DOLBY®

Dolby® Professional Loudness Solutions

Information for Prospective Licensees
Issue 3

Table of Contents

Introduction

Broadcasters and content producers use Dolby® Professional Loudness solutions to help ensure consistent audio loudness levels. Because differences in levels provoke a large number of consumer complaints, all steps in the broadcast production chain require tools to help maintain uniformity. Dolby Laboratories initially implemented loudness measurement with the Dolby LM100 Broadcast Loudness Meter and now provides a range of tools aimed at helping to deliver content with more consistent loudness levels.

As file-based workflows started to become more prevalent, Dolby developed the DP600 Program Optimizer. This award-winning tool enabled broadcasters and content producers to ensure that the loudness levels of audio content were set correctly. Users could now analyze content in a wide range of formats, and if they found that the loudness levels were incorrect, the DP600 would automatically adjust the levels to meet the users' required specifications.

Use of these tools should ultimately lead to a reduction in customer complaints. Television viewers set their preferred listening levels subjectively, based on the loudness of dialogue. They do not adjust their volume controls for naturally occurring variations in programs, such as quiet sections with no dialogue or brief high-level impulses such as gunshots. They do adjust the volume—and complain—when the dialogue level varies significantly from one channel or program to another, or when commercials sound much louder than programs.

The first stage of fixing these problems is to objectively and accurately measure what viewers subjectively experience. Dolby technologies accomplish this by measuring appropriate portions of the signal—those portions that contain dialogue. The Dolby Professional Loudness Metering solution combines the use of standards-based loudness estimation algorithms with Dialogue Intelligence™ technology. Dialogue Intelligence accurately detects when content contains speech, and it measures loudness only during those periods.

The second stage of fixing these problems is to correct the content based on knowledge of the measured loudness levels. Content in a Dolby metadata-aware format, such as Dolby E or Dolby Digital Plus, will have the loudness metadata adjusted so that it correctly matches the associated audio content. Updating of the metadata maintains higher audio quality and reduces the computational complexity of the loudness correction process. For formats that are not metadata-aware, such as linear PCM, the content is adjusted to a user-defined target level. The Dolby Professional Loudness Correction library enables manufacturers to add this functionality, which has proven itself within the Dolby DP600 Program Optimizer, to their Intel® processor-based file-processing products.

Dolby Laboratories licenses professional equipment manufacturers to incorporate Dolby Professional Loudness Metering and Correction solutions in a variety of products, including video encoders and decoders, monitoring products, and test and measurement products. The license covers real-time and file-based applications for measurement, and file-based applications for correction. This license also covers real-time updating of loudness metadata based on premeasured values for Intel processor-based products. This document describes the licensing options and procedures for the developers of products and implementations that incorporate Dolby Professional Loudness Metering and Correction solutions.

Dolby also manufactures a range of products that use the same loudness measuring and correction technologies. Manufacturers that are developing products based on this license may use these Dolby products for reference or comparison. For more details on these products, see the Dolby broadcast loudness solutions overview.

For more information about the Dolby DP600 Program Optimizer, please refer to the Dolby DP600 Program Optimizer page.

1

Technology Overview

The Dolby® Professional Loudness Metering solution combines the use of standards-based loudness estimation algorithms with Dolby Dialogue Intelligence™ technology. Supported algorithms are those most prominently in use worldwide: Leq(A), ITU-R BS.1770-1, ITU-R BS.1770-2, and EBU R 128. These algorithms create a value that can be assigned as the loudness level of a program, using a combination of tools that include frequency-weighting curves to account for the human ear's particular sensitivity to audio in the middle of the hearing spectrum.

In addition to loudness metering, the license program includes support for true-peak measurement. Current peak-metering methods for digital signals often display only the largest sample value, but the actual peak of the audio signal often occurs between samples. True peak gives a better indication of actual location of the peak.

The license also includes the ability to measure and correct a wide range of audio streams, including Dolby E, Dolby Digital, and Dolby Digital Plus, in many cases without the need for separate encoder and/or decoder agreements.

Loudness meters provide a great way to measure the dialogue normalization (dialnorm) value for inclusion in metadata. Meter products can also verify the dialnorm value in preencoded streams. Both Dolby Digital and Dolby Digital Plus—used for audio encoding in many broadcast standards around the world—include full support for Dolby audio metadata. At the core of this metadata is the dialnorm value, which decoders use to adjust the audio content's replay level in the home. Dolby audio metadata is implemented in all Dolby Digital and Dolby Digital Plus decoders, such as in TVs, set-top boxes, and A/V receivers.

File-based loudness correction products such as the Dolby Professional Loudness Correction solution perform dual-pass loudness correction to ensure balanced audio loudness levels without any of the artifacts introduced by real-time correction processes. The first pass measures the audio level, and the second pass corrects it, thus fully retaining the content creator's intent and the original dynamics of the content.

For more information on loudness estimation algorithms and Dialogue Intelligence, please refer to these publications:

Dolby broadcast loudness solutions overview

ITU-R BS.1770-1 and ITU-R BS.1770-2, *Algorithms to Measure Audio Programme Loudness and True-Peak Audio Level*

EBU R 128, *Loudness Normalisation and Permitted Maximum Level of Audio Signals*

For further information on Dolby Digital Plus and metadata, please refer to these publications:

All About Audio Metadata

Introduction to Dolby Digital Plus, an Enhancement to the Dolby Digital Coding System

License Agreement

If you wish to develop products incorporating Dolby® Professional Loudness Metering and Correction solutions, you are required to enter into a Dolby Professional Loudness Metering and Correction license agreement with Dolby Laboratories. Once the agreement has been signed and we have tested and approved your product, your license will be activated. (Please note that this document is not such a license or an offer of such.) We will then authorize you to use our trademarks and offer your product for sale according to the terms of your specific category of license.
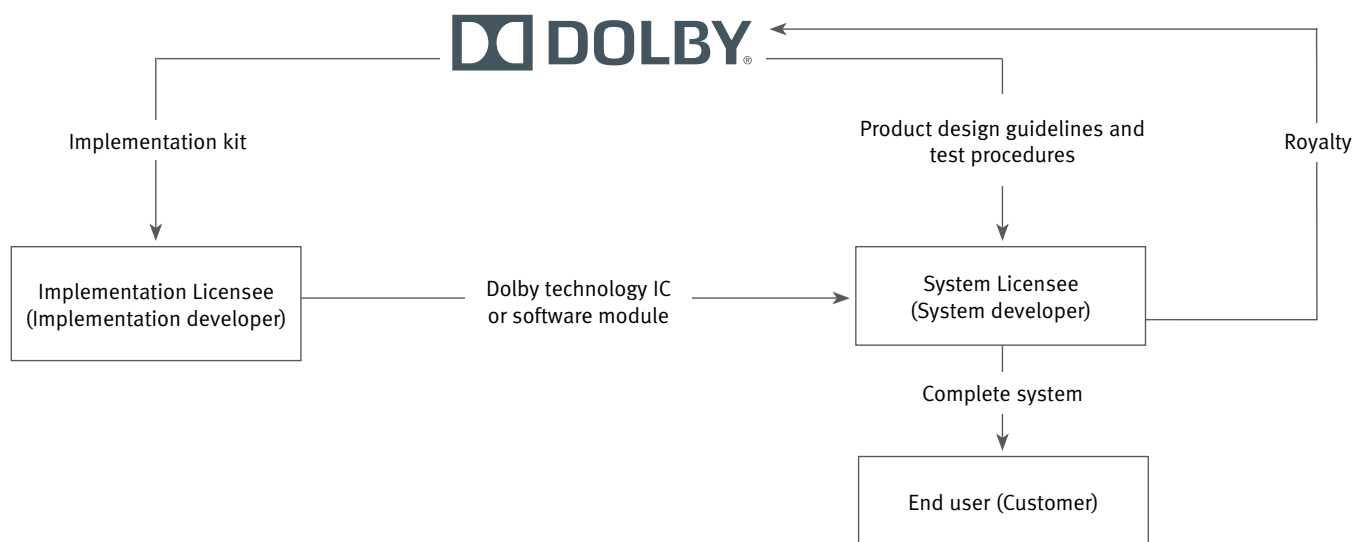
Relationship Between Dolby Laboratories, Its Licensees, and End Users

License Fees and Royalties

There is a one-time licensing fee for Dolby® Professional Loudness Metering and Correction solutions. The license includes patent, copyright, and trademark rights; know-how; and the rights to sell your product to end users. It also includes an appropriate development kit, technical support, and our testing and certifying your products for one year from the time your license is activated. There is an optional per-year maintenance program that covers software updates, documentation and test suite updates, technical support, and additional product testing for the remainder of the license term.

Once approved for sale, your products are also subject to a base royalty on each process contained in each Licensed Product you sell.

Obtaining a License

To obtain approval of your product incorporating Dolby® Professional Loudness Metering and Correction solutions and activate your license, you must complete the following steps:

1. Email licensinginquiries@dolby.com and request the Questionnaire for Prospective Dolby Licensees form. This form will be emailed to you as an interactive PDF, which you can fill out onscreen. Please complete the form in English, then submit it to us. We will notify you after processing the form, and we will send a sample PDF agreement for you to review and approve. The appropriate agreements will subsequently be sent to you to review and sign, along with an invoice for the one-time licensing fee.

2. When you return the signed agreements and payment to us, we will countersign and return the executed agreements to you. To help you develop your product, we will provide a development kit with a variety of support materials and procedures, and assign a Dolby engineer to assist you.

   To develop and test your product, you must have the following equipment:
   - Professional soundcard with AES3/IEC 60958 multichannel digital input and output, capable of supporting 44.1 and 48 kHz sampling rates. Please contact Dolby Laboratories to obtain information about recommended soundcards.
   - Digital audio editing program (AEP) capable of recording and playing back .wav or raw PCM files, such as Adobe® Audition® or Sony® Sound Forge®.
   - For a product with an MPEG-2 Transport Stream I/O, a multiplexer and demultiplexer. These are needed for use with the provided test signals.

3. Build and test a preproduction unit, then send us your test data for review. If your test data meets our requirements, we will ask you to send us a unit that we can test ourselves. If the unit meets our requirements, we will approve your product for production and sale, and activate your license.

Note: If governmental approval is required for your license agreement with Dolby Laboratories, we recommend that you preapply for this approval toward the end of the licensing process to prevent delays.

Product Specification for Dolby® Professional Loudness Metering and Dolby Professional Loudness Correction

This product specification is in two parts: Part One discusses Dolby® Professional Loudness Metering, and Part Two discusses Dolby Professional Loudness Correction for System and Implementation Licenses.

Part One: Dolby Professional Loudness Metering

The Dolby Professional Loudness Metering licensing development kit is used to enable Licensees to use the Dolby loudness-estimation algorithm in metering, monitoring, and encoding applications within the broadcast ecosystem. This specification covers both real-time and file-based applications.

These Licensed Products will usually display (onscreen, remote monitoring, or other output) the measured loudness level as well as details relating to true-peak levels and the percentage of speech-based content. This information will be used for reference/off-air monitoring, comparison to other channels, compliance checking for delivery standards, and for setting of a dialogue-normalization value—if this value is set incorrectly in the incoming bitstream—to be used in Dolby E, Dolby Digital, or Dolby Digital Plus file-based encoding or transcoding applications.

The core loudness-estimation library is available as compiled Intel® software libraries for Mac®, Windows®, or Linux® operating systems. By supplying Licensees with precompiled software, Dolby aims to help reduce integration and development times, and to increase the degree of confidence in the Implementation because the core software has been written and tested at Dolby Laboratories. Additionally, the separation of the core algorithm from the sample application facilitates an easy upgrade path as the core algorithm improves over time.

Licensed Products must meet the following terms.

LICENSED PRODUCT FEATURES AND REQUIREMENTS

The loudness-estimation algorithm includes two types of measurements for Leq(A) and ITU-R BS.1770:[1] speech-based (using multichannel Dialogue Intelligence™) and non-speech–based (without multichannel Dialogue Intelligence).

The loudness-estimation library is not meant to be used as part of a workflow that corrects loudness levels within a compressed bitstream; the library should be used only to estimate loudness values for real-time or file-based metering or file-based encoding/transcoding applications. For file-based encoding/transcoding applications, setting a target loudness level and scaling the audio is not recommended with the current deliverables.

Dialogue normalization or compression metadata should not be applied before inputting PCM to the loudness library. Before attempting to measure loudness, the user should set the upstream decoder to Custom mode.

If the application is a real-time application, the loudness-estimation library must be used only for metering applications and not for setting the dialogue-normalization value for real-time encoding or transcoding.

---

[1] ITU-R BS.1770, *Algorithms to Measure Audio Programme Loudness and True-Peak Audio Level*, including the ITU-R BS.1770-2 update.

Tools or test materials provided as part of the deliverables may be used only for internal testing and development of the Licensed Product and not for any other purpose. Tools or test materials may not be used as part of the Licensed Product itself or provided to or used by any third party other than for the test development of the Licensed Product.

Software Libraries

For real-time or file-based applications, the Dolby Professional Loudness Metering software is delivered in the form of a reentrant software library and sample code. The software library must not be modified; however, the sample code that is supplied as part of the licensed software package may be modified, rewritten, or redesigned as needed to suit the system, subject to any limitations contained in the deliverables.

The sample application illustrates how to call the loudness subroutine by using the application programming interface (API). The user interface is via command-line options or the graphical user interface (GUI); loudness parameters are specified on the command line or in the GUI. Although the sample-application code may be used as is for certain platforms (such as a PC or a Unix® platform–based workstation), Licensees may wish to customize certain aspects of the audio I/O and user interface.

Table 1 shows the mean performance figures using a measurement mode of ITU-R BS.1770-2 plus Dialogue Intelligence for a five-minute, 5.1-channel input file. The *Dolby Professional Loudness Metering Code User's Guide for Intel Processors* provides more-detailed performance figures. Also, the user should keep in mind that these performance figures are estimates only and may differ based on the platform, operating system, and state of that system (for example, if other tasks are running and so on). The performance figures are also exclusive of any file I/O or other system functions.

Table 1 Typical Performance with a Five-Minute, 5.1-Channel Input File

Platform	Operating System	CPU	Execution Time (Seconds)	Number of Times Faster than Real Time	Equivalent MHz
Intel IA-32	Windows 7 32-bit	2.93 GHz Intel Core™ i7 with 4GB RAM	3.7	81	36.1
	Mac OS® X 10.6.6	2.0 GHz Intel Core 2 Duo with 2 GB RAM	7.74	39	51.6
	Ubuntu® Linux		8.46	35	56.4
Intel 64	Windows 7 64-bit		7.73	39	51.6
	Mac OS X 10.6.6		7.07	42	47.1
	Ubuntu Linux		7.69	39	51.3

I/O Interfaces and Files

The loudness-estimation library is responsible for estimating the loudness of an audio file or stream. Currently, loudness estimation is performed only with audio in PCM format. Hence, any coded audio source files and/or streams must be decoded prior to calling the process.

The following information is for I/O interfaces and files.

- For real-time applications, AES3 inputs and outputs are highly recommended: AES-3id-2001, AES3-2003, IEC 60958-4 (2003-05), and IEC 60958-3 (2003-01).

- For file-based applications, PCM or WAV file format with embedded metadata is required. Please refer to the *Dolby Audio Metadata WAVE Chunk: An Extension to the WAVE File Format* document for more information. Modification of metadata within the Dolby audio metadata WAVE chunk of a WAV file as a result of using the loudness-estimation library is allowed only in file-based applications.

- If the Licensed Product is a file-based transcoder product, the provided test signals in PCM or WAV format are also supplied as Dolby Digital, Dolby Digital Plus, and Dolby E encoded files. The output encoded file will be checked for proper setting of metadata.

- If the Licensed Product is a file-based transcoder or encoder product, proper setting of the dialogue-normalization value after loudness estimation is required and will be tested for in the compressed output file.

- External timecode support/input is optional (but highly recommended) if the Licensed Product is being designed to work with other systems that require time-stamp data and/or the time-stamp delay word (as per SMPTE 339M-2000[2]), such as MPEG multiplexers.

Dialogue Intelligence Latency

The latency due to the Dialogue Intelligence algorithm is calculated as 2.048 seconds. This latency applies regardless of whether or not Dialogue Intelligence is enabled. This latency must be properly compensated for when integrating the Dolby Professional Loudness Metering library into a system.

Loudness Metering Parameters and User Interface

The Dolby Professional Loudness Metering library has the following requirements and recommendations for loudness metering parameters and user interface.

- The library provides five different loudness metering modes:
  - ITU-R BS.1770-2 plus Dialogue Intelligence (required)
  - ITU-R BS.1770-1 plus Dialogue Intelligence (optional)
  - ITU-R BS.1770-2 (optional)
  - Leq(A) plus Dialogue Intelligence (optional)
  - EBU (optional)[3]

- For each mode, the following parameters may be implemented:[4]
  - Momentary Loudness (optional)
  - Short Term 10 Sec Loudness (required for real-time products, optional for file-based products)
  - Integrated Loudness (required)

    - Gating technique used: speech, level, or no gating
    - Speech percentage (unless in ITU-R BS.1770-2–only mode)
    - Dialogue-normalization value
  - Speech threshold (required)
  - Sample peak (required)
  - True peak (required)

    - True-peak DC block on/off
    - True-peak emphasis on/off
- Logging of the results is required.

- The units may be LKFS, LUFS, LU, or dBFS for Leq(A), based on the measurement. The sample user interface provided uses LKFS for ITU modes.

- The version of the loudness library used must be shown on the user interface or in the "About" box of the product.

---

[2]   SMPTE 339M-2000, *Format for Non-PCM Audio and Data in AES 3—Generic Data Types.*

[3]   Complies with EBU Tech 3341, *Loudness Metering: "EBU Mode" Metering to Supplement Loudness Normalisation in Accordance with EBU R 128.*

[4]   Please see the development manual for additional requirement information.

- For metering products, adherence to the requirements outlined in EBU Tech 3341 is recommended.[5]
  - A reset control for real-time measurement products is required.

Sample Rate

All loudness-estimation products must support 48 kHz. Support for 32 kHz and 44.1 kHz is optional.

Bit Depth

Bit depths of 16, 20, and 24 bits are required to be supported, with a default of 24 bits.

OEM Development Kits

If the Licensed Product is to be an OEM development kit, such as a software library, the Licensee will act as an implementer and may sell such Licensed Products only to Qualified Recipients that Dolby has licensed and authorized to receive such Implementations. An Implementation Licensee may make an API to enable Qualified Recipients to wrap a real-time system around the core loudness library. The Qualified Recipient must have the software development kit integrated into an application so that its functions are controlled strictly from that application and not from a system designed around it or other applications.

Software Plug-Ins

If the Licensee is building a software plug-in that will be part of a host application, the Licensee will need a System and Implementation License. Both the Licensee and Dolby will test the software plug-in operating within the host application.

Part Two: Dolby Professional Loudness Correction: System and Implementation

Products implementing Dolby Professional Loudness Correction are mainly used for file-based loudness correction of audio. The Dolby Professional Loudness Correction library (DPLC) accepts a Dolby Digital (.ac3), Dolby Digital Plus (.ec3), Dolby E (.dde), or WAV (.wav) file as an input; and it outputs the same bitstream format as the input. DPLC operates as a two-pass process; therefore, it is intended for file-based operation only.

The core loudness correction algorithm is implemented using compiled software libraries for Mac, Windows, or Linux systems using Intel x86. Highlights of the loudness measurement and correction functions included in the library are:

- Loudness metering based on ITU-R BS.1770-2 (optional, A-weighted)
- Dolby Dialogue Intelligence as a speech-gating function, per the Dolby DP600 Program Optimizer
- Level-gating function defined by ITU-R BS.1770-2
- True peak (per channel), based on ITU-R BS.1770-2

File-based loudness correction functionality is as follows:

- Update of dialnorm, dynamic range control (DRC), and cyclic redundancy check (CRC) parameters for Dolby Digital, Dolby Digital Plus, and Dolby E bitstreams
- Normalization (PCM rescaling) and sample/true-peak limiting for Dolby Digital, Dolby Digital Plus, Dolby E, and WAV files

---

[5]   EBU Tech 3341, *Loudness Metering: "EBU Mode" Metering to Supplement Loudness Normalisation in Accordance with EBU R 128.*

Real-time functionality is as follows:

- Real-time update of the dialnorm, DRC, and CRC bitstream parameters of Dolby Digital and Dolby Digital Plus bitstreams, from a previously available loudness measurement

Tools or test materials that are provided as part of the deliverables may not be used as part of the Licensed Product itself.

Software Libraries

By supplying Licensees with precompiled software, Dolby aims to help reduce integration and development times, and to increase the degree of confidence in the Implementation because the core software has been written and tested at Dolby Laboratories. Additionally, the separation of the core algorithm from the sample application facilitates an easy upgrade path as the core algorithm improves over time.

The core subroutine is a relocatable object library file and may not be changed. The system (Licensed Product) developer links the subroutine library with the developer's own customized code. Special consideration must be given to the subroutine-section locations and memory-resource usage.

The sample-application source code that is supplied as part of the licensed software package may be modified, rewritten, or redesigned as needed to suit the system, subject to any limitations contained in the deliverables.

The sample application illustrates how to call the subroutine using the application programming interface (API). The user interface is via command-line options or the graphical user interface (GUI); DPLC parameters are specified on the command line or the GUI. Although the sample code may be used as is for certain platforms (such as a PC or a UNIX platform–based workstation), Licensees may wish to customize certain aspects of the audio input/output and user interface.

The DPLC software is delivered in the form of a software library for Mac, Windows, or Linux systems using Intel x86. The software library code deliverable is divided into two layers: subroutine and sample code. The DPLC subroutine and sample code that are included with this software package may be readily compiled to create a DPLC-executable program.

The three modes of loudness correction in the DPLC are loudness normalization, bitstream loudness correction, and dialnorm update.

- Loudness normalization: Network delivery specifications sometimes call for audio to be normalized at a certain loudness, and with a certain maximum sample/true-peak level (for example, –24 LKFS with a maximum peak of –2 dBTP). Loudness normalization is the technique of scaling PCM audio to achieve a certain loudness and maximum peak specification. This two-pass process involves decoding the incoming bitstream and measuring the loudness, then scaling the PCM, and reencoding with the proper dialnorm value and a predicted DRC profile back to the original bitstream format. A WAV file input is also possible.
- Bitstream loudness correction: *Bitstream loudness correction* is the updating of the dialnorm bitstream parameter to reflect the actual program loudness. This two-pass process involves decoding the incoming bistream and measuring the loudness, but then modifying metadata directly within the bitstream. Modifying the dialnorm value also requires that the following bitstream parameters be updated:
  - DRC words (compr and dynrng) are invalidated when dialnorm is modified, and they must be recalculated and updated.
  - CRC words are invalidated when dialnorm is modified, and they must be recalculated and updated.

- Dialnorm update: Occasionally, it is necessary to update a Dolby Digital or Dolby Digital Plus stream with a new dialnorm value. This occurs when the loudness of the content has been previously measured or is otherwise known a priori. The dialnorm-update function inserts a new dialnorm-parameter value into a Dolby Digital or Dolby Digital Plus bitstream. This is a one-pass or real-time process. If file-based correction is also being implemented in a workflow, real-time dialnorm updating should take place upstream from the file-based correction.

Profiling was achieved by measuring the time required for the file-based command-line interface (CLI) sample application to process a two-minute file. The application was configured as follows:

- Loudness metering mode: ITU-R BS.1770-2 plus Dialogue Intelligence
- Speech threshold: 20 percent
- True-peak DC block: Off
- True-peak emphasis: Off
- Stereo and 5.1 inputs

Complexity profiling was performed on a number of different platforms. Regardless of the platform, the application used only a single core.

Windows IA-32 profiling was performed on a 2.93 GHz Intel Core i7 processor-based PC running 32-bit Windows 7 and equipped with 4 GB of RAM. The results are listed in Table 2.

Table 2 DPLC Performance on Windows IA-32

Input Program	Function	Number of Times Faster than Real Time	Equivalent MHz
Stereo Dolby Digital	Bitstream loudness correction	36.2	80.9
Stereo Dolby Digital Plus		37.1	79.1
Stereo Dolby E		13	225.8
Stereo Dolby Digital	Loudness normalization	23.3	125.6
Stereo Dolby Digital Plus		17.4	168.6
Stereo Dolby E		8.4	348.7
Stereo .wav		30.6	95.7
5.1 Dolby Digital	Bitstream loudness correction	25.9	113.3
5.1 Dolby Digital Plus		26.4	111
5.1 Dolby E		13.1	223.4
5.1 Dolby Digital	Loudness normalization	14.5	202.5
5.1 Dolby Digital Plus		10.4	280.9
5.1 Dolby E		8.2	356.6
5.1 .wav		10	291.9
Stereo Dolby Digital	dialnorm update	90.5	32.4
Stereo Dolby Digital Plus		91.9	31.9
5.1 Dolby Digital		77.7	37.7
5.1 Dolby Digital Plus		79.1	37.1

Licensed Product Requirements

Licensed Products must be developed in accordance with the terms in Tables 3 through 19.

Table 3 General Requirements and Recommendations

	Category	Specification
1.	Logos and Trademarks	Complies with the guidelines in the appropriate Dolby logos and trademarks guide. Only the logos and markings specified there are acceptable for inclusion.
2.	Core Processing	The library object code delivered with the kit cannot be modified.
3.	Outputs	The Licensed Product is recommended to support Dolby Digital, Dolby Digital Plus, Dolby E, and .wav file output.
5.	Outputs	Logging of measurement and correction results is required.
6.	Inputs	The Licensed Product is recommended to support Dolby Digital, Dolby Digital Plus, Dolby E, and .wav file input.
7.	Inputs	Products that encode broadcast .wav files must support the Dolby audio metadata chunk.
8.	Inputs	Dolby E input bitstreams to DPLC must not be SMPTE337 formatted. The Licensee's product must handle deformatting and reformatting for SMPTE 337. Also, each Dolby E program must contain Dolby Digital metadata. The maximum Dolby E frame rate that is supported is 30 fps. The supported frame bit depth must be 16-bit or 20-bit.
9.	Inputs	For the .wav file input support, the file must contain 1–6 channels; must be 16, 20, or 24 bit; must support 32, 44.1, or 48 kHz input; and must contain unencoded PCM data.
10.	Inputs/Outputs	Only 48 kHz Dolby Digital Plus bitstreams are supported by DPLC.
11.	Inputs/Outputs	Only independent bitstreams are supported by the Dolby Digital Plus input/output (no dependent bitstreams, and 5.1 channel maximum). Also, audio description carried in a transport stream as single or dual PID is not supported by DPLC at this time.
12.	Inputs/Outputs	For products such as video transcoders that incorporate DPLC, latency is important as it may introduce A/V sync issues. Designers must be aware of the latency behavior of DPLC and must properly account for it to avoid A/V sync problems. Please refer to the *Dolby Professional Loudness Correction Development Manual* for more detail.
13.	Inputs/Outputs	When integrating DPLC into products that use transport stream input/output, special consideration must be taken to properly set descriptors for Dolby Digital and Dolby Digital Plus. Please refer to the *Dolby Professional Loudness Correction Development Manual* for more detail.
14.	User Interface	Sets the documented default values set for all control parameters.

File-Based Loudness Correction Requirements

If implementing file-based loudness correction, the requirements in Tables 4 through 16 apply.

Table 4 Parameter Requirements for File-Based Loudness Correction

Group	Parameter	Requirement
Configuration parameters	Loudness metering mode Speech threshold True-peak DC block True-peak emphasis Loudness correction mode RF mode DRC profile Line mode DRC profile Correction threshold Normalization target Normalization peak limit Normalization peak reference	Required
Result parameters	Logging	

Table 5 Loudness Metering Mode

Parameter	Mode Settings	Default
Loudness metering mode **(R)**	ITU-R BS.1770-2 plus Dialogue Intelligence **(R)** ITU-R BS.1770-1 plus Dialogue Intelligence **(O)** ITU-R BS.1770-2 **(O)** Leq(A) plus Dialogue Intelligence **(O)**	ITU-R BS.1770-2 plus Dialogue Intelligence

R = Required parameter or mode setting
O = Optional setting

Table 6 Speech Threshold

Parameter	Setting Range Values	Default
Speech threshold **(R)**	0 to 100%, in 1% increments	20%

R = Required parameter

Table 7 True-Peak DC Block

Parameter	Settings	Default
True-peak DC block **(R)**	Enabled (on); disabled (off)	Off

R = Required parameter

Table 8 True-Peak Emphasis

Parameter	Settings	Default
True-peak emphasis **(R)**	Enabled (on), or disabled (off)	Off

R = Required parameter

Table 9 Loudness Correction Mode

Parameter	Loudness Correction Mode Settings	Default
Loudness correction mode **(R)**	Bitstream Loudness Correction **(R)** Loudness Normalization **(R)**	Bitstream loudness correction

R = Required parameter or mode setting

The loudness correction mode input parameter specifies the operating mode of DPLC.

Note: When a product is operating on a .wav file, bitstream loudness correction cannot occur. In this case, the bitstream loudness correction option must be disabled, and the loudness normalization option made the default.

Table 10 Line Mode and RF Mode: DRC Profiles

Parameter	Profile Settings	Default
Line mode DRC profile **(R)** RF mode DRC profile **(R)**	None **(R)** Film Standard **(R)** Film Light **(R)** Music Standard **(R)** Music Light **(R)** Speech **(R)** Predict from input **(R)**	Predict from input

R = Required parameter or setting

Table 11 Correction Threshold

Parameter	Setting Values	Default Setting
Correction threshold **(R)**	0 to 8 dB, in 1 dB increments or finer	1 dB

R = Required parameter

Table 12 Normalization Target

Parameter	Setting Values	Default Setting
Normalization target **(R)**	–31 to–8 LKFS/LUFS/dBFS, in 1 dB increments or finer	–24 LKFS/LUFS/dBFS

R = Required parameter

Note: The normalization target parameter is relevant only when the mode is set to Loudness Normalization. This parameter must be disabled or hidden from the user when the mode is set to Bitstream Loudness Correction.

Table 13 Normalization Peak Limit

Parameter	Setting Value	Default Setting
Normalization peak limit **(R)**	–6 to–0.1 dBTP/dBFS,in 1 dB increments or finer	0 dBTP/dBFS

R = Required parameter

Note: The normalization peak limit parameter is relevant only when the mode is set to Loudness Normalization. This parameter must be disabled or hidden from the user when the mode is set to Bitstream Loudness Correction.

Table 14 Normalization Peak Limit Reference

Parameter	Setting Values	Default Settings
Normalization peak limit reference **(R)**	True peak **(R)** Sample peak **(R)**	True peak

R = Required parameter or setting

Note: The normalization peak limit reference parameter is relevant only when the mode is set to Loudness Normalization. This parameter must be disabled or hidden from the user when the mode is set to Bitstream Loudness Correction.

Table 15 File-Based Loudness Correction Logging

Log Parameter	Description	Rate
Number of streams **(R)**	Products must report the number of streams found in each file by DPLC.	Per file
Number of programs **(R)**	Products must report the number of programs found in each stream by DPLC.	Per stream
CRC counts **(R)**	Products must report the number of CRC checks found (passed and failed) in each stream.	
Applied mode **(R)**	Output: Products must report which loudness correction mode was applied by DPLC: loudness normalization, bitstream loudness correction, or dialnorm update.	
Momentary loudness (maximum) **(O)**	Input: The maximum momentary loudness as corrected by DPLC. –70 to 10 LKFS/dBFS. (Resolution is not defined.) dBFS is used only when DPLM is configured for Leq(A) plus Dialogue Intelligence.	Per program
Short-term 3 s loudness (maximum) **(O)**	Input: The maximum short-term 3 s loudness as corrected by DPLC. Displayed as an absolute loudness or a relative loudness (relative to the EBU target level of –23 LUFS): • Absolute loudness (LUFS, with 0.1 resolution) • Relative loudness (LU, with 0.1 resolution)	
Short-term 10 s loudness (maximum) **(O)**	Input: The maximum short-term 10 s loudness as corrected by DPLC. –70 to 10 LKFS/dBFS with a resolution of 0.1. dBFS is used only when DPLM is configured for Leq(A) plus Dialogue Intelligence.	
Integrated loudness **(R)**	Input: The maximum integrated loudness as calculated by DPLC. –70 to 10 LKFS/dBFS with a resolution of 0.1. dBFS is used only when DPLM is configured for Leq(A) plus Dialogue Intelligence.	
Integrated loudness gating **(R)**	Input: The gating technique invoked by DPLM must be indicated in the DPLC log output: speech gating, level gating, or no gating.	
Integrated speech percentage **(R)**	Input: The speech percentage corrected by DPLC must be displayed if Dialogue Intelligence is enabled. (Refer to DPLM development manual.) The range is 0 to 100%, 1% resolution.	
Integrated loudness range **(O)**	Input: The loudness range display as used by DPLC (0 to 50 LU).	
Sample peak **(R)**	Input: This result contains the largest absolute sample value in any channel since DPLC was reset.	
True peak **(R)**	Input: This result contains the largest absolute true-peak value in any channel since DPLC was reset.	
Input and output dialnorm **(R)**	Input and output: Products must report the dialnorm of the input and output (correction) bitstream. The dialnorm must be presented as a value from 1 to 31, or as Not Present.	

Input and output DRC profiles:
  Line mode **(R)**
  RF mode **(R)**
	Input and output: Products must report both the Line mode and RF mode DRC profiles of the input and output bitstream: None, Film Standard, Film Light, Music Standard, Music Light, Speech, or Not Present.	Per program

Correction (indicator only) **(R)**	Output: Correction indicates the section of parameter values that are DPLC output results.	
Gain **(R)**	Output: Products are required to report the gain actually applied to the program, when performing loudness normalization. (There is no requirement when performing bitstream loudness correction.) The gain must be presented as a numeric figure in dB, with a minimum resolution of 0.1 dB.	
Peak limit **(R)**	Output: Products are required to report the peak limit, in dBFS, actually applied to the audio when the professional limiter is enabled. (There is no requirement when the professional limiter is not enabled.) The peak limit must be presented as a numeric figure in dBFS, with a minimum resolution of 0.1 dB.	
Warnings **(R)**	Please see the Log Warnings Returned by Loudness Correction table (Table 16).	

R = Required parameter in the output log

O = Optional parameter in the output log

Note: Some warnings are returned on a per-stream basis, while others are returned on a per-frame basis. Warnings must be displayed on the basis they are returned.

Table 16. Log Warnings Returned by Loudness Correction

Return Warnings (All Required)	Definition	Applies To
No bitstream loudness correction	Bitstream loudness correction could not be applied. Loudness normalization applied instead.	Stream
Additional information	The input stream contains additional information.	
Could not measure loudness	Program loudness could not be measured. Program has not been corrected.	Program
Program too quiet	The program was too quiet and required more than 30 dB of gain. The program has not been corrected.	
Program too loud	The program was too loud and required less than –30 dB of gain. The program has not been corrected.	
Professional limiter enabled	The professional limiter was enabled.	
Program not loudness corrected	The program was not loudness corrected.	
Program loudness and dialnorm correct	Both the loudness and dialnorm values of the input program are correct.	

Dialnorm Update Requirements

If implementing dialnorm updating, the requirements in Tables 17 through 19 apply.

Table 17 Dialnorm Update User-Interface Parameter Requirements

Parameter	User-Interface Settings and Description	Default Setting

dialnorm update mode **(R)**	The two types of dialnorm update:
	   Absolute mode **(R)**
	   Relative mode **(R)**
		Absolute mode

dialnorm value **(R)** (main program)	In Absolute mode, this parameter selects the dialnorm value (1–31) to write into the output bitstream.	24
	In Relative mode, this parameter selects the offset to the input dialnorm value for use in calculating the output value (–30 to 30).	0

Line mode DRC profile **(R)**
RF mode DRC profile **(R)**
(main program)
	These two parameters set the DRC profile to use for calculating new DRC words to be written into the output bitstream. The following are the profiles for both Line and RF modes:
	   None
	   Film Standard
	   Film Light
	   Music Standard
	   Music Light
	   Speech
		Film Standard

dialnorm2 value **(R)** (dual mono, program 2)	This parameter relates to the second program of a dual-mono bitstream (acmod equal to 0). It is ignored by DPLC for non-dual-mono bitstreams. For settings, refer to dialnorm value (main program).	
Line mode DRC profile2 **(R)** RF mode DRC profile2 **(R)** (dual mono, program 2)	Both these parameters relate to the second program of a dual-mono bitstream (acmod equal to 0), They are ignored by DPLC for non-dual-mono bitstreams. For settings, refer to the Line and RF modes, DRC profile (main program).	
Logging **(R)**	Please see the Dialnorm Update Logging table (Table 18).	

R = Required parameter or setting

Note: The dialnorm update DPLC function is required to accommodate both Dolby Digital (.ac3) and Dolby Digital Plus (.ec3) bitstreams at the input.

Table 18 Dialnorm Update Logging

Log Parameter	Description
dialnorm value **(R)** dialnorm2 value **(R)** (input/output)	Products must report the following: •dialnorm/dialnorm2 parameter values for both input/output as reported by DPLC •Any change in the dialnorm/dialnorm2 values with a time indication (for example, timecode, frame number, and so on)
Warnings **(R)**	Products must report each warning returned by DPLC. Refer to Log Warnings for Dialnorm Update table (Table 19).
Sync state **(R)**	Products must report the sync state (for example, whether or not DPLC is detecting valid frames).
Bitstream information **(O)**	Products may report the acmod and lfeon bitstream parameters as reported by DPLC.

R = Required parameter for output log

O = Optional parameter for output log

Table 19 Log Warnings for Dialnorm Update

Log Warnings (All Required)	Description
Partial frame	A partial frame was detected in the bitstream.
Unsupported frame	An unsupported frame was detected.
CRC error	A CRC error was detected.
Input dialnorm changed	The input dialnorm changed.
Output dialnorm saturated	The output dialnorm is saturated.
Additional information	The input stream contains additional information.

General Licensing Requirements

OEM DEVELOPMENT KITS

If the Licensed Product is to be an OEM development kit, either as a board or software library, the Licensee will act as an implementer and may sell such Licensed Products only to Qualified Recipients that Dolby has licensed and authorized to receive such Implementations. An Implementation Licensee may make an API to enable Qualified Recipients to wrap a streaming or a real-time system around the core loudness library (for real-time dialnorm updating only). The Qualified Recipient must have the software development kit integrated into an application so that its functions are controlled strictly from that application, and not from a system designed around it or other applications.

SOFTWARE PLUG-INS

If the Licensee is building a software plug-in that will be part of a host application, the Licensee will need a System and Implementation License. Both the Licensee and Dolby will test the software plug-in operating within the host application.

FOR MORE INFORMATION

Please see the *Dolby Professional Loudness Metering Development Manual* and the *Dolby Professional Loudness Correction Development Manual* for more details.

APPLICATIONS NOT LISTED IN DOCUMENT

If your application is any other than those listed in this document, please contact Dolby Laboratories at prolicensing@dolby.com to confirm that your application is covered under the license agreement.

Issue 2.0

The Intel C++ compiler and integrated performance primitives (IPP) were used in the development of this product (www.intel.com/software/products).

Appendix

DOLBY PROFESSIONAL PRODUCTS

Dolby Laboratories manufactures a variety of professional products ideally suited for broadcast, DVD mastering, and other applications.

Dolby DP569 Multichannel Dolby Digital Encoder

The DP569 encodes up to 5.1 channels for DTV transmissions and DVD mastering applications.

Dolby DP563 Dolby Surround and Dolby Pro Logic II Encoder

The DP563 encodes up to five-channel program material within the digital domain into Left total/Right total (Lt/Rt) output for two-channel analog or digital applications.

Dolby DP564 Multichannel Audio Decoder

The DP564 decodes and monitors programs with Dolby® Digital (up to 5.1 channels), Dolby Surround Pro Logic®, Dolby Pro Logic II, Dolby Digital Surround EX™, and PCM soundtracks.

Dolby DP568 Professional Reference Decoder

The DP568 decodes and monitors programs with Dolby Digital Plus, HE AAC with Dolby metadata, Dolby Pro Logic II, and Dolby Digital soundtracks.

Dolby DP571 Dolby E Encoder

The DP571 encodes up to eight channels of digital audio and metadata for use in DTV program production and broadcast distribution.

Dolby DP572 Dolby E Decoder

The DP572 decodes Dolby E bitstreams for use in DTV program production and broadcast distribution infrastructure.

Dolby DP570 Multichannel Audio Tool

The DP570 is used with Dolby Digital and Dolby E codecs for DVD and DTV surround sound production, metadata authoring, and confidence monitoring.

Dolby DP583 Frame Synchronizer

The DP583 syncs Dolby Digital and Dolby E bitstreams from asynchronous sources to local reference, and reframes Dolby E streams stored on devices without reference capabilities.

Dolby LM100 Broadcast Loudness Meter

The LM100 Broadcast Loudness Meter measures the subjective loudness of dialogue within broadcast programming and assists authoring metadata in Dolby Digital and Dolby E bitstreams for consistent playback loudness from program to program.

Dolby DP600 Program Optimizer

The DP600 Program Optimizer enables intelligent audio analysis and automated loudness normalization for common broadcast media file and audio formats, as well as Dolby E, Dolby Digital Plus, HE AAC with Dolby metadata, and Dolby Digital encoding, decoding, and transcoding.

Dolby Media Producer Software Suite

Dolby Media Producer is a suite of professional software that supports all Dolby audio codecs used in mastering DVD-Video and Blu-ray Disc™ formats. It supports multichannel non-real-time encoding and file decoding of Dolby Digital Plus, Dolby Digital, Dolby TrueHD, and MLP Lossless™ formats via an intuitive user interface. The suite comprises six separate software products, including Dolby Media Encoder, Dolby Media Decoder, Dolby Media Tools, and Dolby Media Meter.

Dolby Media Emulator

Dolby Media Emulator is a software tool that provides metadata authoring and emulation for Dolby broadcast audio formats. Paired with popular digital audio workstations, Dolby Media Emulator enables you to emulate how your mix will sound when it's played back through a TV or set-top box. Dolby Media Emulator also simplifies the monitoring of metadata for use with Dolby Digital Plus, Dolby Digital, Dolby E, and High-Availability AAC (HE AAC).

DOLBY OEM MODULES

With Dolby OEM modules, you can easily and cost-effectively incorporate Dolby E and/or Dolby Digital Plus technology into your products without having to develop your own Implementations. As the modules are fully assembled and tested by Dolby Laboratories, there are no licensing or royalty fees involved. However, to ensure the interoperability and compatibility of your products incorporating Dolby modules, we reserve the right to test and approve them before you make them available for sale.

Except for the Cat. No. 561, Dolby OEM modules measure 116 x 51 mm (4.55 x 2 inches) and provide a 72-pin edge connector per the JEDEC MO-116 SIMM specification. They can be controlled via control pins or a serial control port.

Cat. No. 561 Dolby Digital Plus Encoder Module

The Cat. No. 561 is capable of simultaneously encoding multiple channels of audio into Dolby Digital (AC-3), Dolby Digital Plus, and HE AAC with Dolby metadata. It accepts PCM, Dolby E, Dolby Digital, and Dolby Digital Plus data in bit-serial (I²S) format, and produces up to four encoded bitstream outputs, which can consist of multiple programs. It also supports transcoding Dolby Digital to Dolby Digital Plus bitstreams; Dolby E to Dolby Digital Plus, Dolby E to HE AAC with Dolby metadata, and Dolby E metadata parsing and decoding; and both serial port and GPI/O control.

Cat. No. 552 Multichannel Dolby E and Dolby Digital Plus Decoder Module

The Cat. No. 552 decodes Dolby E bitstreams with up to eight channels and Dolby Digital Plus bitstreams with up to six channels into separate output channels. It can also downmix multichannel Dolby E and Dolby Digital Plus bitstreams into a stereo or Lt/Rt two-channel output for monitoring purposes. It also passes PCM audio straight through, or applies Dolby Pro Logic decoding to derive L, C, R, and S outputs from two-channel PCM programs. The Cat. No. 552 features two metadata outputs: one provides a full metadata set for use with downstream Dolby E or Dolby Digital Plus encoders, while the other provides a metadata subset designed for OEM use in the display of key metadata parameters.

Cat. No. 559D Dolby Digital Multichannel Encoder Module

The Cat. No. 559D provides up to 5.1-channel Dolby Digital encoding with bit rates from 56 to 640 kbps and sampling rates between 32 and 48 kHz. Metadata can be internally generated or received from externally generated sources, and a metadata subset output is available for the display of key metadata values.

Cat. No. 559E Dolby E Encoder Module

The Cat. No. 559E provides up to eight channels of Dolby E encoding and supports up to 20-bit word lengths at a 48 kHz sampling rate. The Cat. No. 559E can be controlled via control pins or a serial control port, and can pass through PCM audio signals. Externally generated metadata can be accepted into the module, or metadata can be internally generated. A metadata subset output is available for OEM use in the display of key metadata values.

For more information on incorporating Dolby E in your products, email prolicensing@dolby.com.

CORPORATE HEADQUARTERS

Dolby Laboratories, Inc.
Dolby Laboratories Licensing Corporation
100 Potrero Avenue
San Francisco, CA 94103-4813 USA
T +1-415-558-0200
F +1-415-645-4000
dolby.com

EUROPEAN LICENSING LIAISON OFFICES

Dolby Europe Licensing Limited
Royal Wootton Bassett
Wiltshire SN4 8QJ United Kingdom
T +44-1793-842100
F +44-1793-842101
dolby.co.uk

Dolby Germany GmbH
Deutschherrnstrasse 15–19
90429 Nürnberg, Germany
T +49-911-928-91-0
F +49-911-928-91-99

ASIA OFFICES

Dolby Japan K.K.
NBF Higashi-Ginza Square 3F
13–14 Tsukiji 1-Chome, Chuo-ku
Tokyo 104-0045 Japan
T +81-3-3524-7300
F +81-3-3524-7389
dolby.co.jp

Dolby Laboratories International Services
(Beijing) Co., Ltd.
Rooms 907–916, Level 9, West Building
World Financial Center
No. 1, East 3rd Ring Middle Road, Chaoyang District
Beijing 100020 China
T +86-10-5910-3000
F +86-10-5910-3001
dolby.com.cn