Grimm | AUDIO

LevelNorm manual

Please read this manual before operating the software!

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Corresponding to software version 1.2

Introduction

Thank you very much for selecting Grimm Audio's LevelNorm for your audio normalizing tasks within an Avid audio or video workstation environment. LevelNorm is build to support one of the most important achievements in the audio industry for decades: the change from peak normalization to loudness normalization. This "true audio revolution" started when the ITU submitted the BS.1770 'LKFS' (also known as LUFS) loudness metering standard in 2006. The European Broadcast Union EBU took the lead in building a broadcast recommendation upon this fundament, called R128. It was released in 2010. Eelco Grimm of Grimm Audio was one of the active members of the EBU PLOUD committee that created the recommendation. The committees work has been made possible by the aid of a piece of software developed by Grimm Audio's Wouter Snel and Jorn Lemon. Grimm Audio's LevelOne is the enhanced version of the actual software that made R128 happen and LevelNorm is based on LevelOne's algorithms. We are very proud to offer you this great piece of software that will enhance your daily work spectacularly.

Applications

LevelNorm is at home in many applications. File preparation for broadcast delivery during AV post production of all sorts is of course the most obvious application. Broadcast stations around the world start using ATSC and EBU recommendations and the demand for material to be compliant is growing rapidly. Using LevelNorm for the official loudness normalization before deliverance is a huge time saver. But there are more applications for this powerful technology. For instance LevelNorm can be used to loudness normalize raw material received from different reporters and thus loudness jumps are automatically avoided when editing an item. In small scale broadcast stations LevelNorm can even serve as part of an ingest system if Avid workstations are used.

Working with LevelNorm

The best way to mix audio files for broadcasting in the new paradigm may at first seem quite dangerous. Basically you just lower your modulation levels, ignore your meters and focus on the sound. Once in a while you take a glimpse at the digital peak meter to make sure you are not clipping. Don't be afraid that you will loose resolution by modulating low, with current 24 bit high resolution equipment this is no threat anymore.

If you feel insecure working without visual feedback, please consider using a realtime EBU mode loudness meter. Alternatively use your VU's or PPM's but align them correctly. A proper alignment for your VU meter would be 0 VU to equal -18 dBFS. For analog PPM's, keep the 0 PPM = -9 dBFS level. During production, avoid modulating hot all the time. But at the same time do not hesitate to use all that headroom. If you are using a digital peak meter next to the other ones, ATSC and EBU allow peaks to reach almost 0 dBFS, so don't worry about short peaks above -9 dBFS.

The key to almost automatically achieve consistent results without using meters, is to adopt the film postproduction practice of aligning the monitor gain to a fixed acoustic level. In the broadcast case, the EBU tech 3276 document prescribes to align a -18 dBFS (rms measured) pink noise track to a sound pressure level of 82 dBA per speaker for stereo or 78 dBA per speaker for 5 channel surround. If you prefer to have your monitor SPL a little lower, just note the deviation from the calibrated level for later reference. The alternative is to simply normalize a well known file to -23 LUFS, play it back and set the volume control to your preferred listening level. The main concept is to now mark the monitor control position and work at this reference level position 90% of the time or more. Your ears will tell you how loud you should mix, you don't need a meter for that anymore.

When you are satisfied with your mix and overall sound, just print the result to a seperate stereo or surround track, let LevelNorm analyze and process the file on this track and your audio is finished for release. After some practice you will note the necessary adjustment by LevelNorm will be minimal.

Broadcast and post production engineers in the US know that according to the ATSC recommendation A/85 they need to adjust the volume of their program based upon analysis of the dialog track(s) only. LevelNorm can be a big time safer here because it allows to very quickly measure the loudness of (part of) the dialog track. The 2009 edition of ATSC A/85 has no 'relative gate' in the measurement and a target of -24 LKFS (= LUFS) in stead of -23 LUFS like in EBU R128. The LevelNorm-ATSC plugin complies to that standard. Since the ITU has adopted the EBU R128 relative gate in 2011, it is now unclear if ATSC A/85 advices to measure 'all with gating' or dialog only. Grimm Audio will closely follow the developments.

Installation

Windows

After downloading the setup file, double-click it to start the installer. You might see a general security warning that the file is downloaded from a website. If so, just press the "run" button. The installer will guide you through the install process. You don't need to change any of the settings unless you wish, although US customers probably don't want to install the EBU version and EU citizens probably don't want to install the ATSC version. When finished you may remove the setup file.

Using the standard install, LevelNorm will be placed in:
for windows 32 bit:
C:\Program Files\Common Files\Digidesign\DAE\Plug-Ins\
for windows 64 bit:
C:\Program Files (x86)\Common Files\Digidesign\DAE\Plug-Ins\

Please note: LevelNorm requires a "Microsoft Visual C++ 2005 Redistributable Package (x86)" to be installed. This installer is included in the 'LevelNorm Setup.exe' file and will be installed automatically.

Registration

To launch the registration wizard go to Start -> 'All Progams' -> 'LevelNorm' and click on 'LevelNorm-Registration'. A window will open, displaying the current license if any. To add/change the license, click 'Edit your LevelNorm license'. A wizard will guide you through the registration process. For demo purpose a trial serial number can be obtained in the web shop. Please note that you must use the eight character user name of the web shop in step 1. If you have had a previous version of LevelNorm install on the system the wizard will automatically adopt the settings of that installation.

Mac

Download the Levelnorm.dmg file from the web shop. Open the disk image by double clicking it. You might see a general security warning that the file is downloaded from a website. If so, just press the "run" button. Drag the 'LevelNorm-EBU.dpm' and/or the LevelNorm-ATSC.dpm' file to your the plugins folder, "MacHD/Library/Application Support/Digidesign/plugins/" to install the application. After starting your Avid system, LevelNorm can be found in the 'dynamics' section of the plugins and in its own 'Grimm Audio' section.

Registration

With the Levelnorm.dmg disk still mounted, double click the RegWizard app to start the registration procedure. A window will open, displaying the current license if any. To add/change the license, click 'Edit your LevelNorm license'. A wizard will guide you through the registration process. For demo purpose a trial serial number can be obtained in the web shop. Please note that you must use the eight character user name of the web shop in step 1. If you have had a previous version of LevelNorm install on the system the wizard will automatically adopt the settings of that installation.

Check for Updates

To avoid annoying automatic update checks within your Avid environment, you will need to visit the Grimm Audio website manually to check if there is an update available. You can see if you run the latest version by checking the 'rev' number of your LevelNorm in the right down corner of the plugin interface window.

System Requirements

Recommended System Requirements

Windows

- XP (x86), Vista (x86) or Windows 7 (x86)
- Avid Protools 8 or higher, Media Composer (Symphony or News Cutter) 5.5 or higher

Mac

- Intel only
- OSX 10.5 or Higher
- Avid Protools 8 or higher, Media Composer (Symphony or News Cutter) 5.5 or higher

Basic Operation within Protools

LevelNorm is an AudioSuite plugin and therefore only operates on a selected file or part of a file and not in real time on a channel of your virtual mixing desk. Consequence is it will ignore the fader positions of your mixing desk. This means that if you like to use LevelNorm for making your final mix comply to the EBU or ATSC recommendation, first you need to print your mix to a separate mono, stereo or surround track. Operation then is as usual with Audio Suite plugins: select the audio file and open LevelNorm from the AudioSuite menu (it can be found in the 'Grimm Audio' section). The LevelNorm User Interface window now opens, as seen below.

Please make sure that the 'format' is set to the amount of channels in your audio selection. LevelNorm opens in 'stereo' mode by default, you can change this setting by clicking the window next to the text 'format'. A drop down menu will open with a choice of mono, stereo and various surround channel counts and orders. If you analyse or process a smaller track count than indicated in this window, an error message will be shown. Note that if you analyse a larger track count than indicated (for instance a stereo file and a mono setting), LevelNorm will analyse the indicated amount of channels and thus return a wrong value.

AudioSuite processing in Protools has relatively large freedom in selecting files and part of files. For instance for stereo either a true stereo track or two mono tracks can be selected. It is also possible to select part of a file or even more than one file on the timeline and analyse or process the total amount of selected audio.

After the track count format, set the target level. With LevelNorm-EBU, 0 LU equals -23 LUFS. For LevelNorm-ATSC, 0 LU is -24 LKFS. Normally your target will be 0 LU. When using LevelNorm in the course of your mix process however, it can be convenient to normalize for instance background music to your favorite level below 0 LU. This may speed up your production time considerably. To use this function, just select the preferred target level with the slider before clicking 'Process'.

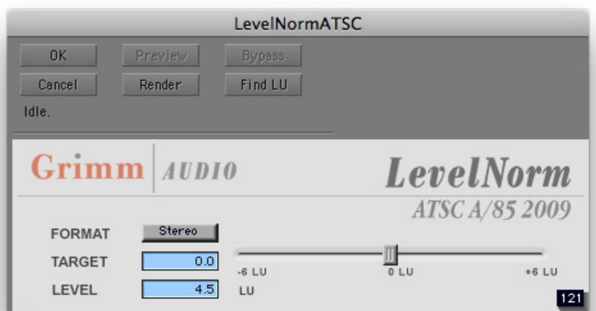
Next step is to either only 'Find LU' of the file, or directly 'Process' it to the loudness target level set with the slider. A progress bar will be shown during the calculation and the normalized file will replace the original file on the timeline. If the average loudness of the file was so low that normalizing it to 0 LU would cause a clip, LevelNorm will show a warning and amplify the file to a maximum peak level of 0 dBTP. Mark that to maintain the correct 0 LU level of a normalized track during a second print, the channel fader and master fader should be set to 0 dB.

Normally your target level will be 0 LU. When using LevelNorm in the course of a mix process however, it can be convenient to normalize for instance background music to your favorite level below 0 LU. This may speed up your production time considerably. To use this function, just select the preferred target level with the slider before clicking 'Process'.

Basic Operation within Avid video workstations

LevelNorm is an AudioSuite plugin and therefore operates on files, not in real time. This file needs to actually exist in the bin (or timeline) and it must be a stereo file if the broadcast is in stereo. In an Avid video editor environment, AudioSuite can only process one file at a time and therefore two mono files do not make a stereo pair. Fortunately, starting from Avids Media Composer 5 stereo tracks can be used in the timeline. Note however that the default track type is still mono. Consult the Media Composer manual on how to create stereo tracks. To learn more about the stereo feature you may also like to view this online tutorial http://youtu.be/IPROSVyy_p4 (3rd party).

Please make sure that the 'format' option of LevelNorm is set to the amount of channels in your audio selection. LevelNorm opens in 'stereo' mode by default, you can change this setting by clicking the window next to the text 'format'. A drop down menu will open with a choice of mono, stereo and various surround channel counts and orders (in Avid video workstations the surround formats are of no use). If you analyse or process a mono file in stereo mode, an error message will be shown. Note that if you analyse a stereo file in mono mode, LevelNorm will analyse just the left channel and thus return a wrong value.



To normalize your audio, the first step is to print your mix to a new stereo file. Now open the AudioSuite toolbox from the 'tools' menu. Select the stereo audio file on the time line or drop it from the bin on the AudioSuite toolbox. Select LevelNorm-EBU or LevelNorm-ATSC from the plugin menu within the AudioSuite toolbox. Open and activate the plugin interface by clicking the purple plugin icon. The user interface looks like the screenshot above. Select your desired format and target level in the LevelNorm interface (usually 'stereo' and '0 LU'). Next click 'Render' and wait for the analysis and rendering to finish. Click OK. The file is now leveled to the selected target. If the average loudness of the file was so low that normalizing it to 0 LU would cause a clip, LevelNorm will show a warning and amplify the file to a maximum peak level of 0 dBTP. Mark that to maintain the correct 0 LU level of a normalized track during a second print, the channel fader should be set to 0 dB. LevelNorm also offers an option to just analyse a selected audio file's loudness. To do so, click 'Find LU' in stead of 'Render'. After analysis, the plugin window shows the file's loudness but has not changed its level.

If your mix mainly consists of dialog, a less precise but much faster method is to normalize the main voice channel and mix the other channels in balance around this. For ATSC compliance this is actually the preferred way of working. In that case, follow the instructions above, but select the voice track, and make sure LevelNorm is set to 'mono' format if your track is mono. This method also works in rushed environments like news editing where often no time is left to normalize the stereo end mix. It also helps to rapidly match the loudness of various tracks like interviews from different origins. Do this by selecting all voice files on the same track and apply LevelNorm to them. They will be analysed and rendered independently.

Normally your target level will be 0 LU. When using LevelNorm in the course of a mix process however, it can be convenient to normalize for instance background music to your favorite level below 0 LU. This may speed up your production time considerably. To use this function, just select the preferred target level with the slider before clicking first 'find LU' and then 'Render'.

Technical specifications

LevelNorm-EBU uses the official EBU R128 settings:

- ITU BS.1770-2 measurement, including -10 LU 'background sound' gate
- 0 LU = -23 LUFS
- Adjustment of the target level to 0 LU +/- 6 LU

LevelNorm-ATSC uses the 2009 version of the ATSC A/85 recommendation:

- ITU BS.1770-1 measurement, without relative gate
- 0 LU = -24 LKFS
- Adjustment of the target level to 0 LU +/- 6 LU

Surround channel order

The LUFS measurement has a different weighting for the surround speakers. Therefore LevelNorm must know the channel order of multichannel files in your facility. There are three options:

- SMPTE/ITU (L-R-C-Lfe-Ls-Rs)
- Film (L-C-R-Ls-Rs-Lfe)
- DTS (L-R-Ls-Rs-C-Lfe)

References

http://tech.ebu.ch/loudness provides all kinds of information about the EBU R128 broadcast loudness recommendation. The official R128 documents and guidelines can be found, as well as introduction papers and videos.

At http://www.atsc.org/cms/index.php/standards/recommended-practices/185-a85-techniques-for-establishing-and-maintaining-audio-loudness-for-digital-television the ATSC A/85 recommendation is available for download.

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