamed, moved to a different directory, or being deleted.

>
> Keyboard: Using Ctrl and any number between 1 and 8 (not on the numeric
> keypad!) will load the corresponding factory default preset. The key Alt
> will load
> the user presets instead.
>
>
> Preset 1
> Description: All playback channels routed 1:1, monitoring of all playback
> channels.
>
> Details: All inputs maximum attenuation. All playback channels 0 dB, routed
> to the same output. All outputs 0 dB. Level display set to RMS +3 dB. View
> Submix active.
>
> Note: This preset is Default, offering the standard functionality of a I/O-card.
>
>
>
> Preset 2
> Same as Preset 1.
>
> Preset 3
> Description: All channels routed 1:1, input and playback monitoring via outputs.
> As Preset 1,
> but all inputs set to 0 dB (1:1 pass through).
>
> Preset 4
> Description: All channels routed 1:1, input and playback monitoring via outputs.
> As Preset 3, but all inputs muted.
>
> Preset 5
> Description: All faders maximum attenuation. As Preset 1, but all playbacks
> maximum attenuation.
>
> Preset 6
> Description: Submix on SPDIF at -6 dB. As Preset 1, plus submix of all playbacks
> on SPDIF.
>
> Preset 7
> Description: Submix on SPDIF at -6 dB. As Preset 1, plus submix of all inputs
> and playbacks on SPDIF.
>
> Preset 8
> Description: Panic. As Preset 4, but playback channels muted too (no output
> signal).
>
>

> Preset Banks
> Instead of a single preset, all eight presets can be stored and loaded at
> once. This is done via
> Menu File, Save All Presets as and Open All Presets (file suffix .mpr). After
> the loading the
> presets can be activated by the preset buttons. In case the presets have
> been renamed (see
> chapter 26.11), these names will be stored and loaded too.
>
>
>
> The preset buttons can get meaningful names in the same way. Move the mouse
> above a preset button, a right mouse click will bring up the dialog box.
> Note that the name shows up as tool tip only, as soon as the mouse stays
> above the preset button.
>
> The preset button names are not stored in the preset files, but globally
> in the registry, so won't change when loading any file or saving any state
> as preset. But loading a preset bank (see chapter 26.8) the names will be
> updated.
>
>
>
> set fader to zero
>
>
>
>
> When you want to set the fader to exactly 0 dB, this can be difficult, depending
> on the mouse configuration. Move the fader close to the 0 position and now
> press the Shift-key. This activates the fine-mode, which stretches the mouse
> movements by a factor of 8. In this mode, a gain setting accurate to 0.1
> dB is no problem at all.
>
>
>
>
> set multiple channels
>
>
>
> Often signals are stereo, i. e. a pair of two channels. It is therefore
> helpful to be able to make the routing settings for two channels at once.
>
>
> Press the Ctrl-key and click into the routing window of 'Out 3' with the
> key pressed. The routing list pops up with a checkmark at '3+4'. Click onto
> 'Analog'. Now, channel 4 has already been set to 'Analog' as well.
>
>

>
> shortcut keys
>
>
> F12, the cpu and disk meter
>
> #toggle Matrix view
> X
>
> #toggle visible or not for Input, Playback, Output, Submix
> I, P, O, S
>
> #Fader
> Set to 0 dB Ctrl-click faders
> Set to -6dB for hardware outputs Ctrl-click faders
> Center pans Ctrl-click pans
> Fine Control Shift-drag
>
>
> #Stereo
> Set faders pairwise in fine mode Shift-Alt
> Move faders or pans in stereo Alt-drag
> Faders jump to 0 dB pair-wise Ctrl-Alt-drag
>
>
> #Presets.....
> Set Preset to default Ctrl-click on preset button
> Load preset Alt-preset_number
>
> #level meter setup dialog
> F2
>
> #preferences
> F3
>
> #toggle Mute Master
> M
>
> #toggle mixer view
> T
>
> #link all faders as stereo pairs
> L
>
>
> #meters
> #Display range 40 or 60 dB
> Key 4 or 6

>
> #Numerical display showing Peak or RMS
> Key E or R
>
> #RMS display absolute or relative to 0 dBFS
> Key 0 or 3
>
> #Numerical display selectable either Peak or RMS
> Hotkey E or R
>
> #Measuring SNR (Signal to Noise) requires to press R (for RMS) and 0 (for
> referring to 0 dBFS, a full scale signal). The text display will then show

>
>
>
>"LaMont" <jjdpro@gmail.com> wrote:
>
>Neil, In SX/Neundo, can you hear the difference btw 24bit vs 82?

You mean 32, right? And the answer is: I don't know, in terms of tracking a whole project at 32-bit, never done that... I use 24-bit 88.2k to track & if I'm mixing down to stems I'll do the stems at 32-bit. Same thing if I'm going straight to a 2-buss mix t

Subject: Re: Got the Paris stuff from John...
Posted by [excelav](#) on Sun, 12 Nov 2006 19:50:04 GMT
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entially: "What if I can sum in the digital domain via DSP;
>>>and if so, would that sound better than what I can do right
>>>now?" At this point I still can't find out, however, due to the
>>>inability of the Pulsar stuff to work at 88.2k.
>>>
>>>If you want to see the minor shitstorm that DeeJ & I started
>>>over on the Pulsar forum over this issue, go here:
>>>
>>><http://www.planetz.com/phpBB2/viewtopic.php?t=20885>
>>>
>>>
>>>Neil
>>>
>>>
>>
>>
>>

>>
>
>--
>Chris Ludwig
>ADK
>chrisl@adkproaudio.com <mailto:chrisl@adkproaudio.com>
>www.adkproaudio.com <http://www.adkproaudio.com/>
>(859) 635-5762When you make you choice please let us know. I'm always looking for something better portable wise. Also if you can post what your running on etc.

Thanks

So far the Emu 1616M wit sx3 has worked well as portable solution for me...

"Nappy" <mgrant01@san.rr.com> wrote:

>
>http://www.tcelectronic.com/Konnekt24D
>
>Havent tried it,but it looks promising. It can also be used as a standalone
>effects
>unit. Street price is \$399 I think? MOTU has a unit with the simalar spec
>without
>effects. <http://www.motu.com/products/motuaudio/ultralite/>
>
>respect
>Nappy

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

>>Hi All
>>
>>I'm looking for an inexpensive way to get adat and/or optical inputs into
>a
>>laptop...new or used
>>
>>any suggestions
>>
>>

>Kind of newbie questions, but I wonder, how hot you guys hit the
paris convertors for a 24bit recording. I read that hitting 0dB
doesn't make sense in a 24bit recording, because you have enough
headroom available to go softer, but where is the under limit or optimal
level?

The Paris 8in has trim pots on the rear, does it make sense to
lower these inputs to make more gain at the mic preamp stage and
instead of using a pad?

Thanks

SimonThe resolution doesn't really matter to me - I always try to maximize my recording level (without clipping, of course).

rock on,
-Carl

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

>

> Kind of newbie questions, but I wonder, how hot you guys hit the
> paris convertors for a 24bit recording. I read that hitting 0dB
> doesn't make sense in a 24bit recording, because you have enough
> headroom available to go softer, but where is the under limit or optimal
> level?

> The Paris 8in has trim pots on the rear, does it make sense to
> lower these inputs to make more gain at the mic preamp stage and
> instead of using a pad?

> Thanks

> Simon

>

>Thad, from what I can tell you probably won't have any stability
issues with the Pulsar stuff... I was playing with it at 44.1k
and the only thing I nticed is that a couple of plugins sure
seem to take awhile to load (like ten or 15 seconds - strange
when it's got all that DSP power on-board), but they
don't "hang" once they're loaded & you start using them or
anything like that.

As for your other comment about spending tons of \$\$\$ for a
vintage mic or pre, but not being willing to fork it over for
some other core need... I'm with you there - this is why I'm
trying the Pulsar stuff - mt feelinds are that I've got killer
signal chain stuff happening in terms of mics & pres, pristine
hi-rez convertors, but I'm still not able to get the full mix
sound that I'm looking for... short of going PTHD (which I
suppose I could do, but that's just a TON of money); the Pulsar
stuff - IF I can get it to work at 88.2 - maybe has a shot at
getting me there.

Neil

"TCB" <nobody@ishere.com> wrote:

>

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

>

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

>

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

>

>TCB

>

>Chris Ludwig <chrisl@adkproaudio.com> wrote:

>>Hi Dj and Neil,

>>It's sadly amusing to see that Creamware's way of doing things hasn't
>>changed at all sense I dealt with them years ago. Some of the user
>>responses you all got are are hilarious in their denial of facts.
>>Any of the effects available could be coded and ran in a current native
>
>>system. Creamware, TC and UAD types do not want to do it for the simple
>
>>reason people could easily steal the software. The big thing that the
>>Creamware DSP cards offer which same goes for Pro Tools is the ability

>>to do it very low latencies so as to make is seem real time. The UAD
>>and TCs don't even offer this and actually quite the opposite. In native
>
>>systems you can't run the latencies to their lowest settings when using
>
>>these plug ins.

>>

>>I have a feeling that the reason the does not do 88.2k is because of
>>hardware limitations. It may be that the clocks they use on their cards
>
>>which are fairly old right now may not have had the ability to support

>>the 88.2k clock. People as fair as I've seen didn't really start trying
>
>>to use 88.2k till the past 2/3 years but 96k has been around a couple
>>more years so has been more commonly available on digital clocks.

>>

>>

>>Chris
>>
>>DJ wrote:
>>
>>><evil grin>
>>>
>>>"Neil" <IUOIU@OIU.com> wrote in message news:457c65bb\$1@linux...
>>>
>>>
>>>>"Don Nafe" <dnafe@magma.ca> wrote:
>>>>
>>>>
>>>>>I guess the big question is are you taking an unacceptable sonic hit
>a
>>>>>44.1
>>>>>
>>>>>
>>>>>vs 88.2 and does the summing using the Pulsar offset the sonic hit you
>
>>>>>take
>>>>>
>>>>>
>>>>>(if in fact

Subject: Re: Got the Paris stuff from John...
Posted by [Tom Bruhl](#) on Sun, 12 Nov 2006 22:10:35 GMT
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t;Got me curious now, I'll have to check that out.
>Think it was -3dB avg
>
>Rob
>
>
>
>
><Simon hexagometer (at) gmx.net> wrote in message news:457d8136\$1@linux...
>>
>> Kind of newbie questions, but I wonder, how hot you guys hit the
>> paris convertors for a 24bit recording. I read that hitting 0dB
>> doesn't make sense in a 24bit recording, because you have enough
>> headroom available to go softer, but where is the under limit or optimal
>> level?
>> The Paris 8in has trim pots on the rear, does it make sense to
>> lower these inputs to make more gain at the mic preamp stage and
>> instead of using a pad?
>> Thanks

>> Simon

>>

>>

>Is there an easy way to figure out if they do? I'm a little fuzzy here.

Thanks for the continued efforts.

"John" <no@no.com> wrote in message news:457d4728\$1@linux...

>

> Yes, as long as the slots don't share IRQs.

> John

>

> "Uptown Jimmy" <johnson314@bellsouth.net> wrote:

> >I'm a little confused here.

> >

> >I should choose, say, IRQs 3, 5 and 7 and assign them to "legacy device"

> in

> >the 1st section, then assign my PCI slots to those IRQs in the 2nd section?

> >

> >Jimmy

> >

> >

> >"John" <no@no.com> wrote in message news:456ecae6\$1@linux...

> >>

> >> You have three cards right? Do you have three slots that don't share

> IRQs

> >> with other devices? If you do have a slot that shares with something

> like

> >> a sound card you can turn off that device in the bios. Then set bios

> to

> >> legacy devices and put a different IRQ on each on of your EDS card slots.

> >>

> >> John

> >>

> >> "uptown jimmy" <johnson314@bellsouth.net> wrote:

> >> >

> >> >Sorry to keep starting new threads on this topic. My internet access

> was

> >> down

> >> >for a while there.

> >> >

> >> >Following the advice of several folks here, in an attempt to get each

> PCI

> >> >slot its own IRQ, I have some data and a question.

> >> >I have PnP O/S disabled. I found the following info, in two

sections,in
 > >> my
 > >> >BIOS:
 > >> >
 > >> >1. a list of these IRQs, with the choice of "PCI device" or "reserved
 > for
 > >> >legacy ISA devices": 3,4,5,7,9,10,11,14,15.
 > >> >
 > >> >2. PCI slot 1/5 IRQ preference
 > >> > PCI slot 2 IRQ preference
 > >> > PCI slot 3 IRQ preference, all with the choice of "Auto" or the
 option
 > >> >to assign an IRQ #.
 > >> >
 > >> >What next?
 > >>
 > >
 > >
 >My buddy Swen has decided that it's best to keep the submix faders pegged at
 the top of their travel, and adjust for distortion in the submixes
 themselves, on individual mixer channels.

I always thought Pairs responded better the exact opposite way, ie, keep the
 submix faders down a bit and push the individual mixer channels to taste.

Anybody?

JimmyI was referring to software conversion, as in wavelab,not sample rate
 converters
 "Jesse Skeens" <jskeens@gmail.com> wrote in message news:457d2827\$1@linux...
 >
 > "alex plasko" <alex.plasko@snet.net> wrote:
 >>less quantization errors. when you downsample to 44.1 it being 1/2 ,
 >>instead
 >
 >>of 96/44.1. same thing with 48/44.1.plus he gets the nyquist frequency
 > well
 >>into doggy ear range recording at 88.2
 >
 > That's a myth. Sample rate converters upsample to a common rate and then
 > downsample to the choosen one while also filtering.
 >Clip in Mec inputs bad
 This corresponds with prefader meters.

Clip in postfader meters is ok.

To get close to an accurate reading, turn off any insert FXs, unclick the
 all-EQ switch, make sure trim is set to zero, and set "show meters as post

fader" to off. Any of these could have an impact on the LED/meter, but don't impact the actual signal being recorded to disk. Now set input gain. Take the signal to point of clipping and watch your hardware LEDs on the MEC. The hardware and software clipping indicators should be quite close. Then set gain so that no clipping occurs.

===

1. The global master meter does not have clip indicators
2. The global master meter sometimes reads incorrectly based on the monitor level
3. If you are using Waves plug-ins on the mac, you cannot count on your settings be restoring correctly during on update.

So basically, sometimes for final mixes, I will "bounce" each submix, then create a new submix and add in all of my bounced files and do a final bounce of that. That way I can use the submixes meter as the master, and I do not have the problem with waves plug-ins

===

Mixer window- under the 'settings' menu at the top, uncheck (that is, turn off) 'Show meters as post fader.' The meters are now pre-fader. The meter will now read whatever signal strength is hitting the input regardless of where you set the fader. Control the signal level with the output of the Sans-amp or whatever signal source you're using. Also, be aware that there is a trim knob in the expanded view of the eq section. Keep in mind that this gain knob never changes the recorded signal, it can only process the output.

Even if you have the meter set to "pre-fader", the meter will show the compression. So if you are recording with compression turned on, make sure that you set the levels with it turned off. or your levels will be wrong.

===

For a workaround, create a new project and import all your submixes from the working project into a single submix. You now have your headroom indicators.

I was wondering what everyone does to deal with the fact that the master "global" meter does not have a head room indicator.

I would like to get the output as hot as possible, but I have no way of knowing if I'm getting any "overs".

Obviously this is not a problem if you only have one submix, but in a multi submix project, it gets tricky.

===

1. The meters in PARIS are peak meters. You can't just "go well into the yellow" 2 get a hot mix that averages --10. You have to stay "way" into the yellow with all the peaks riding into the red. Unless you are doing this

it's very likely your average program is -10 or below. The main rule of thumb is never to have an over. If you are going to be doing further eq processing in another program it's best to leave the meters peaking consistently around -3.

2. In a single submix make absolutely certain that the Global Masters Meters correspond exactly to the meters in your submix. If they do not check for plugins on the Global Master that are reducing gain. Key plugin here would be compression.

===

The clip lights on the MEC are pre conversion and indicate analog clipping. Oh, also clipping eds and native effects is bad> I was wondering what everyone does to deal with the fact that the master

> "global" meter does not have a head room indicator.

> I would like to get the output as hot as possible, but I have no way of

> knowing if I'm getting any "overs".

> Obviously this is not a problem if you only have one submix, but in a multi

> submix project, it gets tricky.

Hey John,

Not sure if this helps, you're probably already aware of the plug, AND, this does not directly answer your question, but.... are you aware of a plug called "RMS Buddy?" I have used this on stems and masters to get a sense of overall output. It's been a very helpful little tool.
MRSo was I.

"alex plasko" <alex.plasko@snet.net> wrote:

>I was referring to software conversion, as in wavelab,not sample rate

>converters

>"Jesse Skeens" <jskeens@gmail.com> wrote in message news:457d2827\$1@linux...

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>>>of 96/44.1. same thing with 48/44.1.plus he gets the nyquist frequency

>> well

>>>into doggy ear range recording at 88.2

>>

>> That's a myth. Sample rate converters upsample to a common rate and then

>> downsample to the choosen one while also filtering.

>>

>

for hard drives or DSP (or that you have a second Firewire port.)

"Rich" <studiodog_99@yahoo.com> wrote:

>

>When you make you choice please let us know. I'm always looking for something
>better portable wise. Also if you can post what your running on etc.

>

>Thanks

>

>So far the Emu 1616M wit sx3 has worked well as portable solution for me...

>

>

>"Nappy" <mgrant01@san.rr.com> wrote:

>>

>><http://www.tcelectronic.com/Konnekt24D>

>>

>>Havent tried it,but it looks promising. It can also be used as a standalone

>>effects

>>unit. Street price is \$399 I think? MOTU has a unit with the simalar spec

>>without

>>effects. <http://www.motu.com/products/motuaudio/ultralite/>

>>

>>respect

>>Nappy

>>

>>

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

>>>Hi All

>>>

>>>I'm looking for an inexpensive way to get adat and/or optical inputs
into

>>a

>>>laptop...new or used

>>>

>>>any suggestions

>>>

>>>

>>

>That's a great tool !

"Mike R." <emarenot@yahoo.com> wrote:

>

>> I was wondering what everyone does to deal with the fact that the master

>> "global" meter does not have a head room indicator.

>> I would like to get the output as hot as possible, but I have no way of

>> knowing if I'm getting any "overs".

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>Hey Gene! Is there any way you could test and give me an idea of the round
trip latency of the ProFire Lightbridge? I'm thinking about using one to
move tracks back and forth between Paris and Cubase SX for processing. I'm
sure FW will add latency, the question is, will it be impractical?

Thanks
James

"Gene Lennon" <glennon@nospMyrealbox.com> wrote:
>

with

>for hard drives or DSP (or that you have a second Firewire port.)
>
>
>
>"Rich" <studiodog_99@yahoo.com> wrote:
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>>When you make you choice please let us know. I'm always looking for something
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>>>respect
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>>>
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>>>"Don Nafe" <dnafe@magma.ca> wrote:
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>>>>
>>>>I'm looking for an inexpensive way to get adat and/or optical inputs
>into
>>>a
>>>>laptop...new or used
>>>>
>>>>any suggestions
>>>>
>>>>
>>>
>>
>I'm referring ONLY to mixing here, not tracking.

I forbid any clipping during tracking, period.

Jimmy

"John" <no@no.com> wrote in message news:457ddb84\$1@linux...

>
> Clip in Mec inputs bad
> This corresponds with prefader meters.
>
> Clip in postfader meters is ok.
>
> To get close to an accurate reading, turn off any insert FXs, unclick the
> all-EQ switch, make sure trim is set to zero, and set "show meters as
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> will now read whatever signal strength is hitting the input regardless of

> where you set the fader. Control the signal level with the output of the

> Sans-amp or whatever signal source you're using. Also, be aware that

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> this gain knob never changes the recorded signal, it can only process the

> output.

>

>

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> So if you are recording with compression turned on, make sure that you set

> the levels with it turned off. or your levels will be wrong.

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> yellow" 2 get a hot mix that averages --10. You have to stay "way" into
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> yellow with all the peaks riding into the red. Unless you are doing this

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thumb

> is never to have an over. If you are going to be doing further eq
processing

> in another program it's best to leave the meters peaking consistently
around

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>
> 2. In a single submix make absolutely certain that the Global Masters
Meters
> correspond exactly to the meters in your submix. If they do not check for
> plugins on the Global Master that are reducing gain. Key plugin here would
> be compression.
> ===
> The clip lights on the MEC are pre conversion and indicate analog
clipping.
> Yes.. I've been watching this product. It looks to be a great front end to
a ITB studio. I would've liked to see at least 1 fader. > Ala Faderport, but
as it stands, it's a pro unit..

Morgan <morganp@ntplx.net> wrote:

> Greetings :)
>
> Check out this new UAD / Euphonix piece :
> <http://www.uaudio.com/webzine/2006/october/index9.html>
>
> Interview about the design etc...
> <http://www.uaudio.com/webzine/2006/october/index3.html>
>
> We will certainly have a group buy on this for sure :)
>
> Check out the UAD holiday specials ending Dec 31
> <http://www.uaudio.com/webzine/2006/october/index9.html>
> * Group buy is still going on -
>
>
> Morgan
> Eastcoast Music Mall
> 800-901-2001
> morganp@ntplx.net
> Noticed this today in the new Sweetwater catalog... looks like
a pretty good deal, appears to have internal summing to a
stereo output (see back panel pic) would be GREAT for something
like several toms you don't want to print to a separate track
for each mic...

[http://www.sweetwater.com/store/detail/8pre/your right, interpolation /decimation is the common
practice](http://www.sweetwater.com/store/detail/8pre/your%20right%2C%20interpolation%20decimation%20is%20the%20common%20practice). thanks for

pointing that out jesse:-)

"Jesse Skeens" <jskeens@gmail.com> wrote in message news:457de6fc\$1@linux...

>
>
> So was I.
>

> "alex plasko" <alex.plasko@snet.net> wrote:
>>I was referring to software conversion, as in wavelab,not sample rate
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>>"Jesse Skeens" <jskeens@gmail.com> wrote in message
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>>>
>>>>of 96/44.1. same thing with 48/44.1.plus he gets the nyquist frequency
>>> well
>>>>into doggy ear range recording at 88.2
>>>
>>> That's a myth. Sample rate converters upsample to a common rate and
>>> then
>>> downsample to the choosen one while also filtering.
>>>
>>
>>
>Saw this on the Pulsar forum... it's FREEEEEEEE(2 frikkin' gigs
worth for free? damn!)

<http://www.soundsonline.com/Free-Virtual-Instruments-Sampler-Disc-pr-EW-D2.html>No problem,
but you will need to wait a few days. The Lightbridge in part

finished.
Gene

"James McCloskey" <excelsm@hotmail.com> wrote:
>
>Hey Gene! Is there any way you could test and give me an idea of the round
>trip latency of the ProFire Lightbridge? I'm thinking about using one to
>move tracks back and forth between Paris and Cubase SX for processing.
I'm
>sure FW will add latency, the question is, will it be impractical?
>
>Thanks
>James
>That does look nice but how good are the Pres. I like the quote that he tested
it for 2 weeks and didn't have one problem. WOW !!!!! Maybe it has a
3 week warranty. kidding

I got 2 presonus digimax fs and they are only 8 each but it's worth looking
at too.

"Neil" <IUIO@OIU.com> wrote:

>
>Noticed this today in the new Sweetwater catalog... looks like
>a pretty good deal, appears to have internal summing to a
>stereo output (see back panel pic) would be GREAT for something
>like several toms you don't want to print to a separate track
>for each mic...
>
><http://www.sweetwater.com/store/detail/8pre/>
>
>
>
>Then you are on the right track. Push faders into red as to taste using
your ears. Don't clip EDS or Native effects though.
Also don't use EQ 1 as your first choice (of the 4 bands). I had some bug.

Subject: Re: Got the Paris stuff from John...
Posted by [excelav](#) on Mon, 13 Nov 2006 02:00:06 GMT
[View Forum Message](#) <> [Reply to Message](#)

track or perhaps a multi-keyboard rig to two
channels for live use, since it has line-in's as well (this is
assuming the mixing can be done without any software that might
be accessible via the FW connection).

NeilGimmie, I want to try one. Just to be clear though the 16 ins are through firewire and NOT
through the
8 ins in the back. Right ?

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

>
>Gimmie, I want to try one. I got the amp Friday. All I can say is WOW!!!
I tried it out using the 12 inch speaker in my Fender Hotrod (40watt) and
was just blown away.

What blew me away right off, was the infinite sustain and clarity I got when
I cranked it full out. Even with the master at 50 - 70% and gain full out.

Then when backing off the volume knob on my guitar, it cleaned up nicely
without the significant drop in volume I get when backing off on my Fender.
I don't use a lot of drive on that amp cause it fizzles out too much at the
low volumes I play.

It really reminded me of the sound I used to get back in the days when I
played out of the two Marshall Super Lead heads (60's vintage) I used to
have. (50 and a 100 watt heads). I still have one of the 4-12 cabinets
(with a couple of blown speakers unfortunately) so I didn't try it with that

yet.

But it's 100 watts. Doesn't it stand to reason that I would want to use a cab closer to the rated power of the amp? I wish I had a low power cab to AB it with the Fender. At 40 watts, it still sounded great. Just wonder what a cab closer to 15 watts would sound like.

BTW, even at 15 watts (or 7 for that matter), this thing still smokes louder than I usually play these days.

Thanks for turning me on to this great find!

TC <tc@spammetodeathyoubastards.org> wrote:

>Hi Paul,

>

>I've played it through a mesa 4x12, and my custom built 4x12/celstion cab, but I've

>been wanting to go THD for the last few months.

>I'm going to be getting 2 THD 2x12 cabs when they are back in stock again from the dealer.

>

>Basically with the Radial ABY I'll be able to switch channels or combine both amps at

>once, and size wise it will still be smaller than a marshall head and 4x12, and much

>easier to carry around.

>

>The day I tried the Orange I traded my Marshall JMP-1/Mesa 2:100 live setup and got

>the two Oranges and a Les Paul Studio. I've also got an AC30 that I record with.

>

>I also want to build or get an iso box for the THD cab for recording, which should work

>pretty well with the Orange. I've got an AxeTrak, but it's hard to get anything

>but that "small speaker, closed mic" sound with it. It works well for some things though,

>but you've got no play with it, as it's all fixed position.

>

>I've been looking at these options:

><http://www.amptone.com/diyisobox.htm>

><http://www.vocalbooth.com/products/ampboxes.html>

>

>Cheers,

>

>TC

>

>Paul wrote:

>> What kind of speaker cab are you running with it?
>> I bought one on EBay the other day and it should get here today.
>> Looking forward to checking it out.
>> Hope I like it as much as your do."John" <no@no.com> wrote:
>
>Just to be clear though the 16 ins are through firewire and NOT through
the
>8 ins in the back. Right ?

I have no idea - you'd probably have to call your favorite MOTU
dealer & inquire.

(although I don't see how you could get 16 total in's with only
8 physical inputs)

Neil"Neil" <IUIO@OIU.com> wrote:

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>Noticed this today in the new Sweetwater catalog... looks like
>a pretty good deal, appears to have internal summing to a
>stereo output (see back panel pic) would be GREAT for something
>like several toms you don't want to print to a separate track
>for each mic...

The deluxe version of this would be the API 3124 with the mixer built in.
I had one and it was a great sounding piece. Obvioulsy not in the same
world price-wise.

>
><http://www.sweetwater.com/store/detail/8pre/>

>
>
>
>
>OK, I finally upgrade from 19" CRTs to 22"-widescreen LCDs, but now I
can't get my Paris machine to recognize the monitor. The machine is a
1GHz Pentium III box with a Chaintech Nvidia Geforce FX5200 video
card. I am driving an Hitachi 19" CRT off the analog port of the card
and my new Acer 2216 monitor off the digital port. I have d//ed the
latest Nvidia drivers that claim to allow me to specify manually the
1680x1050x60ns resolution of the monitor, but whenever I do this, I
get a popup claiming that setting cannot be supported.

I can get things to work at 1600x900 (actually the Nvidia driver then
resets the 900 to 896!) but when I try to do anything after that, the
Acer blinks off/on for the rest of the session...

....so, can anyone suggest a video card for me? I think I am basically
screwed here, because I need:

1. AGP

2. WinME
3. WSXGA+ (i.e. 1680x1050x60s)

Looks like $2 + 3 = 0$ here!

- Paul Artola
Ellicott City, Maryland

p.s. Upgrading the Paris machine to a newer, WinXP one is not an option. Do you have Plug & Play disabled? If not, that might help. That way the driver might be able to force Windows to use the settings of your choice.

Neil

Paul Artola <artola@comcast.net> wrote:

>OK, I finally upgrade from 19" CRTs to 22"-widescreen LCDs, but now I
>can't get my Paris machine to recognize the monitor. The machine is a
>1GHz Pentium III box with a Chaintech Nvidia Geforce FX5200 video
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>- Paul Artola
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>p.s. Upgrading the Paris machine to a newer, WinXP one is not an
>option.

>Lightpipe! 8 mic, line, instrument inputs, 8 lightpipe inputs, 8 lightpipe
outputs, stereo line outputs plus headphone out. It also acts as an 8 analog
input to 8 lightpipe output converter. It looks like an 828MkII sans the
SPDIF plus six more mic pre's. The pre's on my 828MkII are all around

decent, clean, flat pre's. Not stellar by any means, but not crap either.
Here's more detailed info if anyone's interested.

<http://www.motu.com/products/motuaudio/8pre/>

Tony

"Neil" <OIUOIU@OIU.com> wrote in message news:457ed87d\$1@linux...

>

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

>>

>>Just to be clear though the 16 ins are through firewire and NOT through
> the

>>8 ins in the back. Right ?

>

> I have no idea - you'd probably have to call your favorite MOTU
> dealer & inquire.

>

> (although I don't see how you could get 16 total in's with only
> 8 physical inputs)

>

> NeilThat might be a nice addition to any system...

Regards,

DimitriosIt's also got two sets of lightpipe for up to 8 channels at 96KHz. That's
kind of cool.

Tony

"Tony Benson" <tony@standinghampton.com> wrote in message
news:457ee87c\$1@linux...

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> lightpipe outputs, stereo line outputs plus headphone out. It also acts as

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> Tony

>

>

>

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>> I have no idea - you'd probably have to call your favorite MOTU
>> dealer & inquire.
>>
>> (although I don't see how you could get 16 total in's with only
>> 8 physical inputs)
>>
>> Neil
>
>Dang, I didn't notice that... that IS pretty cool!

Anyone ever use MOTU pres? Are they decent?

Neil

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

>> "Neil" <OIUOIU@OIU.com> wrote in message news:457ed87d\$1@linux...
>>>
>>> "John" <no@no.com> wrote:
>>>>
>>>>Just to be clear though the 16 ins are through firewire and NOT through
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>>>>8 ins in the back. Right ?
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>Well, like I said, the pre's on my 828MkII are decent. They're not Gracie,
True Systems or Great River clean, but they're not junk either. Currently, I
use them for my drum overheads, percussion, acoustic guitar, etc. I don't
have a large collection of pre's and the others I have are all more
"coloring". Assuming the pre's in this unit are at least equal or better,
I'd say they be fine for a general, neutral sound. I have heard that MOTU
was using some marketing hype and might be padding their gain specs, so you
might want to investigate that if you want to use this pre with ribbon or
other low output mics. I've never had any gain problems with my 828, but I'm
only using the standard dynamic and condenser mics.

Tony

"Neil" <IUOIU@OIU.com> wrote in message news:457eebbb\$1@linux...
>
> Dang, I didn't notice that... that IS pretty cool!
>
> Anyone ever use MOTU pres? Are they decent?
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> Neil
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> "Tony Benson" <tony@standinghampton.com> wrote:
>>It's also got two sets of lightpipe for up to 8 channels at 96KHz. That's
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>>
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>>"Tony Benson" <tony@standinghampton.com> wrote in message
>>news:457ee87c\$1@linux...
>>> Lightpipe! 8 mic, line, instrument inputs, 8 lightpipe inputs, 8
>>> lightpipe outputs, stereo line outputs plus headphone out. It also acts
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>>> an 8 analog input to 8 lightpipe output converter. It looks like an
>>> 828MkII sans the SPDIF plus six more mic pre's. The pre's on my 828MkII
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>>
>The preamps on my MOTU 828MkII seem clean and reasonably transparent.
Also, the box has been very dependable and MOTU's driver updates for my
system (MacOSX) have been timely. Recommended.

Cheers,
-Jamie
www.JamieKrutz.com

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>Sorry, that would be "Grace" not Gracie. I guess I have a George Burns complex.?

Tony

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>Here are some competing products you guys should look at. The question is
can these products be used in stand alone mode as a mic pre amp. There was
some talk about the Focusrite being able to be used in stand alone mode.

The Alesis is the least expensive and also gives you a turntable in put
with a ground. All of these products have 8 mic pres and 16 Adat light pipe
I/Os, the MOTU only has 8. By the way, they are all using the same TC chip
set for the firewire. I expect TC electronic to come out with their own
26 I/O box, maybe at the NAMM show.

http://www.focusrite.com/product/saffire_pro_26_i_o/

<http://presonus.com/firestudio.html>

<http://alesis.com/product.php?id=96>

James

"Neil" <IUIO@OIU.com> wrote:

>

>Noticed this today in the new Sweetwater catalog... looks like
>a pretty good deal, appears to have internal summing to a
>stereo output (see back panel pic) would be GREAT for something
>like several toms you don't want to print to a separate track
>for each mic...

>

><http://www.sweetwater.com/store/detail/8pre/>

>

>

>

>Paul:

You may be in trouble in ME, due to driver issues. When I got

Subject: Re: Got the Paris stuff from John...

Posted by [JeffH](#) on Mon, 13 Nov 2006 02:09:22 GMT

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

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Chris Ludwig

ADK

chrisl@adkproaudio.com <<mailto:chrisl@adkproaudio.com>>

www.adkproaudio.com <<http://www.adkproaudio.com/>>

(859) 635-5762 Thanks Edna, Neil, and Larry for the responses. After playing around with the setup last night, I was able to get something happening, but it is less than optimal. First, I disconnected my CRT monitor then connected the widescreen to the computer via the analog input. I tried running the LCD solo via the digital input, but that still produced the lighthouse effect - monitor turning ON and OFF with every little action.

Next, I set the display to 1600x900xwhatever. The card seems to be indifferent to 60ns, adapter default, or optimal as the refresh setting, and I tried them each with no difference in quality. At this resolution, I get 16+2+1 nice mixer channels to the screen, but anything beyond 1 aux/eq panel requires scrolling, which is to be expected.

However, at this resolution, text is sub-optimal, but readable. Not too critical since I really only use this computer for Paris work, but in the long run, I see a new (old?) video card in my future. Ideally, I will get both of these widescreen monitors I have working together - that would really rock my Paris world, but for now, I will go with what I have and hope it is somewhat stable.

- Paul Artola
Ellicott City, Maryland

On Tue, 12 Dec 2006 22:04:59 -0600, "RiverLake Farms"
<edna@texomaonline.com> wrote:

>Paul, the GEF 5200 doesn't support that resolution. you will have to move
>up to at least the 6000 series cards. Seems like I saw a 6200 at Newegg for

>\$35.
>Edna
>"Paul Artola" <artola@comcast.net> wrote in message
>news:tbotn2hrqeut0tpkbiqliibb48sumgslaj@4ax.com...
>> OK, I finally upgrade from 19" CRTs to 22"-widescreen LCDs, but now I
>> can't get my Paris machine to recognize the monitor. The machine is a
>> 1GHz Pentium III box with a Chaintech Nvidia Geforce FX5200 video
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>> p.s. Upgrading the Paris machine to a newer, WinXP one is not an
>> option.
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>Brother -

I got your email, but my reply was kicked back for some reason. Happy Birthday, Sakis! Hope you have a wonderful year, and I look forward to catching up with you (and any others from clan Paris) at the AES show in New York.

For those who don't know, Sakis and I were born on the same day!
Clearly, mom liked him better!

Fraternally yours,
PaulHi Cujo,

My understanding is that you do need external wordclock to run more than 2 MECs / 442s. I have 3 EDS cards in my setup running 2 MECs, with nothing attached to the 3rd EDS.

Also, make sure you've got adequate ventilation in your computer - these babies get hot.

I got a set of EDS cables thru <http://www.eastcoastmusic.com> a couple of years ago, from Morgan Pettinato. Maybe they can still supply.

Cheers,

Stewart.

Phil Aiken wrote in message <45800842\$1@linux>...

>

>You just need the same little cables that your other 2 cards have. I don't
>believe you need an interface hooked up, as long as the card with no
interface

>is not tagged as your primary card within Paris. I believe it can get word
>clock passed to it internally from one of the other cards, and that is what
>one of the cables is for. Someone correct this if I am wrong, please..

>

>*****

>****Make sure your power supply is up to the task.****

>*****

>

>Can't answer the bad card question.....

>

>

>"Cujo" <chris@nospamapplemanstudio.com> wrote:

>>

>>

>>I currently have 2 EDS, someone just traded me an MEC and EDS and a C16
>for

>>some work..Well, there was more to the trade no matter.

>>

>>What Do I need to hook it all up..I have a LA Audio WC generator..does the
>>3rd EDS have to be hooked to an interface at all I assume it does. do need
>>the little ribbon cable..any place sell em?

>>

>>Also, I remember this EDS card may have issues..is there a danger in
putting

>>it in if it is a bad card or will it just not work?

>>

>>Thanks all

>>C

>From what I've seen of Vista it's nothing to write home about. But if MSoft
was looking to strive for a better GUI (Vista isn't about a better GUI, IMHO,

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But my guess is that Vista will be the end of the 'OS wars.' Sun had it right (just way ahead of their time, which is nearly as bad as being late), the computer is the network. Neither Apple nor MSoft has been a leader in providing new and exiting ways to connect people with information or entertainment. Bittorrent, Google, YouTube, MySpace. Repeat after me, Bittorrent, Google, YouTube, MySpace . . .

TCB

"James McCloskey" <excelsm@hotmail.com> wrote:

>

>The truth leaks out.

>

> http://www.forbes.com/2006/12/12/apple-microsoft-mac-tech-cz_dl_1212mac.html?partner=yahootix run 6 EDS cards - with one MEC, and various other interfaces depending on what I need. Seems like I remember chaining my wordclock output from my "A" interface to all of the other ones (before I had an external wordclock). You do need the EDS 10 and 16 pin jumper cables though. You can actually make these yourself with ribbon cable and connectors - I actually might have some laying around here.

rock on,
-Carl

"Sound Dog" <dogster@tpg.com.au> wrote in message news:45801b9d\$1@linux...

> Hi Cujo,

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> MECs / 442s. I have 3 EDS cards in my setup running 2 MECs, with nothing
> attached to the 3rd EDS.

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> putting
> >>it in if it is a bad card or will it just not work?
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> >>Thanks all
> >>C
> >
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>Hey Paul,

Nvidia uses a substandard clock on the 5200 fx (and maybe the 6200, I'm not positive so be careful) cards. Because of the clocking problems I think they clunk out when the pixel count gets above 1600x1200 regardless of aspect ratio. I forget the specifics but I discovered this when I tried to get my 24" Dell running on my new Debian box and, lo and behold, no love from the 5200 fx.

In the nVidia line the bang for the buck price point is usually the \$95-125 range which right now means probably a 7300 or 7600. They are pretty fancy cards for an audio box, and there are 7300 models that are passively cooled to keep noise down.

Good luck,

TCB

Paul Artola <artola@comcast.net> wrote:

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>- Paul Artola

> Ellicott City, Maryland

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>p.s. Upgrading the Paris machine to a newer, WinXP one is not an
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>Happy Birthday to both of you!

respect

Nappy

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>Hey Thad,

I'm curious, what of OSX is "busy bubbling gunk" and what is functional yet aesthetic presentation of useful UI elements?

I realize you don't actually work with OSX, but what I see every day is more the latter than the former.

Are you involved in the GUI initiatives on Linux in order to prevent the creation of an unmitigated disaster along the lines of your impression of OSX? :^)

Also, clearly Apple actually has been a leader, in some areas, in connecting people to info and entertainment: iTunes, iPods, their remote control thingie (and perhaps the upcoming iTV).

Not to take anything away from Microsoft and their Xbox strategy - or from Google, Bittorrent, YouTube or MySpace, ranging from providing value-added services to exploiting content producers and encouraging theft to drive ad sales or build value. On a pure technical level, for content distribution the Bittorrent approach is compelling.

Oh, and you forgot AOL. ;^)

Cheers,

-Jamie

www.JamieKrutz.com

TCB wrote:

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>> http://www.forbes.com/2006/12/12/apple-microsoft-mac-tech-cz_dl_1212mac.html?partner=yahootix
>My #1 nomination for busy, bubbling gunk is the Dock, which mysteriously fills up with every single application that is ever installed on the machine. Then the icons get small (and often don't make much sense anyway) so it has to RESIZE itself when it's moused over. Because that makes a lot of sense, in the real world things often change size when I focus on them. And of course the Dock is always there, waiting to resize itself should you mouse over some part of an app that happens to be near it. The fast user switching is also great, when I change users it's REALLY important to me that it all happen like a big cube moved to a different side and the new user is there. That changes my life.

But my real point is that the operating system doesn't matter anymore, except to the people who sell them and a smaller number of people who are interested in them. What most people want from their computers (including me) is the ability to get information, find answers to questions, get their job done, and sometimes be entertained. Those tasks are no longer significantly related to what OS is run. For them one needs a web browser, a media player, some 'office' applications, an email (and maybe news/rss) reader, and a few extras. For me those extras are audio apps, a command line, and a development environment. For you they might be an image editor and a video editor.

Personally, for a window manager (which is what most people mean by an OS anyway) I like Gnome. It's simple, it runs quickly on almost any hardware. The 'stock' install on Debian includes a drop down menu for all the applications installed, a bar where I can drop quick launch icons for the five apps I use all the time, a shortcut to the root filesystem and my home directory, and a trash can.

When I need directions, I go to google maps no matter what OS I'm using. When I want to see a replay of the Michael Essien goal that took a point from Arsenal on Sunday I go to YouTube. When I want to watch the whole game I fire up Azereus and download two VCD ISOs via BitTorrent. If I forget the name of the guy who assassinated McKinley I go to wikipedia. If I'm going to write code I use Eclipse or Emacs. The first time in my day I'm restricted IN ANY WAY by the OS is when I start doing music creation and there aren't apps I find useful enough on linux so I'm stuck using one of the less attractive operating systems so I can use Live, SX, and so on. But in that I'm an exception. Most people could spend 100% of their computing lives using any OS if it had the basic apps I mentioned above.

People talk about whether there will be a 'Google platform' some day. There already IS a google platform, in the sense that if Apple switched their (really irritating) file browser for the (actually better) XP file browser it wouldn't really matter, but if you took away google maps (or MySpace, or Facebook, etc) it would screw up a LOT of people.

Now then, a remote control is a revolutionary way to get people in touch with information and entertainment? Remember what Jamie Zawinski said, if you want to know what software is going to be popular, think if it will help a 20 year old college student in his dorm room get laid. Google, MySpace, BitTornet and YouTube . . .

TCB

Jamie K <Meta@Dimensional.com> wrote:

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> My #1 nomination for busy, bubbling gunk is the Dock, which mysteriously fills
> up with every single application that is ever installed on the machine.

The dock is not mandatory, but I find it handy. Given your opinions, if
you actually used OSX you'd probably soon figure out how to make it more
useful and less intrusive for you (system prefs/dock).

I haven't noticed apps I install appearing there unless I put them
there, other than the apps that come pre-installed.

FYI you can simply drag off any that you don't want there.

You can drag those remaining into any order you like. You can also put
folders on the dock, and documents, anything you want handy.

> Then
> the icons get small (and often don't make much sense anyway) so it has to
> RESIZE itself when it's moused over. Because that makes a lot of sense, in
> the real world things often change size when I focus on them.

Heh. Just that one thing does.

You can specify whether dock items magnify or not. I prefer that they do because I've chosen to put a lot of stuff on the dock, and even on a 24" monitor (same as yours, I think) the icons are pretty small. The magnification works great. It's a useful, even clever approach IMO.

- > And of course
- > the Dock is always there, waiting to resize itself should you mouse over
- > some part of an app that happens to be near it.

So now you know you can turn that feature off.

- > The fast user switching is
- > also great, when I change users it's REALLY important to me that it all happen
- > like a big cube moved to a different side and the new user is there. That
- > changes my life.

Heh. It helps illustrate the user change for those who don't have the intrinsic understanding you do. Dunno if you can disable that, it's never annoyed me so I haven't looked.

- > But my real point is that the operating system doesn't matter anymore, except
- > to the people who sell them and a smaller number of people who are interested
- > in them. What most people want from their computers (including me) is the
- > ability to get information, find answers to questions, get their job done,
- > and sometimes be entertained. Those tasks are no longer significantly related
- > to what OS is run. For them one needs a web browser, a media player, some
- > 'office' applications, an email (and maybe news/rss) reader, and a few extras.

Yep, those can all be open source: TBird, Firefox, OpenOffice/NeoOffice, and VLC are a few I use on OSX.

- > For me those extras are audio apps, a command line, and a development environment.
- > For you they might be an image editor and a video editor.

Same as what you mentioned plus a variety of animation software.

- > Personally, for a window manager (which is what most people mean by an OS
- > anyway) I like Gnome. It's simple, it runs quickly on almost any hardware.
- > The 'stock' install on Debian includes a drop down menu for all the applications
- > installed, a bar where I can drop quick launch icons for the five apps I
- > use all the time, a shortcut to the root filesystem and my home directory,

> and a trash can.

You could, with training (grin), set up the OSX dock to do that.

> When I need directions, I go to google maps no matter what OS I'm using.
> When I want to see a replay of the Michael Essien goal that took a point
> from Arsenal on Sunday I go to YouTube. When I want to watch the whole game
> I fire up Azereus and download two VCD ISOs via BitTorrent. If I forget the
> name of the guy who assassinated McKinley I go to wikipedia. If I'm going
> to write code I use Eclipse or Emacs. The first time in my day I'm restricted
> IN ANY WAY by t

Subject: Re: Got the Paris stuff from John...

Posted by [duncan](#) on Mon, 13 Nov 2006 16:26:59 GMT

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hem, but only as files that are 2k big.

> The original files are on the order of tens of megabytes. And, does anyone
> know where I can find AudioX's "Auto Add" program? I can't locate audiox
> anywhere on the net.

> Thanks

> MR

>Or better yet, here's the link:

<http://www.lightlink.com/tjweber/StripWav/StripWav.html>

Jeff hoover wrote:

> Do you need to run the wavs through stripwave? Sometimes newer header
> versions give Paris fits. I've got a copy if you need it

>

> Hoov

>

> Mike R. wrote:

>

>> I have some wav files that were recorded in Live 5. I can't, however,

>> load

>> them into Paris. Well, it loads them, but only as files that are 2k big.

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>>What he said.

r
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 >>> anywhere on the net.
 >>> Thanks
 >>> MR
 >>>Thanks Jeff and Rod.
 Looks like Strip Wave was the way to go.
 MR

"Rod Lincoln" <rlincoln@nospam.kc.rr.com> wrote in message
 news:4581fa42\$1@linux...
 >
 > What he said.
 > r
 > Jeff hoover <jkhoover@excite.com> wrote:
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> >>>
> This is a multi-part message in MIME format.

-----=_NextPart_000_0017_01C71FD5.EFD72D20
Content-Type: text/plain;
charset="iso-8859-1"
Content-Transfer-Encoding: 7bit

Just fartin' around on the web and found this. Maybe helpful.

<http://www.fonerbooks.com/pcrepair.htm>

MR

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Content-Transfer-Encoding: quoted-printable

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</HEAD>
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this.&nbsp; Maybe helpful.</FONT></DIV>
<DIV><FONT face=3DArial size=3D2></FONT>&nbsp;</DIV>
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<DIV> </DIV>
<DIV>MR</DIV></BODY></HTML>

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charset="iso-8859-1"

Content-Transfer-Encoding: quoted-printable

I hadn't made any changes except the hardware wordclock to 48k
for this project. When it made it into the project just once, the =
latencies were
all off. There are a bunch of UADs in use. It had played well in the =
past.

So I switched to my cloned C drive and the latencies were off there
too. ? . Does the UAD need a setting changed?

I reloaded the UAD 4.3 drivers (original latency I've been using) and =
now=20

I have all these plugins at Paris boot that can't be found, most of =
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click=20

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After switching the clock back to 44.1 I tried other projects at 44.1 =
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This is across the board. =20

I rescanned them 3 times with FXpansion 3.3 but no help. I check no VST =
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I directed FXpansion
to the single folder that has all my VSTs.

Any suggestions?

Thanks because I am really confused now.

Tom

I choose Polesoft Lockspam to fight spam, and you?

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

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Subject: Re: Got the Paris stuff from John...
Posted by [John \[1\]](#) on Mon, 13 Nov 2006 20:12:59 GMT
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>"John" <no@no.com> wrote:

>

>I'm guessing Edmund is on that team! grrr Guess I'm gonna have to sell
>the freaking car to get Protools. Who would ever want to make music when
>you can dink around your whole life with crappy software instead.

>

>Can you tell I'm pissed ? Updates SOON.

This is why I never get version "point-zero" of ANYTHING. Or at
least I try to avoid it until I find out from other people if
it's relatively bug-free.

Sorry to hear this, John. Can you back out to v3.xxx? At least
then you'll be able to work until they release v4-point-bugfix.

NeilTotally useless, as I see no mention in any of the flowcharts
of "hammer" or "shotgun" as a final solution.

:D

"Mike R." <emarenot@yahoo.com> wrote:

>

>

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>

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> John

"Neil" <IUOIU@OIU.com> wrote:
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```

If you have a good rapport with your audio goods dealer you probably can... otherwise when v4-point-bugfix comes out, you just download it & go.

Neill've been thinking of a workaround that will create a situation where in some of the current from the UPS is tapped and sent to electrodes which are sent to 600 hamster cages, wherein upon charging the floors of these cages,

they must hop up on the wheel and start furiously running. These wheels will, in turn power small resistors that will reduce the current in the cage the faster they run so the faster they run, the more comfortable they are and the wheels will in turn 600 small power generators which will charge 12v backup batteries which can supplement the power of the UPS, effectively keeping the NG up until there are 600 exhausted and pissed off hamsters.

;O)

"Kim" <hiddensounds@hotmail.com> wrote in message news:45826a92\$1@linux...

>

> "Aaron Allen" <know-spam@not_here.dude> wrote:

>>Awesome man. Keep up the good work

>

>
