Subject: OK...how lucky is this?
Posted by DJ on Thu, 16 Nov 2006 22:27:45 GMT
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/>
><BODY bgColor=3D3D#ffffff>
><DIV><FONT face=3D3DArial size=3D3D2>So, do you have it there and you =
are =3D
>having=3D20
>problems with it? </FONT></DIV>
><DIV><FONT face=3D3DArial size=3D3D2></FONT> </DIV>
><DIV><FONT face=3D3DArial size=3D3D2>Deej</FONT></DIV>
><BLOCKQUOTE dir=3D3Dltr=3D20
>style=3D3D"PADDING-RIGHT: 0px; PADDING-LEFT: 5px; MARGIN-LEFT: 5px; =
=3D
>BORDER-LEFT: #000000 2px solid; MARGIN-RIGHT: 0px">
> <DIV>"Tom Bruhl" <<A=3D20
> href=3D3D"mailto:arpegio@comcast.net">arpegio@comcast.net</A>> =
wrote =3D
>in message=3D20
> <A href=3D3D"news:458cd310@linux">news:458cd310@linux</A>...</DIV>
> <DIV><FONT face=3D3DArial size=3D3D2>I got things really cooking =
here with
=3D
>my new rig=3D20
> except the</FONT></DIV>
> <DIV><FONT face=3D3DArial size=3D3D2>Creamware folks are on =
vacation =3D
>from today=3D20
> until 1/2/07. </FONT></DIV>
> <DIV><FONT face=3D3DArial size=3D3D2>I picked a great day to =
install the =3D
>Pro card . .=3D20
> .</FONT></DIV>
> <DIV><FONT face=3

Subject: Re: OK...how lucky is this?
Posted by dc[3] on Thu, 16 Nov 2006 23:50:08 GMT
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e=3D3D2>NOT!</FONT></DIV>
> <DIV><FONT size=3D3D2>

I choose Polesoft Lockspam to fight =
spam,
=3D
>and=3D20
> you?
<A=3D20
> =3D

=
>href=3D3D"http://www.polesoft.com/refer.html">http://www.polesoft.com/re=
fer=3D
>.html</A> </FONT></DIV></BLOCKQUOTE></BODY></HTML>
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>

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Content-Transfer-Encoding: quoted-printable

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charset=3Diso-8859-1">
<META content=3D"MSHTML 6.00.2800.1400" name=3DGENERATOR>
<STYLE></STYLE>
</HEAD>
<BODY bgColor=3D#ffffff>
<DIV><FONT face=3DArial size=3D2>Exactly. No =
allkeys.skf</FONT></DIV>
<DIV><FONT face=3DArial size=3D2></FONT> </DIV>
<DIV><FONT face=3DArial size=3D2>1/2/07 unless there's an automated =
</FONT><FONT=20
face=3DArial size=3D2>way to get them.</FONT></DIV>
<DIV><FONT face=3DArial size=3D2></FONT> </DIV>
<DIV><FONT face=3DArial size=3D2></FONT> </DIV>
<BLOCKQUOTE=20
style=3D"PADDING-RIGHT: 0px; PADDING-LEFT: 5px; MARGIN-LEFT: 5px; =
BORDER-LEFT: #000000 2px solid; MARGIN-RIGHT: 0px">
<DIV>"Neil" <<A href=3D"mailto:IOUOI@OIU.com">IOUOI@OIU.com</A>> =
wrote in=20
message <A=20
=
href=3D"news:458d4771$1@linux">news:458d4771$1@linux</A>...</DIV>
I'm =
sure=20
it's that he can't get the "allkeys" file.

Tom, did you e-mail=20
Ali? (<A =
href=3D"mailto:info@creamware.com">info@creamware.com</A>) &=20
give him your
card's SN#?

Neil

"DJ" <<A=20
href=3D"mailto:nowayjose@dude.net">nowayjose@dude.net</A>>=20
wrote:
>
>
>So, do you have it there and you are =
having=20
problems with it?=3D20
>
>Deej
> "Tom Bruhl" =
<<A=20
href=3D"mailto:arpegio@comcast.net">arpegio@comcast.net</A>> wrote =
in message=20

=3D
>news:458cd310@linux...
> I got

Subject: Re: OK...how lucky is this?
Posted by Mark McDermott on Fri, 17 Nov 2006 00:19:25 GMT
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0
0px">
> <DIV>"Tom Bruhl" =
<<A=3D20
> =20
href=3D3D"<A=20
=
href=3D'mailto:arpegio@comcast.net">arpegio@comcast.net</A'>mailto:arpegi=
o@comcast.net">arpegio@comcast.net</A</A>>>=20
wrote =3D
>in message=3D20
> <A href=3D3D"<A=20
=
href=3D'news:458cd310@linux">news:458cd310@linux</A>...</DIV'>news:458cd3=
10@linux">news:458cd310@linux</A>...</DIV</A>>
>&nbs=
p;=20
<DIV><FONT face=3D3DArial size=3D3D2>I got things really =
cooking here=20
with
=3D
>my new rig=3D20
> except=20
the</FONT></DIV>
> <DIV><FONT =
face=3D3DArial=20
size=3D3D2>Creamware folks are on vacation =3D
>from=20
today=3D20
> until 1/2/07. =
</FONT></DIV>
> =20
<DIV><FONT face=3D3DArial size=3D3D2>I picked a great day =
to install=20
the =3D
>Pro card . .=3D20
> =20
.</FONT></DIV>
> <DIV><FONT =
face=3D3DArial=20
size=3D3D2></FONT> </DIV>
> =
<DIV><FONT=20
face=3D3DArial size=3D3D2>Hurry up and=20
wait.</FONT></DIV>
> <DIV><FONT =
face=3D3DArial=20
size=3D3D2></FONT> </DIV>
> =
<DIV&a

Subject: Re: OK...how lucky is this?
Posted by DJ on Fri, 17 Nov 2006 04:48:10 GMT
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polesoft.com/refer.html"
target="_blank">http://www.polesoft.com/refer.html">http://www.polesoft.com/refer=

>'>http://www.polesoft.com/refer.html">http://www.polesoft.com/refer</A=
>>=3D
>.html</A>=20
> =
></FONT></DIV></BLOCKQUOTE></BODY></HTML><BR=
>>>
>
</BLOCKQUOTE></BODY></HTML>
>
>I was part of that thread (kdm) and did those tests - I actually took them
a step further than Jake or Fredo. As you can see I incorrectly thought
there was something in the group summing process, but it was just my boneheaded
interpretation of output data (using a small sample section for FFT rather
than the full file mainly). :-((

What Fredo is talking about is when you go over 0dBFS what happens to the
"over" data, and the references to truncation are in that case, which isn't
normal for mixing. This is the same decision every native DAW developer
has to make.

We were actually discussing what happens when you sum to a group vs. summing
to the main bus, without overs. I did my test with all files summing to
-20dB, so there was no chance of pushing the upper limits of 32-bit float's
truncation back down to 24-bits. And I actually simplified it by using two
copies of the same file (just as Fredo did), one phase inverted, both sample
aligned. They cancelled to below 24 bits just as expected, and just as they
should. The variations below 24 bits that I saw (and thought were above
24-bits at one point) are correlation of lower frequencies when gain and
equivalent reduction are introduced (which is what Chuck stated that Paris
does up front on every track). That really doesn't impact the audio itself
since data below -136dB is quantization noise for 24-bit audio.

Sonar, Nuendo, Cubase 4 and Sequoia all behaved exactly the same way in this
test - which tells me they are handling the LSB's the same way. When data
is summed to groups, there will be quantization noise below -136dB. This
is completely normal for any native DAW and they all are subject to it.
As you might read in the thread my conclusion was that we proved digital
audio theory exists - e.g. no uncharted territory, no digital audio frontiers,
no bugs in Nuendo. yeeha. But that's what I get for second guessing talented
developers. ;-)

Fwiw, to take it a step further, Samplitude/Sequoia and Nuendo handle overs,
or "into the red" identically. I checked that too a while back after the
reports of extra headroom, etc in Samplitude. Believe me, I've tried hard
to find where any differences might appear, not just noticeable differences,
but any differences at the lowest levels, but it seems the major native DAW
players are making the same decisions when it comes to truncation, etc, and
there really aren't that many to make. In my tests, dither really wasn't
an issue (I turned it off in all DAWs I tested just to test with pure truncation).

Regards,

Dedric

"LaMOnt" <jjdpro@ameritech.net> wrote:
>
>Dedric, check out this post from our dear friend Fredo: Neundo Moderator:
>Explaining how Steingberg's audio engine works. Note the trade-offs..Meaning,
>Steinberg's way of coding an audio-engine 32bit float is different than
say
>Magix Samplitude:
>
>Fredo
>Administrative Moderator
>
>
>Joined: 29 Dec 2004
>Posts: 4213
>Location: Belgium
> Posted: Fri Dec 08, 2006 2:33 pm Post subject:
>
> ---------- ----------
>
>I think I see where the problem is.
>In my scenario's I don't have any track that goes over 0dBfs, but I have
>always lowered one channel to compensate with another.
>So, I never whent over the 0dB fs limit.
>
>Here's th

Subject: Re: OK...how lucky is this?
Posted by Neil on Fri, 17 Nov 2006 05:33:36 GMT
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Subject: Re: OK...how lucky is this?
Posted by rick on Fri, 17 Nov 2006 09:44:59 GMT
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s you boost one channel over 0dBfs. (or merge two files
>that are @ 0dB)
>Also, this problem will be pretty much gone as soon as we switch to the
64
>bit engine.
>

>
>For the record, the test where Jake hears "music" as residue must be flawed.
>You should hear noise/distortion from square waves.
>
>HTH
>
>Fredo
>
>
>
>
>
>"Dedric Terry" <dedric@echomg.com> wrote:
>>I can't tell you why you hear ProTools differently than Nuendo using a

>>single file.
>>There isn't any voodoo in the software, or hidden character enhancing dsp.
>
>>I'll see if
>>I can round up an M-Powered system to compare with next month.
>>
>>For reference, everytime I open Sequoia I think I might hear a broader,
>
>>clean,
>>and almost flat (spectrum, not depth) sound, but I don't - it's the same
>as
>>Nuendo, fwiw.
>>Also I don't think what I was referring to was a theory from Chuck - I
>
>>believe that was what he
>>discovered in the code.
>>
>>Digital mixers all have different preamps and converters. Unless you are
>
>>bypassing every
>>EQ and converter and going digital in and out to the same converter when
>
>>comparing, it would be hard
>>to say the mix engine itself sounds different than another mixer, but taken
>
>>as a whole, then
>>certainly they may very well sound different. In addition, hardware digital
>>mixers may use a variety of different paths between the I/O, channel
>>processing, and summing,
>>though most are pretty much software mixers on a single chip or set of
dsps
>
>>similar to ProTools,

>>with I/O and a hardware surface attached.
>>

Subject: Re: OK...how lucky is this?
Posted by IOUOI on Fri, 17 Nov 2006 14:39:04 GMT
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e are only slight visual differences (some workflow differences
>
>>are a definite improvement for me though).
>>
>>ProTools' GUI always made me want to write one dimensional soundtracks
in
>
>>mono for public utilities, accounting offices
>>or the IRS while reading my discreet systems analysis textbook - it was
>also
>>grey. ;-)
>>
>>Regards,
>>Dedric
>>
>>"LaMont" <

Subject: Re: OK...how lucky is this?
Posted by TCB on Fri, 17 Nov 2006 15:32:37 GMT
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per Chuck's
>>>
>>>>findings - so correct me if I'm missing part of the equation Chuck),
>>>>is drop the track gain by 20dB or so, then added it back at the master
>
>>>>buss
>>>
>>>>to create the effect of headroom (probably
>>>>because the master buss is really summing on the card, and they have
more
>>>
>>>>headroom there than on the tracks
>>>>where native plugins might be used). I don't know if Paris passed 32-bit
>>>
>>>>float files to the EDS card, but sort of
>>>>doubt it. I think Chuck has clarified this at one point, but don't recall
>>>

>>>>the answer.
>>>>
>>>>Also what Paris did is use a greater bit depth on the hardware than
>>>>ProTools
>>>
>>>>did - at the time PT was just
>>>>bring Mix+ systems to market, or they had been out for a year or two
(if
>>> I
>>>>have my timeline right) - they
>>>>were 24-bit fixed all the way through. Logic and Cubase were native
DAWs,
>>>
>>>>but native was still too slow
>>>>to compete with hardware hybrids. Paris trumped them all by running

>>>>32-bit
>>>
>>>>float natively (not new really, but
>>>>better than sticking to 24-bit) and 56 or so bits in hardware instead
>of
>>>
>>>>going to Motorola DSPs at 24.
>>>>The onboard effects were also a step up from anything out there, so the
>>> demo
>>>>did sound good.
>>>>I don't recall which, but one of the demos, imho, wasn't so good (some
>>>>sloppy production and
>>>>vocals in spots, IIRC), so I only listened to it once. ;-)
>>>>
>>>>Coupled with the gain drop and buss makeup, this all gave it a "headroom"

Subject: Re: OK...how lucky is this?
Posted by rick on Fri, 17 Nov 2006 15:59:02 GMT
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f the 48-bit fixed
>>> vs.
>>>>>
>>>>>>32-bit float wasn't a factor.
>>>>>>
>>>>>>When trying other tests, even when adding and subtracting gain, Nuendo,
>>>>>
>>>>>>Sequoia and Sonar cancel - both audibly and
>>>>>>visually at inaudible levels, which only proves that one isn't making
>>> an
>>>>>

>>>>>>error when calculating basic gain. Since a dB is well defined,
>>>>>>and the math to add gain is simple, they shouldn't. The fact that
they
>>>>> all
>>>>>>use 32-bit float all the way through eliminates a difference
>>>&gt

Subject: Re: OK...how lucky is this?
Posted by DJ on Fri, 17 Nov 2006 16:40:41 GMT
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 has been proven) that summing isn't different
>>>>>
>>>>>>unless
>>>>>>there is an error somewhere, or variation in how the user duplicates
>the
>>>>>
>>>>>>same mix in two different apps.
>>>>>>
>>>>>>Imho, that's actually a very good thing - approaching a more consistent
>>>>>
>>>>>>basis for recording and mixing from which users can make all
>>>>>>of the decisions as to how the final product will sound and not be
>>>>>>required
>>>>>
>>>>>>to decide when purchasing a pricey console, and have to
>>>>>>focus their business on clients who want "that sound". I believe we
>are
>>>>>
>>>>>>actually closer to the pure definition of recording now than
>>>>>>we once were.
>>>>>>
>>>>>>Regards,
>>>>>>Dedric
>>>>>>
>>>>>>
>>>>>>>
>>>>>>> I the answer is yes, then,the real task is to discover or rather
>>>>>>> un-cover
>>>>>>> what's say: Motu's vision of summing, versus Digidesign, versus
>>>>>>> Steinberg
>>>>>>> and so on..
>>>>>>>
>>>>>>> What's under the hood. To me and others,when Digi re-coded their

>>>>>>> summing
>>>>>>> engine, it was obvious that Pro Tools has an obvious top end (8k-10k)

>>>>>
>>>>>>> bump.
>>>>>>> Where as Steinberg's summing is very neutral.
>>>>>>>
>>>>>>> "Dedric Terry" <dedric@echomg.com> wrote:
>>>>>>>>Hi Neil,
>>>>>>>>
>>>>>>>>Jamie is right. And you aren't wacked out - you are thinking this
>>>>>>>>through
>>>>>>>
>>>>>>>>in a reasonable manner, but coming to the wrong
>>>>>>>>conclusion - easy to do given how confusing digital audio can be.
>
>>>>>>>>Each
>>>>>>>> word
>>>>>>>>represents an amplitude
>>>>>>>>point on a single curve that is changing over time, and can vary
with
>>>>> a
>>>>>>>
>>>>>>>>speed up to the Nyquist frequency (as Jamie described).
>>>>>>>>The complex harmonic content we hear is actually the frequency
>>&g

Subject: Re: OK...how lucky is this?
Posted by DJ on Fri, 17 Nov 2006 16:41:37 GMT
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t;>>>>modulation
>>>>>>> of
>>>>>>>>a single waveform,
>>>>>>>>that over a small amount of time creates the sound we translate -
>we
>>>
>>>>>>>>don't
>>>>>>>
>>>>>>>>really hear a single sample at a time,
>>>>>>>>but thousands of samples at a time (1 sample alone could at most
>>>>>>>>represent
>>>>>>> a
>>>>>>>>single positive or negative peak
>>>>>>>>of a 22,050Hz waveform).
>>>>>>>>
>>>>>>>>If one bit doesn't cancel, esp. if it's a higher order bit than number
>>>>> 24,
>>>>>>>
>>>>>>>>you may hear, and will see that easily,

>>>>>>>>and the higher the bit in the dynamic range (higher order) the more
>>>>>>>>audible
>>>>>>>
>>>>>>>>the difference.
>>>>>>>>Since each bit is 6dB of dynamic range, you can extrapolate how "loud"
>>>>>
>>>>>>>>that
>>>>>>>
>>>>>>>>bit's impact will be
>>>>>>>>if there is a variation.
>>>>>>>>
>>>>>>>>Now, obviously if we are talking about 1 sample in a 44.1k rate song,
>>>>> then
>>>>>>>
>>>>>>>>it simply be a
>>>>>>>>click (only audible if it's a high enough order bit) instead of an
>>>>>>>>obvious
>>>>>>>
>>>>>>>>musical difference, but that should never
>>>>>>>>happen in a phase cancellation test between identical files higher
>
>>>>>>>>than
>>>>>>> bit
>>>>>>>>24, unless there are clock sync problems,
>>>>>>>>driver issues, or the DAW is an early alpha version. :-)
>>>>>>>>
>>>>>>>>By definition of what DAWs do during playback and record, every audio
>>>>>
>>>>>>>>stream
>>>>>>>
>>>>>>>>has the same point in time (judged by the timeline)
>>>>>>>>played back sample accurately, one word at a time, at whatever sample
>>>>>
>>>>>>>>rate
>>>>>>>
>>>>>>>>we are using. A phase cancellation test uses that
>>>>>>>>fact to compare two audio files word for word (and hence bit for
bit
>>>
>>>>>>>>since
>>>>>>>
>>>>>>>>each bit of a 24-bit word would
>>>>>>>>be at the same bit slot in each 24-bit word). Assuming they are

>>>>>>>>aligned
>>>>>>> to
>>>>>>>>the same start point, sample
>>>>>>>>accurately, and both are the same set of sample words at each sample

>>>>>>>>point,
>>>>>>>
>>>>>>>>bit for bit, and one is phase inverted,
>>>>>>

Subject: Re: OK...how lucky is this?
Posted by DJ on Fri, 17 Nov 2006 16:43:10 GMT
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>>they will cancel through all 24 bits. For two files to cancel
>>>>>>>>completely
>>>>>>>
>>>>>>>>for the duration of the file, each and every bit in each word
>>>>>>>>must be the exact opposite of that same bit position in a word at
>the
>>>>> same
>>>>>>>
>>>>>>>>sample point. This is why zooming in on an FFT
>>>>>>>>of the full difference file is valuable as it can show any differences
>>>>> in
>>>>>>>
>>>>>>>>the lower order bits that wouldn't be audible. So even if
>>>>>>>>there is no audible difference, the visual followup will show if
the
>>> two
>>>>>>>
>>>>>>>>files truly cancel even a levels below hearing, or
>>>>>>>>outside of a frequency change that we will perceive.
>>>>>>>>
>>>>>>>>When they don't cancel, usually there will be way more than 1 bit
>>>>>>>>difference - it's usually one or

Subject: Re: OK...how lucky is this?
Posted by Jamie K on Fri, 17 Nov 2006 17:20:04 GMT
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more bits in the words for
>>>>>>>>thousands of samples. From a musical standpoint this is usually
in
>>> a
>>>>>>>>frequency range (low freq, or high freq most often) - that will
>>>>>>>>show up as the difference between them, and that usually happens
due
>>> to
>>>>>>> some

>>>>>>>>form of processing difference between the files,
>>>>>>>>such as EQ, compression, frequency dependant gain changes, etc. That
>>> is
>>>>>>> what
>>>>>>>>I believe you are thinking through, but when
>>>>>>>>talking about straight summing with no gain change (or known equal
>
>>>>>>>>gain
>>>>>>>
>>>>>>>>changes), we are only looking at linear, one for one
>>>>>>>>comparisons between the two files' frequency representations.
>>>>>>>>
>>>>>>>>Regards,
>>>>>>>>Dedric
>>>>>>>>
>>>>>>>>> Neil wrote:
>>>>>>>>> "Dedric Terry" <dedric@echomg.com> wrote:
>>>>>>>>>> The tests I did were completely blank down to -200 dB (far below
>>> the
>>>>>>>
>>>>>>>>>> last
>>>>>>>>>
>>>>>>>>>> bit). It's safe to say there is no difference, even in
>>>>>>>>>> quantization noise, which by technical rights, is considered
below
>>>>> the
>>>>>>>
>>>>>>>>>> level
>>>>>>>>>
>>>>>>>>>> of "cancellation" in such tests.
>>>>>>>>>
>>>>>>>>> I'm not necessarily talking about just the first bit or the
>>>>>>>>> last bit, but also everything in between... what happens on bit
>>>>>>>>> #12, for example? Everything on bit #12 should be audible, but
>>>>>>>>> in an a/b test what if thre are differences in what bits #8
>>>>>>>>> through #12 sound like, but the amplutide is stll the same on
>>>>>>>>> both files at that point, you'll get a null, right? Extrapolate
>>>>>>>>> that out somewhat & let's say there are differences in bits #8
>>>>>>>>> through #12 on sample points 3, 17, 1,000, 4,523, 7,560, etc,
>>>>>>>>> etc through 43,972... Now this is breaking things down well
>>>>>>>>> beyond what I think can be measured, if I'm not mistaken (I
>>>>>>>>> dn't know of any way we could extract JUST that information
>>>>>>>>> from each file & play it back for an a/b test; but would not
>>>>>>>>> that be enough to have to "null-able" files that do actually
>>>>>>>>> sound somewhat different?
>>>>>>>>>
>>>>>>>>> I guess what I'm saying is that since each sample in a musical
>>>>>>>>> track or full song file doesn't represent a pure, simple set of

>>>>>>>>>> content like a sample of a sine wave would - there's a whole
>>>>>>>>>> world of harmonic structure in each sample of a song file, and
>>>>>>>>>> I think (although I'll admit - I can't "prove") that there is
>>>>>>>>>> plenty of room for some variables between the first bit & the
>>>>>>>>>> last bit while still allowing for a null test to be successful.
>>>>>>>>>>
>>>>>>>>>> No? Am I wacked out of my mind?
>>>>>>&g

Subject: Re: OK...how lucky is this?
Posted by DC on Fri, 17 Nov 2006 23:46:42 GMT
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;news:458cd310@linux</A>...</DIV</A>>
>&nbs=
>>p;=20
>> <DIV><FONT face=3D3DArial size=3D3D2>I got things really =
>>cooking here=20
>> with
=3D
>my new rig=3D20
> except=20
>> the</FONT></DIV>
> <DIV><FONT =
>>face=3D3DArial=20
>> size=3D3D2>Creamware folks are on vacation =3D
>from=20
>> today=3D20
> until 1/2/07. =
>></FONT></DIV>
> =20
>> <DIV><FONT face=3D3DArial size=3D3D2>I picked

Subject: Re: OK...how lucky is this?
Posted by Paul Braun on Sat, 18 Nov 2006 00:26:19 GMT
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long enough, this will do the job.
>
>voice of experience

No, Deej, I'm ging to send the audio out through Lightpipe - 4
channels per Multiface (2 stereo submixes out of each
Multiface's lightpipe "out", into each of the two
lightpipe "in's" on the Pulsar card).

Rat Shack's here are carrying Monster digilight cables now,
too... profit margin must be just a LITTLE bit bigger on a $30
cable than on a $5 one :D may as well go to Guitar Center
& see if they have anything else at a lower price point (plus,
check & see if they got anything new in stock that my gear
sluttery just can't resist LOL!)

NeilThat is really funny!! Thanks Gene. Merry christmas.

;o)

"Gene Lennon" <glennon@NOSPmyrealbox.com> wrote in message
news:458

Subject: Re: OK...how lucky is this?
Posted by rick on Sat, 18 Nov 2006 09:36:26 GMT
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>>>> Dedric good post..
>>>>>
>>>>> However, I have PT-M-Powered/M-audio 410 interface for my laptop and
>it
>>>
>>>>> has
>>>>> that same sound (no eq, zero fader) that HD does. I know their use
the
>>>
>>>>> same
>>>>> 48 bit fix mixer. I load up the same file in Nuendo (no eq, zero
>>>>> fader)..results.
>>>>> different sonic character.
>>>>>
>>>>> PT having a top end touch..Nuendo, nice smooth(flat) sound. And I'm
>just
>>>>> taking about a stereo wav file nulled with no eq..nothing
>>>>> ..zilch..nada..
>>>>>
>>>>> Now, there are devices (keyboards, dum machines) on the market today
>
>>>>> that
>>>>> have a Master Buss Compressor and EQ set to on with the top end notched
>>>
>>>>> up.
>>>>> Why? because it gives their product an competitive advantageover the
>>>>> competition..
>>>>> Ex: Yahama's Motif ES, A

Subject: Re: OK...how lucky is this?
Posted by DJ on Sat, 18 Nov 2006 15:44:55 GMT
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gt;>

>>>>>>Samplitude/Sequoia, DP, Audition (I presume at least).
>>>>>>I don't know what Acid or Live use. Saw promotes a fixed point engine,
>>>>> but
>>>>>>I don't know if it is still 24-bit, or now 48 bit.
>>>>>>That was an intentional choice by the developer, but he's the only
one
>>> I
>>>>>
>>>>>>know of that stuck with 24-bit for summing
>>>>>>intentionally, esp. after the Digi Mix system mixer incident.
>>>>>>
>>>>>>Long answer, but to sum up, it is certainly physically *possible* for
>>> a
>>>>>
>>>>>>developer to code something differently intentionally, but not
>>>>>>in reality likely since it would be breaking some basic fixed point
>or
>>>>>>floating point math rules. Where the differences really
>>>>>>showed up in the past is with PT Mix systems where the limitation was
>>>
>>>>>>really
>>>>>
>>>>>>significant - e.g. 24 bit with truncation at several stages.
>>>>>>
>>>>>>That really isn't such an issue anymore. Given the differences in
>>>>>>workflow,
>>>>>
>>>>>>missing something in workflow or layout differences
>>>>>>is easy enough to do (e.g. Sonar doesn't have group and busses the
way
>>>>>>Nuendo does, as it's outputs are actually driver outputs,
>>>>>>not software busses, so in Sonar, busses are actually outpu

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Subject: Re: OK...how lucky is this?
Posted by rick on Sun, 19 Nov 2006 10:18:37 GMT
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will probaly will get the sell. That's
>>>> what
>>>>>> happend to me back in 1997 when I heard Paris. I was floored!!! Still
>>>> to
>>>>>> this day, nothing has floored me like that "Road House Blues Demo"
>I
>>
>>>>>> heard
>>>>>> on Paris.
>>>>>>

>>>>>> Was it the hardware ? was it the software. I remember talking with
>
>>>>>> Edmund
>>>>>> at the 2000 winter Namm, and told me that he & Steve set out to
>>>>>> reproduce
>>>>>> the sonics of big buck analog board (eq's) and all.. And, summing
was
>>>> a
>>>>>> big
>>>>>> big issue for them because they (ID) thought that nobody has gotten
>>>>>> it(summing)
>>>>>> right. And by right, they meant, behaved like a console with a wide
>>lane
>>>>>> for all of those tracks..
>>>>>>
>>>>>>
>>>>>>
>>>>>>
>>>>>> "Dedric Terry" <dedric@echomg.com> wrote:
>>>>>>>"LaMont" <jjdpro@ameritech.net> wrote in message
>>>>>>>news:458be8d5$1@linux...
>>>>>>>>
>>>>>>>> Okay...
>>>>>>>> I guess what I'm saying is this:
>>>>>>>>
>>>>>>>> -Is it possible that diferent DAW manufactuers "code" their app
>>>>>>>> differently
>>>>>>>> for sound results.
>>>>>>>
>>>>>>>Of course it is *possible* to do this, but only if