Subject: cubase mixing levels
Posted by John [1] on Thu, 26 Oct 2006 00:06:39 GMT
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If you are recording @ 24 bit you really don't need to get that high.
Peaks of -25 to -15 are more than enough. Terry Manning of Compass Point Studios (AC/DC, ZZ Top etc) turned me on to this on a different forum and is a big advocate of it. I tried it. I have to agree with him that it made a significant improvement in the resulting sound of the recording.

http://www.cubase.net/phpbb2/viewtopic.php?t=55258&highl ight=clipping

Subject: Re: cubase mixing levels
Posted by John [1] on Thu, 26 Oct 2006 00:09:01 GMT
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also on that thread,

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

John wrote:
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Subject: Re: cubase mixing levels
Posted by Martin Harrington on Fri, 27 Oct 2006 23:20:20 GMT
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Otherwise you are not using all the bits available to you, and noise will be the end result.
This is why the good/great engineers are what they are...they make sure the levels are hot....just not to the stage of distortion.
it's a balancing act, but, hey, who said anything done properly is easy.

--
Martin Harrington
www.lendanear-sound.com

"John" <no@no.com> wrote in message news:453ff9d7@linux...
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Subject: Re: cubase mixing levels
Posted by John [1] on Sat, 28 Oct 2006 00:15:27 GMT
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Martin, so do you know anything about the Cubase mix bus? Do they maybe
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Just wondering how the Cubase mix bus behaves.

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Subject: Re: cubase mixing levels
Posted by Martin Harrington on Sat, 28 Oct 2006 03:16:04 GMT
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Subject: Re: cubase mixing levels
Posted by AlexPlasko on Sat, 28 Oct 2006 04:03:50 GMT
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"Martin Harrington" <lendan@bigpond.net.au> wrote in message news:4542c966$1@linux...
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Subject: Re: cubase mixing levels
Posted by Dedric Terry on Sat, 28 Oct 2006 05:04:36 GMT
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I agree with Martin completely. To add my own general opinion on this long
running "sound of summing" debate, that is fast become urban legend:

I think it is time to start debunking some of the fables around digital
summing. Paris cutting levels and then adding the gain back at the master
does one and only one thing: pushes all tracks down by 22dB to make it
easier to sum them well below 0dBFS, and then make some of it back before
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any native DAW, you will in effect do the same thing - lower track levels
such that the peak of the summation remains below 0dBFS.

In simple math terms:

1- Native DAW: 2+2=4
2- Paris: ((2-1)+(2-1)) + 2 = 4

With SX you adjust the gain yourself on the way in, if you like, or better yet, set the levels for the mix at hand as needed, as you go.

So why do this in Paris? I am guessing Paris had to convert all audio to 24-bit if sent to the native cpu for native plugins in order to prevent clipping before you even start dropping faders on the mix (unless they somhow converted to 32-bit float on the EDS chip first, which isn't a 32-bit float chip, so that seems unlikely). (No one would want to mix with most faders at -40dB - it would "seem" wrong). 24-bits for any portion of summing (not tracking) is a limitation esp. if tracked audio files are near 0dBFS to begin with - there is no where to go with gain addition, and only subtraction to work with. You can add a lot more -22dB peak audio files to a mix without clipping (with all faders at 0), where only 2 audio files peaking just below 0dBFS will automatically clip. At least that seems to be the reasoning behind it, but imho, only applicable to prevent a potential problem in the DAW itself, not as prescribed digital audio practice.

Back to native DAWs: While this approach may seem somehow capable of producing a different sound to a final mix, the only think you gain by lowering tracks by 22dB from the start is lower bit resolution. This is especially true if you happen to record your tracks at lower levels (e.g. -10dB peaks, then your peaks are now at -32dB, which is 1/3 the resolution of 16 bit audio - that isn't insignificant). So, if you combine the flawed advice to record at -10 to -20dB, with the concept of lowering all tracks by 22dB before you start mixing, and you end up removing most of the resolution we work hard for with quality mics, preamps and converters, and mixing as much more noise and quantization error than necessary.

As Martin suggested, record just below, or comfortably below clipping (as the source dictates) to maximize your use of bit resolution - e.g. keep audible that ambience or depth of the recording that tails off down at those lower levels, rather than mixing it with mic self-noise, etc.

John, what I think the reference you quoted is actually saying (or should be at least) is just that 24-bit digital (with a quality front end and converters) affords a higher signal to noise ratio than analog did, so pushing record levels to widen the gap between peaks and the noise floor isn't as critical. But since quite a few really great mics have a noise floor of around -70 to -74dB, we are still putting noise into half of the bits of that glorious 24-bit range.

In terms of signal vs. noise - why record less signal when you can record more?

Imho, there are a lot of "famous" engineers out there spouting complete technical rubbish out of lack of true knowledge, or passing along conversational heresay and conjecture. Sadly, engineering is becoming more about letting the gear dictate the recording process than the engineer, and I believe that's what's wrong with music today. Too my people think xyz piece of gear can make a hit, a vibe, or a certain sound just because someone else did it, but few actually use their skills to create the sound by knowing what combinations of gear will help get them there.

9/10 times it isn't about obtaining one piece of gear to get "that sound", but understanding the 1000 different possibilities and knowing how to use any of them.

The Nuendo and SX audio engines are identical. They also are identical in summing to Sequoia/Samplitude, and probably Logic, Audition, and DP too (I've tested Nuendo and Sequoia beyond the limits of normal recording to verify this, so this comes from experience and listening, not speculation or internet urban legend).

Regards,
Dedric

On 10/27/06 10:03 PM, in article 4542d488@linux, "alex plasko" <alex.plasko@snet.net> wrote:

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Subject: Re: cubase mixing levels
Posted by Ted Gerber on Sat, 28 Oct 2006 09:51:56 GMT
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HI Dedric

Thank you for your excellent posts and the time taken on this topic. I
appreciate it.

As far as specific math goes, there is one statement from the link John
referenced that contradicts your statement regarding resolution at
various levels

Your quote here:

" if you happen to record your tracks at lower levels (e.g.
>-10dB peaks, then your peaks are now at -32dB, which is 1/3 the resolution
>of 16 bit audio - that isn't insignificant). "

The other statement said that 24bit recording at -48db is equal to a full
range
16bit recording...

These 2 statements look like they're talking about the same thing (apples
to apples). If they are, how are they reconciled? If they're not and I'm
misunderstanding...

Ted

Dedric Terry <dterry@keyofd.net> wrote:
>I agree with Martin completely. To add my own general opinion on this long
>running "sound of summing" debate, that is fast become urban legend:
>
>I think it is time to start debunking some of the fables around digital
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Posted by John [1] on Sat, 28 Oct 2006 10:25:09 GMT
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great post! thanks

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>>> I record as hot as I can, and use the channel faders to mix, usually never
>>> moving the master fader, although having said that, my mixes for TV/ Doco
>>> work are not quite as complicated as most decent size music mixes would
>>> be.
>>> --
>>> Martin Harrington
>>> www.lendanear-sound.com
>>> "John" <no@no.com> wrote in message news:45429eda@linux...
>>>> Martin, so do you know anything about the Cubase mix bus? Do they maybe
>>>> mean that on mixdown you pull the faders way back but still record hot?
>>>> Just wondering how the Cubase mix bus behaves.
>>>>
>>>> Thanks
>>>>
>>>> Martin Harrington wrote:
>>>>> That's not true, and dont let anyone tell you it is.
>>>>> You still need to get all levels as optimised as possible, as we all did
>>>>> with tape.
>>>>> Otherwise you are not using all the bits available to you, and noise
>>>>> will be the end result.
>>>>> This is why the good/great engineers are what they are...they make sure
>>>>> the levels are hot....just not to the stage of distortion.
>>>>> it's a balancing act, but, hey, who said anything done properly is easy.
>>>>>
>>>
>>
>

Subject: Re: cubase mixing levels
Posted by Neil on Sat, 28 Oct 2006 14:47:44 GMT
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Dedric Terry <dterry@keyofd.net> wrote:

>I agree with Martin completely. To add my own general opinion
>on this long running "sound of summing" debate,

Great post, Dedric! BTW, "The Sound of Summing" - wasn't that a
song by Summon & Nullfunkel?

Anyway, a couple of things I wanted to bring up. You said:

>In simple math terms:
>1- Native DAW: 2+2=4
>2- Paris: ((2-1)+(2-1)) + 2 = 4
>With SX you adjust the gain yourself on the way in, if you like, or better

>yet, set the levels for the mix at hand as needed, as you go.

I just wanted to point out that if anyone thought I meant to
adjust your levels "on the way in" (i.e.: using the Channel
input trim control), when I had said to default all your
channels to -6, for example, as a starting point - that I did
NOT, in fact, mean the input trims!!! I meant the regular
ol' "please make it louder or softer" control. :D

>So why do this in Paris? I am guessing Paris had to convert all audio to
>24-bit if sent to the native cpu for native plugins in order to prevent
>clipping before you even start dropping faders on the mix (unless they
>somhow converted to 32-bit float on the EDS chip first, which isn't a 32-bit
>float chip, so that seems unlikely). (No one would want to mix with most
>faders at -40dB - it would "seem" wrong). 24-bits for any portion of
>summing (not tracking) is a limitation esp. if tracked audio files are near
>0dBFS to begin with - there is no where to go with gain addition, and only
>subtraction to work with. You can add a lot more -22dB peak audio files
to
>a mix without clipping (with all faders at 0), where only 2 audio files
>peaking just below 0dBFS will automatically clip. At least that seems to
be
>the reasoning behind it, but imho, only applicable to prevent a potential
>problem in the DAW itself, not as prescribed digital audio practice.
>
>Back to native DAWs: While this approach may seem somehow capable of
>producing a different sound to a final mix, the only think you gain by
>lowering tracks by 22dB from the start is lower bit resolution. This is
>especially true if you happen to record your tracks at lower levels (e.g.
>-10dB peaks, then your peaks are now at -32dB, which is 1/3 the resolution
>of 16 bit audio - that isn't insignificant). So, if you combine the flawed
>advice to record at -10 to -20dB, with the concept of lowering all tracks
by
>22dB before you start mixing, and you end up removing most of the resolution
>we work hard for with quality mics, preamps and converters, and mixing as
>much more noise and quantization error than necessary

Which is why I suggest starting at -6... maybe -10 if you're
going to be loading up a buttload of tracks. Think about it
with an analog analogy again, gang: If your analog mixing
console goes up to +10 on each channel, would you start a mix
with every fucking fader maxed out at +10??? Hit "play", and
how would THAT summing buss sound right about then? No, to
start out with, you'd bring all the faders down to 0, or -5,
or -10, or whatever the your comfortable starting point was,
knowing the console & how much headroom it's got, etc., right?
Pushing everything up to +10 to start off with would be just

simply way too much, yes?

So why are people not willing to get their heads around the
fact that in digital anything past "0" is "way too much"?
The only reason your DAW has +6 or +8 on any given channel is
in case you fuck up & record the cowbell track on your band's
cover version of "Mississippi Queen" at peak levels of -17db
.... "0" gain on the channel level just wouldn't cut it at that
stage, the cowbell just wouldn't be audible enough to drive
that tune. :D

So, just like in the analog console, where 40+ channels
recorded to needle-bending levels on overbiased tape machines,
and every channel set to +10 at the start of your mixdown
session would sound like crap; so does 40+ channels recorded to
nice hot levels, barely missing overs by a tenth of a db,
with every channel set to "0" result in a similar thing.

Neil

Subject: Re: cubase mixing levels
Posted by Dedric Terry on Sat, 28 Oct 2006 15:30:54 GMT
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Hi Ted,

16 bit and 24 bit both only represent up to 0dB full scale (FS). The
dynamic range afforded by 24-bit extends down to -144dB rather than the
-96dB of 16 bit. That's what we are interested in with digital audio, not
the theoretical limits above 0.

Once in the DAW, -48dB is represented by the lower 8 bits in 16-bit words,
and the lower 16 bits in 24-bit words. That's probably how they came to
that conclusion, but it's mathematically incorrect since the same bit
actually represents -48dB, you just add an extra 8 onto the bottom of the
dynamic range, not the top as the quote seems to assume.

Regards,
Dedric

On 10/28/06 3:51 AM, in article 4543283c$1@linux, "Ted Gerber"
<tedgerber@rogers.com> wrote:

> Your quote here:
>
> " if you happen to record your tracks at lower levels (e.g.
>> -10dB peaks, then your peaks are now at -32dB, which is 1/3 the resolution

>> of 16 bit audio - that isn't insignificant). "
>
> The other statement said that 24bit recording at -48db is equal to a full
> range
> 16bit recording...
>
> These 2 statements look like they're talking about the same thing (apples
> to apples). If they are, how are they reconciled? If they're not and I'm
> misunderstanding...
>
> Ted

Subject: Re: cubase mixing levels
Posted by audioguy_editout_ on Sat, 28 Oct 2006 16:43:09 GMT
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Ah, now, this is where people get confused.

The assumption that FULL bits (all 1111's) represents 0dbfs,
and that empty bits (all 00000's) represents the noise floor
is false.

The full bits actually represent the maximum positive
amplitude, and the empty bits represent the maximum
negative amplitude of the bi-phase audio signal. (I have
attached a pic below of a 4 bit signal capture, the vertical
axis shows the 4 bits, while the horizontal axis shows the
sample rate)

So, what is actually happening in 24 bit vs 16 bit is that
there are more bits to represent the vertical axis. This
means that "no signal" on the input of the recorder would
yield a number half way up, not all zeros.

Does this help?

David.

Dedric Terry wrote:

> Hi Ted,
>

> 16 bit and 24 bit both only represent up to 0dB full scale (FS). The
> dynamic range afforded by 24-bit extends down to -144dB rather than the
> -96dB of 16 bit. That's what we are interested in with digital audio, not
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>
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>
> Regards,
> Dedric
>
> On 10/28/06 3:51 AM, in article 4543283c$1@linux, "Ted Gerber"
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>
>
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>>
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>>misunderstanding...
>>
>>Ted
>
>

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Subject: Re: cubase mixing levels
Posted by audioguy_editout_ on Sat, 28 Oct 2006 17:16:12 GMT
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Yah, I was refering to the sampling phase of straight PCM
encoding. I don't even want to know what goes on after
that! Geek!! ;-)

David.

chuck duffy wrote:

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>
> Why? Check any math text, sine wavs are always represented with signed numbers,
> doesn't matter if it is integer or float.
>
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>>>On 10/28/06 3:51 AM, in article 4543283c$1@linux, "Ted Gerber"
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>>>>Your quote here:
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Subject: Re: cubase mixing levels
Posted by chuck duffy on Sat, 28 Oct 2006 17:55:05 GMT
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So... In a sense you are right, but your graph is wrong :-)

Chuck

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Subject: Re: cubase mixing levels
Posted by Dedric Terry on Sat, 28 Oct 2006 19:09:36 GMT
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I think I have clouded the issue. No, wait, I know I have. ;-)

Chuck is right about how bits are translated to amplitude representations, and I knew this to be the case, but when talking about dynamic range we are talking about power in terms of dBFS. Here we don't refer to the +/- aspect of amplitude, but rather signal power only relative to 0dBFS (power is an absolute value since negative power is only theoretical); but, since the maximum value chosen for digital is 0dBFS, we use negative dB values to represent the power range of the signal.

So in actuality, all 1's (sign bit excluded) represent the maximum amplitude of the signal which can be a + or - amplitude, in absolute value, (which translates to 0dB Full Scale digital, but not 0db!), and all 0's represent 0db actual power level ("silence", aka anything below -96dB for 16 bit and -144dB for 24-bit - correct Chuck?).

Since audio is quantized into the word based on voltage, not power, 1.1 volt (I believe this is standard) is given the reference level of 0dBFS at the ADC. So in effect, 24-bits provides better resolution as we approach 0 volts, and 0db in signal power, than 16 bits. The dynamic range is a relative value. 16 or 24-bit words are actual signed values for voltages between 0 and 1.1 volts. The area under that point in the curve is where we get power but that is inverted to create the dB digital range in steps of -6dB per bit from 0dBFS down. So, adding 8 bits to a 16 bit word doesn't decrease the power step size to less than 6dB per bit, but rather extend the power range down an extra -48dB to -144dB - just a matter of detecting lower voltage levels at the ADC so we get better sensitivity in the recording, not the headroom to record louder drummers. ;-)

Do you guys agree with this explanation or is that even more convoluted?

Regards,
Dedric

"chuck duffy" <c@c.com> wrote in message news:45438b69$1@linux...
>
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Subject: Re: cubase mixing levels
Posted by neil[1] on Sat, 28 Oct 2006 21:10:58 GMT

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"Dedric Terry" <dedric@echomg.com> wrote:
>Do you guys agree with this explanation or is that even more convoluted?

I'm totally confused now - I thought it was linear...
everything else in the digital world is until you get into
floating-point stuff, innit? so why not the number of bits in a
given equal-amplitude (vertical) segment of the signal vs the
next segment of the same amplitude differential?

Neil

Subject: Re: cubase mixing levels
Posted by AlexPlasko on Sat, 28 Oct 2006 21:30:32 GMT
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I must be confused too neil .
I thought that with a 0dbfs wave ,the positive peak of the wave would have
a word value of 1111111111111111 and the negative peak word was
0000000000000000. with a 16 bit word
"Neil" <IOUOIU@OIU.com> wrote in message news:4543b952@linux...
>
> "Dedric Terry" <dedric@echomg.com> wrote:
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Subject: Re: cubase mixing levels
Posted by Dedric Terry on Sat, 28 Oct 2006 21:54:25 GMT
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Yes with amplitude you get greater resolution with higher bit depths (more
levels to quantize each sample). This isn't the same as the dynamic range
though.

For each extra bit added, we basically reduce the quantization noise by
6dB - we halve the distance between quantization levels (doubling the number
of quantization levels for that same range of amplitude - 0v to 1.1v).
Since power ratio (dB) is a log function we get a consistent change of

relative power of 6dB for each halving of the distance between quantization levels (when referring to the quantization noise floor - 6dB is also often used to refer to "twice the loudness" - but whether you are referring to 6dB as lowering noise or raising volume the power change is the same, but with digital audio we are referring to lowering quantization noise, not doubling the "loudness" with each bit).

So, dB is a measurement of power ratios (change in power).

dB = 10*log10(P2/P1)
where P1 is the power being measured, and P1 is the reference to which P2 is being compared.

Likewise, voltage ratios are:
A = 20*log10(V2/V1)

So the dynamic range increases by 6dB with every bit we add (by lowering quantization noise with twice the quantization levels), and in the digital realm, the reference is 0dBFS so the range (really "depth") extends down from there as we push the quantization noise lower and lower (hence -144dB for 24 bit). Amplitude level changes are more accurately represented by more bits since we have more levels to depict each level between 0v and 1.1v, but the 0 to 1.1volt range never changes in the converter whether it's a 4 bit or 24 bit converter (of course the max 1.1 volts may vary depending on the converter application, design and analog input transform, but with audio I believe 1.1v is most common).

Anyone else feel free to add or expand on this. I may have still missed some informative bit of techdom, or mis-stated something, considering how I originally turned this murky water into thick mud. ;-)

Dedric

"Neil" <IOUOIU@OIU.com> wrote in message news:4543b952@linux...
>
> "Dedric Terry" <dedric@echomg.com> wrote:
>>Do you guys agree with this explanation or is that even more convoluted?
>
> I'm totally confused now - I thought it was linear...
> everything else in the digital world is until you get into
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> given equal-amplitude (vertical) segment of the signal vs the
> next segment of the same amplitude differential?
>
> Neil

Subject: Re: cubase mixing levels
Posted by Dedric Terry on Sat, 28 Oct 2006 22:04:54 GMT
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Actually for 16 bit, the positive peak is 0111111111111111, and the negative peak is 100000000000000 in signed 2s complement binary. 1 sign bit on the far left. In sign and magnitude, the positive peak is 0111111111111111 and the negative is 1111111111111111.

0 is 0000000000000000 in both cases. Only binary offset puts 0000000000000000 at the most negative peak, but that isn't used in digital audio (just fyi).

Dedric

"alex plasko" <alex.plasko@snet.net> wrote in message news:4543c9d6@linux...
>I must be confused too neil .
> I thought that with a 0dbfs wave ,the positive peak of the wave would
> have a word value of 1111111111111111 and the negative peak word was
> 0000000000000000. with a 16 bit word
> "Neil" <IOUOIU@OIU.com> wrote in message news:4543b952@linux...
>>
>> "Dedric Terry" <dedric@echomg.com> wrote:
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>> Neil
>
>

Subject: Re: cubase mixing levels
Posted by AlexPlasko on Sat, 28 Oct 2006 22:21:21 GMT
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dedric:
so if we use 32 bit floating point math, the quantization error created by
the software data processing will be eliminated?or were you referring to
errors created at the converters?
"Dedric Terry" <dedric@echomg.com> wrote in message news:4543cf8b$1@linux...
> Yes with amplitude you get greater resolution with higher bit depths (more
> levels to quantize each sample). This isn't the same as the dynamic range
> though.

>
> For each extra bit added, we basically reduce the quantization noise by
> 6dB - we halve the distance between quantization levels (doubling the
> number of quantization levels for that same range of amplitude - 0v to
> 1.1v). Since power ratio (dB) is a log function we get a consistent change
> of relative power of 6dB for each halving of the distance between
> quantization levels (when referring to the quantization noise floor - 6dB
> is also often used to refer to "twice the loudness" - but whether you are
> referring to 6dB as lowering noise or raising volume the power change is
> the same, but with digital audio we are referring to lowering quantization
> noise, not doubling the "loudness" with each bit).
>
> So, dB is a measurement of power ratios (change in power).
>
> dB = 10*log10(P2/P1)
> where P1 is the power being measured, and P1 is the reference to which P2
> is being compared.
>
> Likewise, voltage ratios are:
> A = 20*log10(V2/V1)
>
> So the dynamic range increases by 6dB with every bit we add (by lowering
> quantization noise with twice the quantization levels), and in the digital
> realm, the reference is 0dBFS so the range (really "depth") extends down
> from there as we push the quantization noise lower and lower (hence -144dB
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>> next segment of the same amplitude differential?

>>
>> Neil
>
>

Subject: Re: cubase mixing levels
Posted by Ted Gerber on Sat, 28 Oct 2006 23:16:11 GMT
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Thanks Dedric-

Either the author made the mistake you've postulated, or the author
was misqouted.

Thanks again, the people on this newsgroup (like you) make it the great
place to hang out that it is - virtually speaking :)

Ted

Dedric Terry <dterry@keyofd.net> wrote:
>Hi Ted,
>
>16 bit and 24 bit both only represent up to 0dB full scale (FS). The
>dynamic range afforded by 24-bit extends down to -144dB rather than the
>-96dB of 16 bit. That's what we are interested in with digital audio, not
>the theoretical limits above 0.
>
>Once in the DAW, -48dB is represented by the lower 8 bits in 16-bit words,
>and the lower 16 bits in 24-bit words. That's probably how they came to
>that conclusion, but it's mathematically incorrect since the same bit
>actually represents -48dB, you just add an extra 8 onto the bottom of the
>dynamic range, not the top as the quote seems to assume.
>
>Regards,
>Dedric
>
>On 10/28/06 3:51 AM, in article 4543283c$1@linux, "Ted Gerber"
><tedgerber@rogers.com> wrote:
>
>> Your quote here:
>>
>> " if you happen to record your tracks at lower levels (e.g.
>>> -10dB peaks, then your peaks are now at -32dB, which is 1/3 the resolution
>>> of 16 bit audio - that isn't insignificant). "
>>
>> The other statement said that 24bit recording at -48db is equal to a full

>> range
>> 16bit recording...
>>
>> These 2 statements look like they're talking about the same thing (apples
>> to apples). If they are, how are they reconciled? If they're not and I'm
>> misunderstanding...
>>
>> Ted
>

Subject: Re: cubase mixing levels
Posted by Neil on Sun, 29 Oct 2006 00:01:04 GMT
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"Dedric Terry" <dedric@echomg.com> wrote:

>Yes with amplitude you get greater resolution with higher bit
>depths (more levels to quantize each sample).

Yes, that's what I meant, and it's linear, right? In other
words, if a sine wave is full-on 0db flat-out at peaks, then
(in a simplistic example) 12 bits are taken up in half that
amplitude, and 12 bits are taken up in the remaining half, or 6
bits are taken up in a quarter of it, and so on, right?

>This isn't the same as the dynamic range though.

Oooooooh, ok, now I'm confused again.

If you're recording at a low level... let's say peaks of -40
or -50 db on a given track, chances are you're only using 12 or
so bits of that file capacity even though you may
be "recording" at 24-bits, yes? Or are you saying becasue db is
log, not linear, you might actually be only using 6 or so of
your available bits if you're recording at those low peaks of
-40 or -50?

Neil

Subject: Re: cubase mixing levels
Posted by Aaron Allen on Sun, 29 Oct 2006 01:37:50 GMT
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If all mix busses are the same, why did the PT III rigs suck out loud so
badly unless you kicked the master fader down to -15db/-20db?

AA

"Dedric Terry" <dterry@keyofd.net> wrote in message
news:C1684104.4BF6%dterry@keyofd.net...
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> running "sound of summing" debate, that is fast become urban legend:
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> In simple math terms:
>
> 1- Native DAW: 2+2=4
> 2- Paris: ((2-1)+(2-1)) + 2 = 4
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> With SX you adjust the gain yourself on the way in, if you like, or better
> yet, set the levels for the mix at hand as needed, as you go.
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> producing a different sound to a final mix, the only think you gain by
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>
> On 10/27/06 10:03 PM, in article 4542d488@linux, "alex plasko"
> <alex.plasko@snet.net> wrote:
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>> If we can duplicate the way paris handles the mix bus DJ can sleep nights
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>> "Martin Harrington" <lendan@bigpond.net.au> wrote in message
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>>>>>
>>>
>>>
>>
>>
>

Subject: Re: cubase mixing levels
Posted by Dedric Terry on Sun, 29 Oct 2006 02:46:24 GMT
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That's why I referred to native summing - most native apps (all but Saw
Plus) are 32-bit floating point.

ProTools is hardware hybrid like Paris so I categorize them differently.
The earlier systems (through Mix+) used a 24-bit data path with no dither,
until they added the dither mixer somewhere in the Mix+ lifecycle, so in
that case you really did lop bits off the bottom when dropping the fader.
The dither engine slightly improved things, but not the lost bits (may have
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Still the 24-bit buss was still a big problem.

The only reason PTHD sounds better is they doubled up the processing path to
48bits, but it's the same 24-bit dsp chip family they seem to be stuck with.

Dedric

On 10/28/06 7:37 PM, in article 454403d3@linux, "Aaron Allen"
<know-spam@not_here.dude> wrote:

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Subject: Re: cubase mixing levels
Posted by AlexPlasko on Sun, 29 Oct 2006 03:05:07 GMT
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>>> 0dBFS to begin with - there is no where to go with gain addition, and
>>> only
>>> subtraction to work with. You can add a lot more -22dB peak audio files
>>> to
>>> a mix without clipping (with all faders at 0), where only 2 audio files
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>>> -10dB peaks, then your peaks are now at -32dB, which is 1/3 the
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>>> of 16 bit audio - that isn't insignificant). So, if you combine the
>>> flawed
>>> advice to record at -10 to -20dB, with the concept of lowering all
>>> tracks
>>> by
>>> 22dB before you start mixing, and you end up removing most of the
>>> resolution
>>> we work hard for with quality mics, preamps and converters, and mixing
>>> as
>>> much more noise and quantization error than necessary.
>>>
>>> As Martin suggested, record just below, or comfortably below clipping
>>> (as
>>> the source dictates) to maximize your use of bit resolution - e.g. keep
>>> audible that ambience or depth of the recording that tails off down at
>>> those
>>> lower levels, rather than mixing it with mic self-noise, etc.
>>>
>>> John, what I think the reference you quoted is actually saying (or
>>> should
>>> be

>>> at least) is just that 24-bit digital (with a quality front end and
>>> converters) affords a higher signal to noise ratio than analog did, so
>>> pushing record levels to widen the gap between peaks and the noise floor
>>> isn't as critical. But since quite a few really great mics have a noise
>>> floor of around -70 to -74dB, we are still putting noise into half of
>>> the
>>> bits of that glorious 24-bit range.
>>>
>>> In terms of signal vs. noise - why record less signal when you can
>>> record
>>> more?
>>>
>>> Imho, there are a lot of "famous" engineers out there spouting complete
>>> technical rubbish out of lack of true knowledge, or passing along
>>> conversational heresay and conjecture. Sadly, engineering is becoming
>>> more
>>> about letting the gear dictate the recording process than the engineer,
>>> and
>>> I believe that's what's wrong with music today. Too my people think xyz
>>> piece of gear can make a hit, a vibe, or a certain sound just because
>>> someone else did it, but few actually use their skills to create the
>>> sound
>>> by knowing what combinations of gear will help get them there.
>>>
>>> 9/10 times it isn't about obtaining one piece of gear to get "that
>>> sound",
>>> but understanding the 1000 different possibilities and knowing how to
>>> use
>>> any of them.
>>>
>>> The Nuendo and SX audio engines are identical. They also are identical
>>> in
>>> summing to Sequoia/Samplitude, and probably Logic, Audition, and DP too
>>> (I've tested Nuendo and Sequoia beyond the limits of normal recording to
>>> verify this, so this comes from experience and listening, not
>>> speculation
>>> or
>>> internet urban legend).
>>>
>>> Regards,
>>> Dedric
>>>
>>> On 10/27/06 10:03 PM, in article 4542d488@linux, "alex plasko"
>>> <alex.plasko@snet.net> wrote:
>>>
>>>> hi martin
>>>> I think what john is referring to is what started this thread. We are
>>>> trying to emulate the way paris handles files at mixdown, not at

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

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

Subject: Re: cubase mixing levels
Posted by Dedric Terry on Sun, 29 Oct 2006 05:53:14 GMT
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Amplitude quantization is linear in the sense that we are subdividing
voltage in voltage steps - e.g. if the full voltage range is 10V (+/- 5V),
then with 16 bits, we have 65,536 divisions, each representing a 153uV step.

As far as dynamic range, if you are recording at 24 bit, then -48dB would be
technically represented by the lower 16 bits. 8 bits gives us 48dB of
dynamic range, but -48dB within a 144dB dynamic range would fall in the
lower 16 bits, since 0dBFS is the maximum power ratio for any bit depth.

You would still be recording/filling 24-bits, but the upper bits would be
0's.

Software encoding of amplitude would represent this a bit differently, but
the theory in terms of how bit depth relates to representing the audio level
we refer to is the same.

Dedric

On 10/28/06 6:01 PM, in article 4543e130$1@linux, "Neil" <OIUOIU@OIU.com>
wrote:

>
> "Dedric Terry" <dedric@echomg.com> wrote:
>
>> Yes with amplitude you get greater resolution with higher bit
>> depths (more levels to quantize each sample).
>
> Yes, that's what I meant, and it's linear, right? In other
> words, if a sine wave is full-on 0db flat-out at peaks, then
> (in a simplistic example) 12 bits are taken up in half that
> amplitude, and 12 bits are taken up in the remaining half, or 6
> bits are taken up in a quarter of it, and so on, right?
>
>> This isn't the same as the dynamic range though.
>
> Oooooooh, ok, now I'm confused again.
>
> If you're recording at a low level... let's say peaks of -40
> or -50 db on a given track, chances are you're only using 12 or
> so bits of that file capacity even though you may
> be "recording" at 24-bits, yes? Or are you saying becasue db is
> log, not linear, you might actually be only using 6 or so of
> your available bits if you're recording at those low peaks of
> -40 or -50?
>
> Neil

Subject: Re: cubase mixing levels
Posted by Neil on Sun, 29 Oct 2006 07:56:28 GMT
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Dedric Terry <dterry@keyofd.net> wrote:
>That's why I referred to native summing - most native apps (all but Saw
>Plus) are 32-bit floating point.
>
>ProTools is hardware hybrid like Paris so I categorize them differently.
>The earlier systems (through Mix+) used a 24-bit data path with no dither,
>until they added the dither mixer somewhere in the Mix+ lifecycle, so in
>that case you really did lop bits off the bottom when dropping the fader.
>The dither engine slightly improved things, but not the lost bits (may have
>added a double precision summing section - don't recall specifically).
>Still the 24-bit buss was still a big problem.

Plus, those 888 convertors sucked hind tit - that sure didn't
help matters, summing issues or no.

>The only reason PTHD sounds better is they doubled up the processing path
to 48bits

And the newer Digi convertors really do sound good - start with
crap, end with crap - highly polished crap, but crap
nonetheless. Start with something good - it's up to the user to
fuck it up from there!   LOL

Neil

Subject: Re: cubase mixing levels
Posted by animix on Mon, 30 Oct 2006 13:33:06 GMT
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I knew that a cowbell would somehow ind it's way into this.

"Neil" <OIUOIU@OIU.com> wrote in message news:45436d90$1@linux...
>
> Dedric Terry <dterry@keyofd.net> wrote:
>
> >I agree with Martin completely. To add my own general opinion
> >on this long running "sound of summing" debate,
>
> Great post, Dedric! BTW, "The Sound of Summing" - wasn't that a
> song by Summon & Nullfunkel?
>
> Anyway, a couple of things I wanted to bring up. You said:
>
> >In simple math terms:
> >1- Native DAW: 2+2=4
> >2- Paris: ((2-1)+(2-1)) + 2 = 4
> >With SX you adjust the gain yourself on the way in, if you like, or
better
> >yet, set the levels for the mix at hand as needed, as you go.
>
> I just wanted to point out that if anyone thought I meant to
> adjust your levels "on the way in" (i.e.: using the Channel
> input trim control), when I had said to default all your
> channels to -6, for example, as a starting point - that I did
> NOT, in fact, mean the input trims!!! I meant the regular
> ol' "please make it louder or softer" control. :D
>
>
> >So why do this in Paris? I am guessing Paris had to convert all audio to
> >24-bit if sent to the native cpu for native plugins in order to prevent
> >clipping before you even start dropping faders on the mix (unless they
> >somhow converted to 32-bit float on the EDS chip first, which isn't a
32-bit
> >float chip, so that seems unlikely). (No one would want to mix with most

> >faders at -40dB - it would "seem" wrong). 24-bits for any portion of
> >summing (not tracking) is a limitation esp. if tracked audio files are
near
> >0dBFS to begin with - there is no where to go with gain addition, and
only
> >subtraction to work with. You can add a lot more -22dB peak audio files
> to
> >a mix without clipping (with all faders at 0), where only 2 audio files
> >peaking just below 0dBFS will automatically clip. At least that seems to
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> >the reasoning behind it, but imho, only applicable to prevent a potential
> >problem in the DAW itself, not as prescribed digital audio practice.
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> >Back to native DAWs: While this approach may seem somehow capable of
> >producing a different sound to a final mix, the only think you gain by
> >lowering tracks by 22dB from the start is lower bit resolution. This is
> >especially true if you happen to record your tracks at lower levels (e.g.
> >-10dB peaks, then your peaks are now at -32dB, which is 1/3 the
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> >of 16 bit audio - that isn't insignificant). So, if you combine the
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> >advice to record at -10 to -20dB, with the concept of lowering all tracks
> by
> >22dB before you start mixing, and you end up removing most of the
resolution
> >we work hard for with quality mics, preamps and converters, and mixing as
> >much more noise and quantization error than necessary
>
> Which is why I suggest starting at -6... maybe -10 if you're
> going to be loading up a buttload of tracks. Think about it
> with an analog analogy again, gang: If your analog mixing
> console goes up to +10 on each channel, would you start a mix
> with every fucking fader maxed out at +10??? Hit "play", and
> how would THAT summing buss sound right about then? No, to
> start out with, you'd bring all the faders down to 0, or -5,
> or -10, or whatever the your comfortable starting point was,
> knowing the console & how much headroom it's got, etc., right?
> Pushing everything up to +10 to start off with would be just
> simply way too much, yes?
>
> So why are people not willing to get their heads around the
> fact that in digital anything past "0" is "way too much"?
> The only reason your DAW has +6 or +8 on any given channel is
> in case you fuck up & record the cowbell track on your band's
> cover version of "Mississippi Queen" at peak levels of -17db
> ... "0" gain on the channel level just wouldn't cut it at that
> stage, the cowbell just wouldn't be audible enough to drive
> that tune. :D

>
> So, just like in the analog console, where 40+ channels
> recorded to needle-bending levels on overbiased tape machines,
> and every channel set to +10 at the start of your mixdown
> session would sound like crap; so does 40+ channels recorded to
> nice hot levels, barely missing overs by a tenth of a db,
> with every channel set to "0" result in a similar thing.
>
> Neil
>

Subject: Re: cubase mixing levels
Posted by fernando on Tue, 31 Oct 2006 16:31:08 GMT
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Do you know, my E-MU 1212M and new 1616M PCI cards has the
same converters as PTHD, A/D convertor is AK5394A and D/A
convertor is CS4398, sounds real good for not colored hi-end stuff. Very
cheap too, $199 with bundle of SOANAR LE, Cubase LE, Ableton Live Lite 4,
Wavelab Lite and other cool stuff like Proteus X LE. But I think 1212M is
now not supported, sounds familiar, yes?

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>Neil