

## Levels in Digital Audio

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## About this Article

Unless otherwise stated, this article deals with levels in digital audio, ITB mixing and floating point DAWs. This includes sequencers such as Logic Pro, Cubase/Nuendo, Pro Tools LE (not HD), FL Studio, Ableton Live and Digital Performer.

## Word Explanations

dBFS = decibels relative to full scale digital audio

0 dBFS = maximum possible level in digital audio before clipping occurs

A/D = analog to digital conversion (going into the sound card)

Bits = dynamic range in digital audio

Clipping = exceeding the available headroom or flat-topping the waveform

DAW = Digital Audio Workstation

D/A = digital to analog conversion (output from the sound card)

Floating point = calculations that allow for high internal dynamic range and lossless scaling

Headroom = amount of dB left before clipping

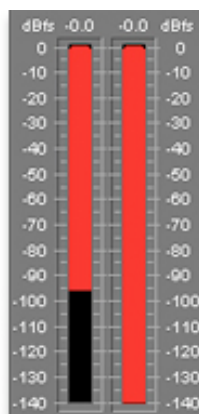
ITB (In-The-Box) = working in a computer based system

Summing = adding individual tracks in a bus or master output

Unity = default setting where the signal is passed through 1:1 in level

## Bit Depth when Recording

Record in 24 bits instead of 16 bits for the highest dynamic range.



**LEFT** - 16 bit dynamic range (CD quality equivalent) is a theoretical 96 dB

**RIGHT** - 24 bit dynamic range is a theoretical 144 dB in comparison

**NOT PICTURED** - 32 bit floating point internal range (for overflow) exceeds 1,500 dB

🔑 The extra bits when recording in 24 bits mean better resolution for low level signals. But it also enables you to record at lower peak levels with more headroom before clipping the input - and without sacrificing sound quality. Even though your target bit depth may be 16 bits (CD) or MP3 there is a lot to be gained from using more bits during the recording, mixing and mastering process.

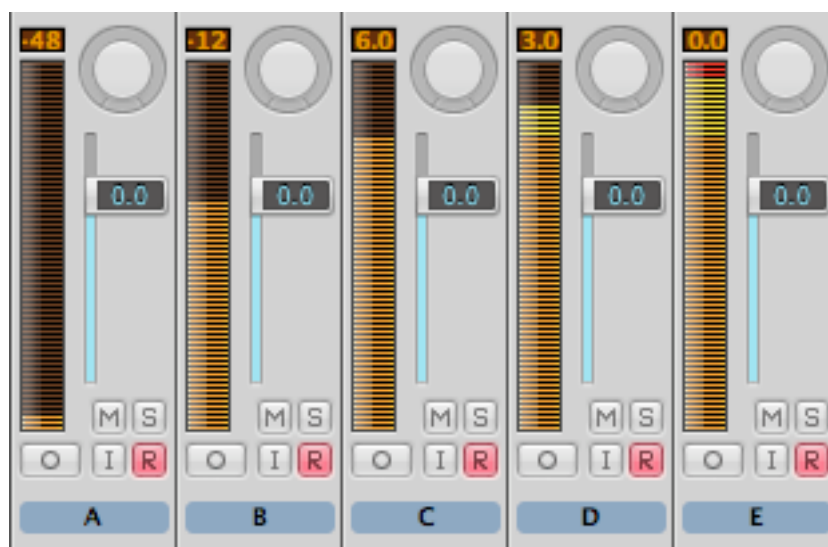
## A Word about Meter Values

Some DAWs cap the displayed overload value at e.g. +6 dB but the actual overload could be higher. Also, some DAWs do not always display a - (minus sign) in front of dBFS values due to space restrictions but it is implied. Some DAWs disagree on how many consecutive clipped samples constitute an “over” but this is of little practical relevance.

## Levels when Recording

Record with a maximum peak of -3 dBFS to get the best dynamics and signal-to-noise ratio during A/D conversion in your sound card or converter.

You can also record at lower levels, e.g. with a peak of -6 dBFS or -12 dBFS without any significant loss in signal quality. Record too low though, and the quality begins to suffer.



- A** The peak level is too low - you are not taking advantage of the dynamic range
- B** Peaking at around -12 dBFS is recommended
- C** Peaking at around -6 dBFS is recommended, too
- D** Peaking at -3 dBFS is the recommended maximum peak level when recording
- E** The peak level is too high - you are likely to get unwanted artifacts or distortion

🔑 Recording very hot levels near the clipping point of 0 dBFS can result in the A/D producing inferior results - even before it officially overloads the input. Notice that the channel meter in a DAW will not necessarily show if you overloaded during recording. It may show a 0 dBFS reading instead, no matter the amount of overload. So you need to watch your levels before it is too late and always leave some headroom.

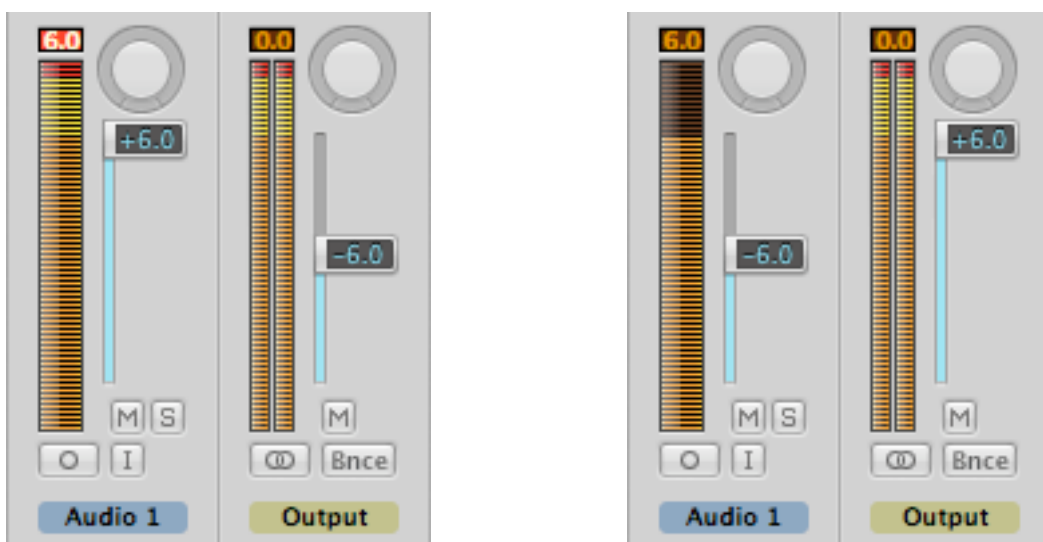
## Debunking some Myths

Here are some examples of when overloading does not degrade or even change the result - as long as you pay attention. However, this does not mean that overloading is encouraged since other stages of the audio chain can be susceptible to overloads.

### Myth 1: Overloading the output of a channel will cause irreparable damage to the audio

*Fact:* There is no damage or even change to the audio if you lower the master fader until the overload goes away on the master bus.

🔑 Overloading a channel by 6 dB and lowering the master fader by the same amount makes the overload go away even though the channel output is still overloading. This is possible due to the lossless scaling used in the DAW summing engine.



**LEFT** Channel fader overloads by 6 dB but the output fader recovers perfectly

**RIGHT** Channel fader down by 6 dB and the output fader scales the signal up perfectly

Both results are 100% identical in sound, despite the left example having an overloading channel, and the right example is using an upscaled signal.

### Myth 2: Using high levels on many tracks can cause problems in the summing engine

*Fact:* While there are some definite workflow benefits to using lower levels during mixing there's nothing inherently better sounding about using lower levels as long as you pay attention to not overloading plug-ins or overloading the D/A.

🔑 Summing 60 tracks using low levels and then pulling up the master fader will result in exactly the same result as summing the same tracks with high or overloading outputs and then pulling down the master fader. Again, this is possible due to floating point math and works in exactly the same way as described in myth 1. Even if you overload all 60 tracks each by 12 dB you would not be able to exceed even half the headroom of the summing engine.

### Myth 3: Using low levels means better channel fader resolution

*Fact:* Since a channel fader has most of its resolution at the top of the scale, the myth stipulates that using high levels will force you to set your faders at the bottom of the scale where resolution is worse. However, this argument for using low levels has little impact since you can overload the individual channel outputs or the master output - and then lower the master fader without any quality loss. Still, there are some workflow benefits to using lower levels because you avoid having to use gain or trim plug-ins on the busses.

### Levels Matter with some Plug-ins

Once you start using dynamic plug-ins without input attenuation the rules change a bit. This is why gain structure still matters when working in all kinds of DAWs.



So overloading during mixing or processing ITB can happen in two ways:

- A** Overloading the channel output by pulling up the channel fader
- B** Overloading the plug-in chain internally due to an increase in level inside a plug-in
- C** Shows overloading in the plug-in chain but it is hidden due to a lowered channel fader

In the case of **A**, we're not overloading the gain structure of the plug-in chain. This is due to the fact that the channel fader is located after any processing occurs, and that the overload occurs at this later point - something which can be rectified by lowering the master fader.

In the case of **B**, the overload occurs inside the processing chain before the channel fader. So any output overload in a plug-in will affect the input of the the next plug-in in the chain. This can cause distortion or unpredictable behavior in some plug-ins, especially dynamic plug-ins such as compressors and limiters. You can see the amount of actual overload by setting the channel fader at unity, but you need to bypass any overloading plug-ins first.

Internal plug-in chain overload is not audible until you insert a plug-in that actually reacts to the overload, usually a dynamic plug-in. And as example **C** shows, it is not necessarily visible on the channel output meter either, i.e. if you lowered the channel fader until the

output signal went below the clipping point. So you need to pay attention to your output level in each plug-in in your plug-in chain which is why lower levels from the start is smart.

🔑 Playback a sine test tone peaking at 0 dBFS on a channel. Insert a gain or trim plug-in (or an equalizer) and boost the output level by 6 dB using the plug-in. This overloads the channel by 6 dB internally, which will also show on the output meter if the channel fader is set to unity level. Insert a Waves L1 or a similar dynamic plug-in that lacks input attenuation. Notice how the plug-in will do 6 dB of gain reduction without the possibility of changing this amount in the plug-in, effectively making the threshold unusable. This experiment is repeatable to various degrees with several other plug-ins including other 3rd party and bundled sequencer plug-ins.



Uncontrollable overload of a dynamic plug-in that lacks an input attenuation option

## How to Avoid Overloading a Plug-in

The traditional solution is to use the output gain knob in the previous plug-in to lower the signal being passed on to the next point plug-in. This is similar to the workflow of analog mixing. There is no quality loss with this method.

🔑 A similar option is to attenuate the input of the overloading plug-in. However, not all plug-ins have an input gain knob but most have an output knob, which is why the above solution is usually better. Another option is to insert a dedicated gain or trim plug-in before the overloading plug-in. Use the gain plug-in to lower the level of the signal before it reaches the next plug-in. There is no quality loss with this method either.


## How to Avoid Overloading a Bus or Master

If you are not using any plug-ins on the bus or master channel you can simply lower the relevant bus or master fader to remove any overloads without problems.

If you are using plug-ins then the best option is to use a gain or trim plug-in as the first insert on the bus, and use that plug-in to lower the level until it is safe again.

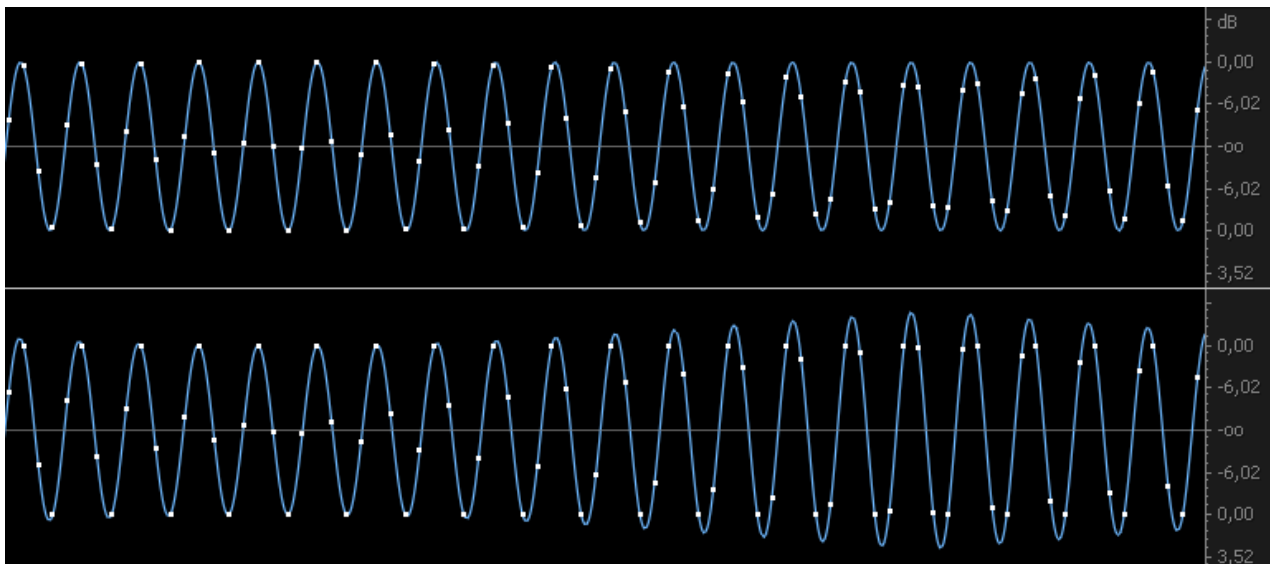
## Levels when Processing with Plug-ins

Aim for a peak of around -6 dBFS on individual channels to be on the safe side, though you are likely not to experience any problems using higher values. This value is not necessarily the peak value shown on the output meter of the channel since that value can be offset and hidden by a lowered channel fader, as explained earlier. Rather it is the internal level of the signal chain which you can see by setting the channel fader at unity.

 The above suggestions make sense if you care about fidelity. Many professional mixing engineers will routinely ignore this and clip on purpose using dedicated plug-ins, or even intentionally overloading the master channel before the mastering process. Overloading the master channel during the mixing process is not recommended since it somewhat limits what the mastering engineer can do.

## Levels when Outputting from a D/A Converter

When audio is output from the DAW to the D/A converter, the digital sample points are reconstructed as an analog waveform in the D/A. The sample points do not appear exactly on the peaks all the time which means that the actual analog waveform reproduced in the D/A can be higher in amplitude, and potentially overload the output.



Extra D/A headroom is needed for some waveforms once reconstructed (lower row)

If you want to be on the safe side you should output a signal that peaks somewhere between -12 dBFS and -6 dBFS. This also has the benefit of matching the input requirements of most analog gear better than very high output levels. It also means that you can safely sample rate convert any exported file without worrying about overloading.

🔑 A sine wave can require up to 3 dB of extra D/A headroom to be reproduced without distortion. However, artificially created square waves can require 6 dB or even more D/A headroom due to high level intersample peaks. Clipping audio in the mix can create something similar to square waves. The actual amount of audible distortion during D/A conversion depends on the quality of the D/A. You can also use an intersample peak meter to watch for these problems though it's rarely necessary as long as you follow the advice in this article.

## Mix Output Levels

The D/A output level issue is also relevant to the final mix level. This has been covered in more detail in the PDF document entitled "Mixdown for Mastering", available for download here: <http://www.onlinemastering.dk/mastering-faq.html>

🔑 The mastered mix is subject to the same potential D/A overload problems, in fact even more so due to the often digitally limited or intentionally clipped audio in combination with high RMS (average) levels. This kind of processing provides a very loud master but the trade-off is potential and unpredictable distortion during playback on various D/As, including sound cards, CD players and iPods. When bouncing or exporting a WAV file to MP3 conversion you should leave at least 0.3 dB of headroom. This will avoid at least some of the potential distortion during the MP3 conversion process.

## Normalizing

Normalizing can never improve the existing sound quality, and except for a very few situations (usually when dealing with automated batch processing of files) normalizing is a completely unnecessary procedure that should be avoided.

🔑 Some DAWs have a normalize option in the bounce window. If this function is enabled your final mix will have its peak level normalized to 0 dBFS. So if you had 3 dB of headroom in your mix before clipping the master bus, normalizing would raise the total level by 3 dB. However, if your mix was overloading by 3 dB, floating point normalizing would bring your mix back to a peak level of 0 dBFS without clipping, and in the process making your mix 3 dB lower. Logic Pro also offers an option called Overload Protection Only, which works like floating point normalizing except it will never raise the level, only lower it if necessary. However, none of these functions are recommended since you should watch your mix levels and leave some extra headroom for mastering.

## Dithering

Explaining dithering is outside the scope of this article. It is always a good idea to apply dithering when going from a higher bit depth to a lower bit depth. This is especially important when reducing to 16 bits, where quantizing artifacts in the lowest bits can sometimes be heard as low level distortion.

## Related Articles

Mixdown for Mastering - written by Holger Lagerfeldt  
<http://www.onlinemastering.dk/mastering-faq.html>

Loudness when Producing and Mixing - written by Holger Lagerfeldt  
<http://www.gearslutz.com/board/music-computers/468170-loudness-when-producing-mixing-tips.html>

Inter-sample Peaks - written by Andreas Nordenstam, who also provided some pictures  
<http://www.gearslutz.com/board/tips-techniques/334385-intersample-peaks.html>

Overload in Signal Conversion - written by Søren H. Nielsen and Thomas Lund  
[http://www.tcelectronic.com/media/nielsen\\_lund\\_2003\\_overload.pdf](http://www.tcelectronic.com/media/nielsen_lund_2003_overload.pdf)

0 dBFS+ Levels in Digital Mastering - written by Søren H. Nielsen and Thomas Lund  
[http://www.tcelectronic.com/media/nielsen\\_lund\\_2000\\_0dbfs\\_le.pdf](http://www.tcelectronic.com/media/nielsen_lund_2000_0dbfs_le.pdf)

Level Control in Digital Mastering - written by Søren H. Nielsen and Thomas Lund  
[http://www.tcelectronic.com/media/nielsen\\_lund\\_1999\\_level\\_co.pdf](http://www.tcelectronic.com/media/nielsen_lund_1999_level_co.pdf)

Stop Counting Samples - written by Thomas Lund  
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