# Dolby® Digital and Dolby Digital Plus Bitstream Metadata Parameters Guide

Issue 3

*Confidential Information*

# Dolby Laboratories Licensing Corporation

## Corporate Headquarters

**Dolby Laboratories, Inc.**
**Dolby Laboratories Licensing Corporation**
100 Potrero Avenue
San Francisco, CA  94103-4813  USA
**Telephone** 415-558-0200
**Fax** 415-863-1373
www.dolby.com

## European Licensing Liaison Office

**Dolby Laboratories, Inc.**
Wootton Bassett
Wiltshire  SN4 8QJ  England
**Telephone** 44-1793-842100
**Fax** 44-1793-842101

## Asia

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



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# Table of Contents

# List of Figures

# List of Tables

# Introduction

This document describes the bitstream parameters, or metadata, that are carried in Dolby® Digital and Dolby Digital Plus bitstreams. Metadata provides unprecedented capability for content producers to deliver the highest quality audio to consumers in a range of listening environments. It also provides choices that allow consumers to adjust their settings to best suit their listening environments.

In this document, we first discuss the concept of metadata. We then discuss the factors controlled by metadata that most directly affect the consumer's experience:

- Dialogue level
- Dynamic range control (DRC)
- Downmixing
- Operational modes

Finally, all the parameters will be defined. Separate sections are included that cover the extended bitstream parameters and new parameters introduced by Dolby Digital Plus.

All the parameters described in this document are originally defined in the ATSC document A/52B, *Digital Audio Compression Standard (AC-3, E-AC-3)*, Revision B, available for download from www.atsc.org.

## 1.1 Metadata Overview

Dolby Digital and Dolby Digital Plus are both data-rate reduction codec technologies that use metadata. Metadata is carried within a Dolby Digital or Dolby Digital Plus bitstream, describing the encoded audio and conveying information to control parameter settings of downstream decoders. In normal operation, the encoded audio and metadata are carried together as a data stream on two regular digital audio channels (AES3 or IEC 60958). Metadata allows content providers unprecedented control over how original program material is reproduced in the home.

Dolby Digital is a transmission bitstream (sometimes called an emission bitstream) intended for delivery to the consumer at home through a medium such as DTV or DVD. It consists of a single encoded program of up to six channels of audio described by one metadata stream. The consumer's Dolby Digital decoder reproduces the program audio according to the metadata parameters set by the program creator and according to settings for speaker configuration, bass management, and dynamic range that are chosen by the consumer to match his or her specific home theater equipment and environmental conditions.

Dolby Digital Plus is a new coding system based on Dolby Digital coding technology. Dolby Digital Plus supports all the functionality of Dolby Digital but provides additional coding tools and a more sophisticated program carriage. Any reference to the functionality provided by Dolby Digital in this document also applies to Dolby Digital Plus.

All metadata parameters for Dolby Digital and Dolby Digital Plus bitstreams should be set by the content producer. In the case of a live broadcast, these parameters are typically carried using Dolby E and then incorporated in the Dolby Digital or Dolby Digital Plus bitstream before transmission. This system allows the program mixer in the broadcast truck to control the way the content is delivered in the consumer's home. Metadata parameters carried in the Dolby E bitstream that control the Dolby Digital encoding process but that are not explicitly carried in the Dolby Digital bitstream are not defined here. These extra parameters typically control filtering and other preprocessing.

## 1.2 Abbreviations Used

Table 1-1 defines the abbreviations used throughout this document.

**Table 1-1** Abbreviations

| Abbreviation | Definition |
|---|---|
| A/V | Audio/video |
| BSI | Bitstream information |
| dBFS | dB with respect to digital full scale |
| DRC | Dynamic range control |
| DTV | Digital television |
| Lo/Ro | Left only/Right only: A downmix from a multichannel to a two-channel output that is compatible for stereo or mono reproduction |
| Lt/Rt | Left total/Right total: A downmix from a multichannel to a two-channel output that is Dolby Surround compatible |
| SPL | Sound pressure level |
| UI | User interface |

Table 1-2 lists abbreviations for channel names used throughout this document.

**Table 1-2** Channel Abbreviations

| Abbreviation | Channel Name |
|---|---|
| L | Left |
| C | Center |
| R | Right |
| Ls | Left Surround |
| Rs | Right Surround |
| LFE | Low-Frequency Effects |

## 1.3 Bitstream Information Parameters

The bitstream parameters described in this document are listed by group in Table 1-3, Table 1-4, and Table 1-5. Definitions for all parameters are listed in Chapter 6.

### 1.3.1 Dolby Digital Bitstream Parameters

Dolby Digital bitstream parameters are listed in Table 1-3. (Refer to Section 6.1 for definitions.)

**Note:** For Table 1-3, Table 1-4, and Table 1-5, the code word for each parameter is defined in the A/52B standard.

**Table 1-3** Dolby Digital Bitstream Parameters

| Dolby Digital Bitstream Parameter | Code Name |
|---|---|
| Audio coding mode | `acmod` |
| Audio production information exists | `audprodie` |
| Bitstream identification | `bsid` |
| Bitstream mode | `bsmod` |
| C downmix level | `cmixlev` |
| Copyright bit | `copyrightb` |
| Dialogue normalization | `dialnorm` |
| Dolby Surround mode | `dsurmod` |
| Extended bitstream information exists | `xbsie` |
| LFE channel | `lfeon` |
| Line mode compression words | `dynrng` |
| Mixing level | `mixlev` |
| Original bitstream | `origbs` |
| RF mode compression words | `compr` |
| Room type | `roomtyp` |
| Surround downmix level | `surmixlev` |

### 1.3.2 Dolby Digital Extended Bitstream Parameters

Dolby Digital extended bitstream information parameters are listed in Table 1-4. (Refer to Section 6.2 for definitions.)

**Table 1-4** Dolby Digital Extended Bitstream Information Parameters

| Dolby Digital Extended Bitstream Parameter | Code Name |
|---|---|
| A/D converter type | adconvtyp |
| Dolby Digital Surround EX™ mode | dsurexmod |
| Lo/Ro C downmix level | lorocmixlev |
| Lo/Ro surround downmix level | lorosurmixlev |
| Lt/Rt C downmix level | ltrtcmixlev |
| Lt/Rt surround downmix level | ltrtsurmixlev |
| Preferred stereo downmix mode | dmixmod |

### 1.3.3 Dolby Digital Plus Bitstream Parameters

Dolby Digital Plus bitstream parameters are listed in Table 1-5. (Refer to Section 6.3 for definition of the parameters unique to Dolby Digital Plus.)

**Table 1-5** Dolby Digital Plus Bitstream Parameters

| Dolby Digital Plus Bitstream Parameter | Code Name |
|---|---|
| A/D converter type | adconvtyp |
| Audio coding mode | acmod |
| Bitstream identification | bsid |
| Bitstream mode | bsmod |
| Bitstream type | strmtyp* |
| C downmix level | cmixlev |
| Copyright bit | copyrightb |
| Dialogue normalization | dialnorm |
| Dolby Surround mode | dsurmod |
| LFE channel | lfeon |
| LFE downmix level code | lfemixlevcod* |
| LFE downmix level code exists | lfelevcode* |
| Line mode compression words | dynrng |
| Lo/Ro C downmix level | lorocmixlev |
| Lo/Ro surround downmix level | lorosurmixlev |
| Lt/Rt C downmix level | ltrtcmixlev |
| Lt/Rt surround downmix level | ltrtsurmixlev |
| Mixing level | mixlev |
| Original bitstream | origbs |
| Preferred stereo downmix mode | dmixmod |
| RF mode compression words | compr |
| Room type | roomtyp |
| Source sample rate code | sourcefscod* |
| Substream ID | substreamid* |
| Surround downmix level | surmixlev |

* Indicates a unique Dolby Digital Plus parameter (not included with Dolby Digital)

# Dialogue Normalization

When audio from different programs is reproduced, the apparent loudness often varies from program to program, as illustrated in Figure 2-1. The horizontal marker represents the average program loudness level, and the lightly shaded area represents the maximum signal headroom available above that point. Because the loudness of these various programs is often different, the consumer must typically readjust the listening volume each time the program changes. For example, the volume may need to be readjusted when switching between different broadcast channels or when a broadcast transitions from the main program to a commercial.

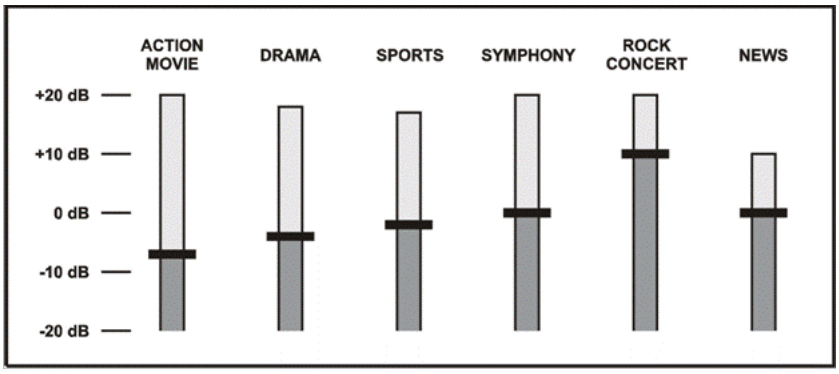


**Figure 2-1** Typical Audio Program Levels (0 dB = 200 $mV_{RMS}$)

Dolby® Digital eliminates the need to continuously readjust the volume by providing a feature known as dialogue normalization. The dialogue normalization parameter (`dialnorm`) represents a known dialogue reference level that is sent with each audio program delivered using Dolby Digital. The process of dialogue normalization aligns the dialogue reference level to a specific reference output level, thus ensuring that the dialogue level of any program is played back at the same output level. This allows all programs to be reproduced at the same subjective loudness and avoid large changes in loudness between different sources and programs.

The `dialnorm` parameter is conveyed by the five-bit variable in the Dolby Digital bitstream. Using the subjective level of normal spoken dialogue as a reference, the `dialnorm` value indicates how far the average dialogue level of the encoded program is below digital full scale. Valid values are 1 to 31, which are interpreted as –1 to –31 dBFS.

For example, if the programs in Figure 2-1 were encoded using Dolby Digital, the `dialnorm` value would be set to correspond to the average loudness level shown by the heavy marker. When decoded, dialogue normalization would align the average loudness level to the same output level, as shown in Figure 2-2.

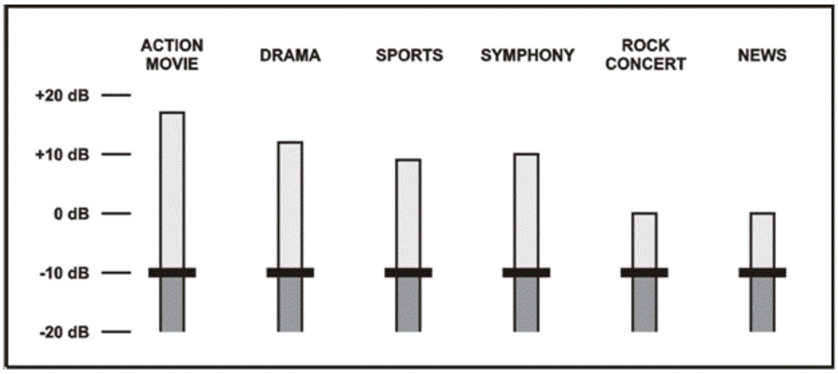


**Figure 2-2** Normalized Audio Level for Equal Average Loudness (0 dB = 200 $mV_{RMS}$)

# Dynamic Range Control

A consistent problem in the delivery of audio programming is that members of the audience desire or even need different amounts of dynamic range depending on the listening situation. Original high-quality programs, such as feature films, are typically mixed with a wide dynamic range. Using dialogue as a reference, loud sounds (such as explosions) are often 20 dB louder, and faint sounds (such as leaves rustling) may be 50 dB quieter. In many listening situations, it is objectionable to allow the sound to become very loud, and thus the loudest sounds must be compressed downward in level. Likewise, the very quiet sounds would be inaudible and must be brought upward in level to be heard.

Because most of the audience benefit from a limited program dynamic range, audio programs that have been mixed with a wide dynamic range are generally compressed before being delivered to the listener. The dynamic range is reduced by bringing down the level of the loud sounds and bringing up the level of the quiet sounds. While this satisfies the needs of most of the audience, it prevents some members of the audience from experiencing the original program in its intended form.

Dolby® Digital satisfies the needs of all members of the audience by allowing each audio program to be encoded with its full dynamic range intact and sending along a compression control signal that can be applied as needed to reduce the dynamic range to fit the individual listening situation.

## 3.1 The dynrng Variable

The dynamic range control (DRC) parameter, conveyed by the variable `dynrng`, indicates a gain change to be applied in the Dolby Digital decoder in order to implement dynamic range compression. The sequence of `dynrng` values represents a compression control signal and is generated by the Dolby Digital encoder. The `dynrng` values typically indicate gain reductions (cut) during loud passages and gain increases (boost) during quiet passages based on desired compression characteristics. These characteristics are determined by preset compression curves or by a mixing engineer manually applying compression during the encoding process. Sounds that are at a level similar to the dialogue level will typically not have their gain changed.

A variety of `dynrng` reproduction alternatives are available, because the audio program is encoded with full dynamic range and dynamic range compression is applied during the decoding process.

It is possible for the Dolby Digital decoder to ignore the `dynrng` values in the bitstream, which results in reproduction of the audio program with full dynamic range. It is also possible for the decoder to use a chosen fraction of the DRC value, or to use different fractions for high-level cut and low-level boost.

In summary, depending on the type of product and user options provided, the Dolby Digital decoder can reproduce audio with the following dynamic range compression:

- Fully compressed dynamic range (as indicated by `dynrng`)
- Uncompressed dynamic range (full dynamic range)
- Partially compressed dynamic range, with different amounts of compression available for high-level signals and low-level signals

## 3.2 Scale Factors: Partial Compression

Partial compression is achieved by multiplying the `dynrng` value by a scale factor from 1.0 to 0.0. A scale factor of 1.0 equals full compression. A scale factor of 0.0 equals no compression. A fractional scale factor, such as 0.5, results in partial compression with a linear change in the scale factor, resulting in a linear change in the amount of compression applied.

Values of `dynrng` representing gain reductions are multiplied by one scale factor, while values representing gain increases are multiplied by a separate scale factor, allowing independent control of high-level cut and low-level boost compression.

## 3.3 The compr Variable

Some products deliver decoded audio via a path with very restricted dynamic range. To constrain the decoded audio program to a known peak level, a second compression parameter (conveyed by the variable `compr`) may be present in the bitstream. The sequence of `compr` values is similar to the `dynrng` control signal, in that it indicates a gain change to be applied in the decoder.

For example, a television signal decoder must modulate the received video and audio signal onto an RF channel for delivery to a low-cost television receiver. In this situation, it is necessary to restrict the maximum peak output level to a known value with respect to dialogue level in order to prevent excessive overmodulation. Most of the time, the compression control signal `dynrng` produces adequate gain reduction so that the absolute peak level is constrained. Because the DRC system is intended only to provide a subjectively pleasing gain reduction, however, there is no guarantee of an adequate reduction of all instantaneous signal peaks to prevent excessive overmodulation.

# Downmixing

Downmixing is used to reproduce the complete audio program when the actual decoder outputs do not match the encoded channel format (audio coding mode) of the audio signal. The process of downmixing takes the information in the channels that do not have corresponding outputs, and mixes this information into the remaining channels according to a standardized set of downmix equations.

Standardized downmix equations allow program producers to monitor the downmixed audio and make any changes necessary to achieve optimal results for all listening situations. For this purpose, the C mix level and surround mix level parameters are available to allow the program producer to affect the relative balance of C and surround channels with respect to the L and R channels when downmixing.

The particular downmix equation used depends on the audio coding mode and the output channel configuration, which is determined by the listening situation.

## 4.1 Output Channel Configuration

The output channel configuration indicates the actual channels to be reproduced. The configuration is determined based on the number of speakers in the system, or on the number of audio outputs needed for a particular listening situation, such as stereo for headphone listening. Valid output channel configurations are listed in Table 4-1.

**Table 4-1** Output Channel Configurations

| Output Channel Configuration | Channels | Listening Setup Environment |
|---|---|---|
| 1/0 | C | TV with mono RF input |
| 2/0 Lo/Ro | L, R | Headphone listening |
| 2/0 Lt/Rt | L, R | For reproduction via Dolby® Pro Logic® decoder |
| 3/0 | L, C, R | No surround speakers |
| 2/2 | L, R, Ls, Rs | No C speaker |
| 3/2 | L, C, R, Ls, Rs | Full multichannel reproduction |

**Note:** The output channel configurations 2/1 and 3/1 are not listed in Table 4-1 because all Dolby Surround systems use at least two surround speakers.

Dolby Surround systems use at least two surround speakers even though the surround signal is mono. Dolby Digital systems must support stereo surround signals with two-speaker surround outputs from the decoder. Therefore, there is no need to support the 2/1 and 3/1 output channel configurations. Programs that have an audio coding mode of 2/1 or 3/1 are still reproduced, but there is no need to downmix to these output channel configurations.

## 4.2 Downmix Versus Reproduction

A downmix is necessary when the encoded channel format (audio coding mode) does not match the output channel configuration. In this situation, the output channel configuration indicates not only the actual channels to be reproduced but also the particular set of downmix equations to use. If the encoded channel format (audio coding mode) matches the output channel configuration, then downmixing is not necessary and the audio is simply reproduced.

For example, assume that the output channel configuration is set to 2/0 Lt/Rt. A program with an audio coding mode of 2/0 will be reproduced with no downmixing. A program with any other audio coding mode will be downmixed according to the set of downmix equations for producing a 2/0 Lt/Rt output.

## 4.3 Compatibility with Dolby Pro Logic Systems

The 2/0 Lt/Rt downmix is a two-channel output compatible with Dolby Surround that can be decoded by Dolby Surround (passive), Pro Logic (active), and Pro Logic II (active) decoders. This downmix provides backward compatibility with existing Dolby Surround systems.

While the Lt/Rt downmix is compatible with Dolby Surround decoders, it is incorrect to state that the Dolby Digital decoder contains a Dolby Surround encoder. The proper way to describe this feature is to state that the Dolby Digital decoder is capable of downmixing the 5.1 channels of audio to a two-channel output compatible with Dolby Surround that can be decoded by Dolby Surround and Dolby Surround Pro Logic decoders.

It is also incorrect to describe the Lt/Rt downmix as Dolby Pro Logic encoding, because Dolby Pro Logic refers to a specific decode process. The proper way to describe the Lt/Rt downmix is to refer to it as a downmix compatible with Dolby Surround.

## 4.4 Downmix Overload Protection

Whenever channels are mixed together during downmixing, the possibility exists for peak-level overload. The typical solution for avoiding overload is to attenuate the result by the minimum amount required to prevent the worst-case buildup from overloading. Unfortunately, this minimum amount of attenuation differs depending on the particular downmix, and having different amounts of attenuation makes the implementation of dialogue normalization very complex.

To promote a uniform solution to the problem of peak-level overload and to simplify the implementation of dialogue normalization, Dolby Digital decoders apply dynamic range compression during downmixing. More specifically, the gain reduction portion of dynamic range compression is used to prevent peak-level overload when downmixing. To prevent overload in all cases, some minimum amount of gain reduction is required. This minimum amount is determined by the encoder, based on the setting of the `dialnorm` variable and on the contents of the audio channels themselves, and automatically included in the `dynrng` and `compr` values.

In most cases, the artistic compression included by the encoder to provide a subjectively pleasing reduction in peak levels will also be enough to prevent overload when downmixing. In cases where this gain reduction is not enough, it is automatically increased to the minimum required to prevent overload.

# Operational Modes

To simplify the implementation of dialogue normalization, DRC, and downmixing, Dolby® Digital decoder Implementations offer standard operational modes (called Line mode and RF mode) that are included within the decoder Implementation itself. Two custom modes offer additional design flexibility for product designers who wish to bypass the standard operational modes and implement these features elsewhere in the product. Typically, the Line and RF modes are used in the majority of products; custom modes are used primarily in more esoteric audio products, where additional Implementation cost and complexity are not of primary concern.

Setting the decoder Implementation to operate in one of these modes automatically configures the decoder to implement dialogue normalization, DRC, and downmixing according to the following descriptions.

## 5.1 Line Mode

Line mode is intended for use in products providing line- or speaker-level outputs, and is applicable to the widest range of products. Products such as set-top boxes, DVD players, DTVs, A/V surround decoders, and outboard Dolby Digital decoders typically use this mode.

Summary of Line mode features:

- Dialogue normalization enabled
- Dialogue reproduced at a constant level of –31 dBFS
- DRC using the `dynrng` compression variable
- High-level cut compression scaling allowed when not downmixing
- Low-level boost compression scaling allowed

Dialogue normalization is implemented as part of the decoder Implementation, and the dialogue reference output level for Line mode is defined as –31 dBFS. Dialogue normalization attenuates all programs by the difference between the `dialnorm` value and –31 dBFS, thereby aligning the dialogue reference level of each program to –31 dBFS.

For example, movie soundtracks on DVD typically have a `dialnorm` value of –27 dBFS. Dialogue normalization would therefore attenuate the soundtrack by 4 dB (the difference between –27 and –31 dB).

DRC is used to set the `dynrng` compression variable. When the decoder is downmixing, high-level cut scaling is not allowed, and full peak-level compression is applied to prevent peak-level overload. Scaling of low-level boost is allowed at all times. With this flexibility, full compression, no compression, and partial compression can all be implemented in a product using this mode.

**Note:** Scaling of high-level cut is allowed only when the decoder is downmixing.

Figure 5.2 shows the signal relationships in Line mode under different listening situations. Note that whether or not the decoder is downmixing or applying compression, the average program loudness (dialogue) remains constant.

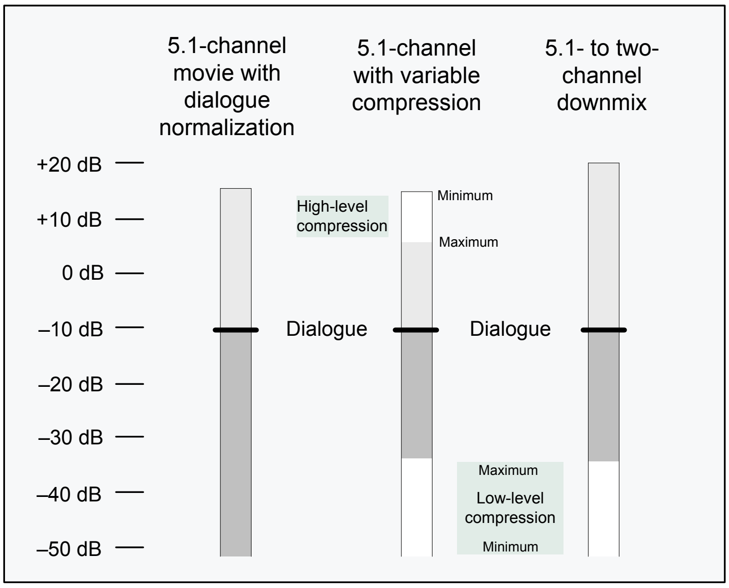


**Figure 5-1** Signal Relationships in Line Mode (0 dB = –20 dBFS

## 5.2 RF Mode

RF mode is intended for products generating a downmixed signal for subsequent channel 3 RF modulation and transmission to, for example, a low-cost television receiver. In this mode, the overall program level is raised 11 dB. By limiting headroom to a maximum of 20 dB above average dialogue level, excessive overmodulation of the television receiver is prevented. The maximum headroom limit provides an RF modulation level that causes dialogue to be reproduced at a level that compares well with quality television broadcasts and premium movie channels.

Summary of RF mode features:

- Dialogue normalization enabled
- Dialogue reproduced at a constant level of –20 dBFS
- DRC using the `dynrng` compression variable
- Compression scaling not allowed (always fully compressed)
- +11 dB gain shift to raise overall program level

Dialogue normalization is again implemented as part of the decoder Implementation, but due to the +11 dB gain shift, the dialogue normalization reference output level for RF mode is defined as –20 dBFS.

Both the `dynrng` variable and the `compr` variable are used for DRC. In RF mode, full dynamic range compression is applied at all times, and scaling is not allowed.

Figure 5-2 shows the signal relationships in RF mode under different listening situations. Note that whether or not the decoder is downmixing, the average program loudness remains constant.

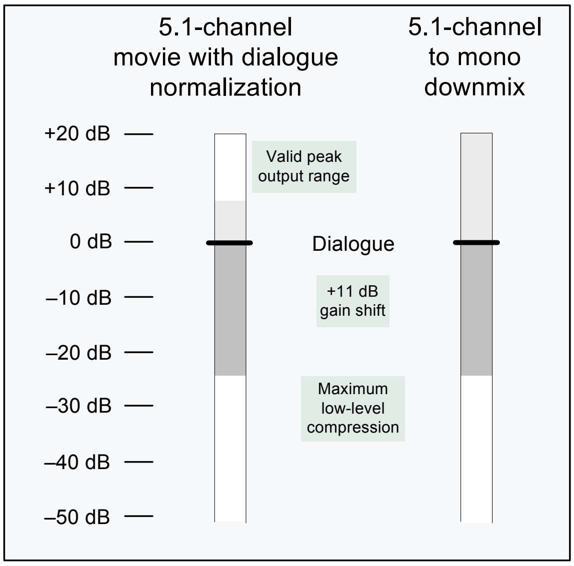


**Figure 5-2** Signal Relationships in RF Mode (0 dB = –20 dBFS)

Figure 5-3 shows an example of how signals generated with RF and Line modes relate to the modulation index of a typical RF modulator circuit. Note that while Line mode may be used for this purpose, the improvement in dynamic range is offset by a lower average loudness compared with other television signal sources.

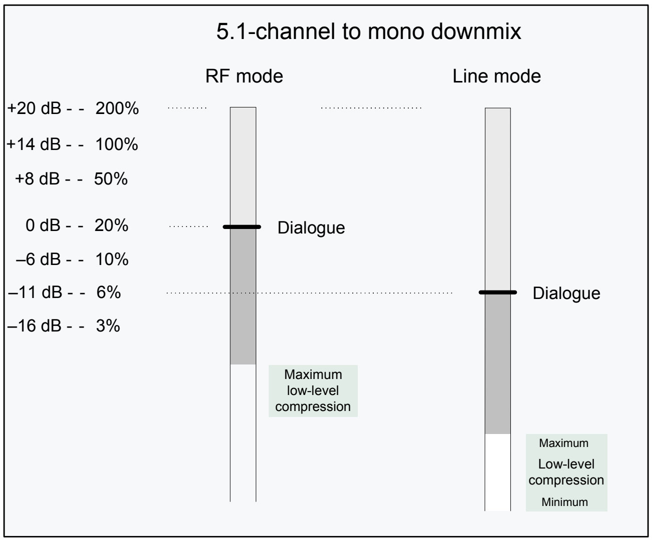


**Figure 5-3** RF Modulator Signal Levels (0 dB = –20 dBFS)

## 5.3 Custom Modes

The custom modes bypass some or all of the features of Line and RF modes and provide additional design flexibility for product designers who wish to implement these features elsewhere in the product. No product is required to use the custom modes. If the custom modes are used, dialogue normalization, DRC, and downmixing must still be implemented in the final product. There are two custom modes: Custom mode 1 and Custom mode 0.

**Note:** Custom mode on the DP564 Multichannel Audio Decoder is a scalable version of Line mode and is not the same as either of the custom modes described here.

Summary of Custom mode 1 features:

- Dialogue normalization enabled
- DRC using the `dynrng` compression variable
- High-level cut compression scaling allowed
- Low-level boost compression scaling allowed
- –11 dB gain shift imposed when downmixing to prevent peak-level overload

Summary of Custom mode 0 features:

- Dialogue normalization disabled
- DRC using the `dynrng` compression variable
- High-level cut compression scaling allowed
- Low-level boost compression scaling allowed
- –11 dB gain shift imposed when downmixing to prevent peak-level overload

In Custom mode 1, dialogue normalization is implemented as part of the decoder Implementation, and the dialogue reference output level is defined as –31 dBFS.

In Custom mode 0, dialogue normalization is not implemented as part of the decoder Implementation and must be implemented elsewhere in the product, under control of the `dialnorm` variable in the bitstream. In this case, dialogue normalization is usually implemented in the master volume control. No changes in the position of the volume control shall be visible to the user when dialogue normalization occurs.

Both custom modes use the `dynrng` variable for DRC. Scaling of high-level cut and low-level boost is allowed at all times.

Recall that in Line mode, when the decoder is downmixing, full peak-level compression is applied and its effect cannot be reduced. For this reason, both custom modes impose a –11 dB gain shift to ensure that peak-level overload does not occur when the decoder is downmixing. This gain shift affects the output level and also dialogue normalization in Custom mode 1. Products using the custom modes must apply +11 dB of gain correction to the downmixed output to ensure the correct level. Although the custom modes allow more flexibility in applying dynamic range compression during downmixing, the result is that the final audio outputs may exhibit a higher noise floor in this situation.

Figure 5-4 shows the signal relationships in Custom mode 0 under different listening situations with no gain corrections. Dialogue normalization is disabled, and the average dialogue level is reproduced at the level that it was encoded.

**Note:** When the decoder is downmixing, the average program loudness is reduced by 11 dB.

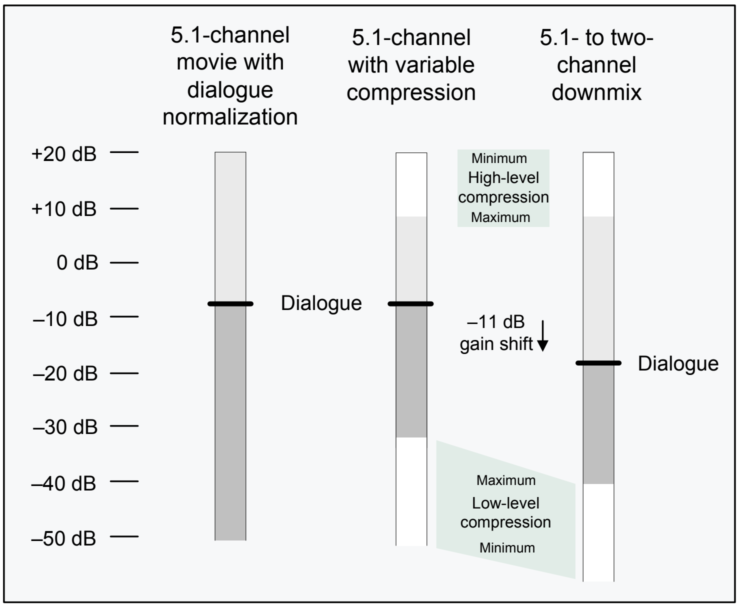


**Figure 5-4** Signal Relationships in Custom Mode 0 (0 dB = –20 dBFS)

## 5.4 Dual-Mono Modes

Programs with an audio coding mode of 1+1 contain two independent mono channels and are referred to as dual-mono programs. Dual-mono programs are typically used to carry two different language versions of the same program in the same bitstream. Decoders that support dual-mono programs will have additional functionality similar to televisions that support the secondary audio program (SAP) feature.

**Note:** The implementation or use of dual-mono mode is not allowed in some systems.

This additional functionality comes in the form of four dual-mono reproduction modes that determine which channels are reproduced at the output. The four dual-mono reproduction modes are:

- Stereo (channels 1 and 2)
- Mono channel 1 (only channel 1)
- Mono channel 2 (only channel 2)
- Mixed mono (channel 1 + channel 2)

Table 5-1 shows how the independent channels are reproduced depending on the dual-mono mode selected and the output channel configuration.

**Table 5-1** Dual-Mono Reproduction Modes

| Dual-Mono Mode | Output Channel Configuration | | | | | |
|---|---|---|---|---|---|---|
| | 1/0 | 2/x | | 3x | | |
| | Center | Left | Right | Left | Center | Right |
| Stereo | Ch 1 + 2 | Ch 1 | Ch 2 | Ch 1 | NA | Ch 2 |
| Mono channel 1 | Ch 1 | Ch 1 | Ch 1 | NA | Ch 1 | NA |
| Mono channel 2 | Ch 2 | Ch 2 | Ch 2 | NA | Ch 2 | NA |
| Mixed mono | Ch 1 + 2 | Ch 1 + 2 | Ch 1 + 2 | NA | Ch 1 + 2 | NA |

Ch = channel

NA = Not available

# Parameter Definitions

This section defines the parameters contained within Dolby® Digital and Dolby Digital Plus bitstreams. Dolby Digital Plus bitstreams contain all the parameters defined in both the Dolby Digital and Dolby Digital Plus sections.

## 6.1 Dolby Digital Bitstream Parameters

Dolby Digital bitstream parameters, as listed in Table 1-3, are defined in this section.

### 6.1.1 Dialogue Normalization (dialnorm)

The dialogue normalization parameter (`dialnorm`) is used to adjust the dialogue level of the bitstream program at the decoder output (see Chapter 2). The `dialnorm` value range and the default reference output level for Line, RF, and custom mode operation are listed in Table 6-1.

**Table 6-1** Dialogue Normalization Parameter

| Dialogue Normalization Value | Dialogue Reference Output Level | | | |
|---|---|---|---|---|
| | Line Mode | RF Mode | Custom Mode | |
| | | | 1 | 0 |
| –31 to 0 dBFS | –31 dBFS | –20 dBFS | –31 dBFS | NA |

NA = Dialogue normalization is not available in Custom mode 0.

For more details about the function of dialogue normalization in Line and RF modes, refer to Section 5.1 and Section 5.2. For the custom modes, refer to Section 5.3.

### 6.1.2 Audio Coding Mode (acmod)

The audio coding mode parameter (`acmod`) indicates the active channels within the encoded bitstream and affects both the encoder and consumer decoder. The audio coding mode instructs the encoder about which inputs to use for a particular program and informs the decoder what channels are present to ensure that the decoded audio is directed to the correct speakers.

The setting is described as X/Y, where X is the number of front channels (L, C, R) and Y the number of surround channels.

The availability of certain channel modes depends on the encoder data rate and whether the LFE channel is present. For example, a mono bitstream is not allowed with an LFE channel (1.1), or a 3/2 bitstream at 96 kbps. The typical minimum data rates are listed in Table 6-2 for each audio coding mode setting.

**Note:** The presence of the LFE channel is indicated through a different metadata parameter.

**Table 6-2** Audio Coding Mode (acmod)

| Audio Coding Mode | Minimum Data Rate |
|---|---|
| 1+1 | Dual mono is not valid for DTV broadcast or DVD production. |
| 1/0 (mono) | 56 kbps. (96 kbps is common.) |
| 2/0 (stereo) | 96 kbps. (192 kbps is common.) |
| 3/0 | 256 kbps. |
| 2/1 | 256 kbps. |
| 3/1 | 320 kbps. |
| 2/2 | 320 kbps. |
| 3/2 | 384 kbps. (448 kbps is common.) |

### 6.1.3 LFE Channel (lfeon)

The status of the LFE channel parameter (`lfeon`) indicates to an encoder whether an LFE channel is present within the bitstream. The audio coding mode determines whether the LFE channel parameter is allowed. At least three channels are required to enable the LFE channel, as listed in Table 6-3.

**Table 6-3** LFE Channel Parameter

| LFE Channel | Audio Coding Mode |
|---|---|
| Enabled | 2/1, 3/0, 2/2, 3/1, 3/2 |
| Disabled | 1/0, 1+1, 2/0 |

### 6.1.4 Bitstream Identification (bsid)

The bitstream identification parameter (`bsid`) indicates the version of the bitstream as a number within the range 0–31. Table 6-4 shows the bitstream ID values for both a Dolby Digital and Dolby Digital Plus bitstream.

**Table 6-4** Bitstream ID Parameter

| Bitstream ID Value | Bitstream Present |
|---|---|
| 8 or less | Dolby Digital |
| 16 or greater | Dolby Digital Plus |

The decoder checks the `bsid` value to avoid decoding bitstreams that contain unsupported parameters or formats.

### 6.1.5 Bitstream Mode (bsmod)

The bitstream mode parameter (`bsmod`) describes the audio service contained within the Dolby Digital bitstream. A complete audio program may consist of the following:

- Main audio service: A complete mix of the entire audio program
- Associated audio service
- One main service combined with an associated service

A detailed description of each bitstream mode setting is listed in Table 6-5.

**Table 6-5** Bitstream Mode (bsmod)

| Bitstream Mode Setting | Definition |
|---|---|
| Complete main (CM) (Default)* | CM flags the bitstream as the main audio service for the program and indicates that all elements are present to form a complete audio program. Currently, this is the most common default setting. The CM service may contain from one (mono) to six (5.1) channels. |
| Main music and effects (ME) | This bitstream is the main audio service for the program, minus a dialogue channel. The dialogue channel, if any, is intended to be carried by an associated dialogue service. Different dialogue services can be associated with a single ME service to support multiple languages. |
| Visually impaired (VI) (associated service) | This is typically a single-channel program intended to provide a narrative description of the picture content to be decoded along with the main audio service. The VI service may also be a complete mix of all program channels, comprising up to six channels. |
| Hearing impaired (HI) (associated service) | This is typically a single-channel program intended to convey audio that has been processed for increased intelligibility and decoded along with the main audio service. The HI service may also be a complete mix of all program channels, comprising up to six channels. |
| Dialogue (D) (associated service) | This is typically a single-channel program intended to provide a dialogue channel for an ME service. If the ME service contains more than two channels, the D service is limited to only one channel; if the ME service is two channels, the D service can be a stereo pair. The appropriate channels of each service are mixed together. (This requires special decoders.) |
| Commentary (C) (associated service) | This is typically a single-channel program intended to convey additional commentary that can be optionally decoded along with the main audio service. This service differs from a dialogue service because it contains an optional, rather than a required, dialogue channel. The C service may also be a complete mix of all program channels, comprising up to six channels. |
| Emergency (E) (associated service) | This is a single-channel service that is given priority in reproduction. When the E service appears in the bitstream, it is given priority in the decoder and the main service is muted. |
| Voiceover (VO) (associated service) | This is a single-channel service intended to be decoded and mixed to the C channel (requires special decoders). |
| Karaoke (K) (main service) | This bitstream is a special service for karaoke playback. In this case, the L and R channels contain music, the C channel has a guide melody, and the Ls and Rs channels carry optional backing vocals. |

* CM is commonly used as the default setting. An example exception to this rule is karaoke DVD, or an emergency service within DTV.

### 6.1.6 Line Mode Compression Words (dynrng)

Line mode is discussed in Section 5.1.

### 6.1.7 RF Mode Compression Profile (compr)

RF mode is discussed in Section 5.2.

### 6.1.8 Center Downmix Level (cmixlev)

When the encoded audio has three front channels (L, C, R) but the consumer has only two front speakers (left and right), the `cmixlev` parameter indicates the nominal downmix level for the C channel with respect to the L and R channels. Dolby Digital decoders use the `cmixlev` parameter during downmixing in Lo/Ro mode when extended BSI parameters are not active. Table 6-6 provides definitions for all C downmix level parameter settings.

**Table 6-6** Center Downmix Level (cmixlev)

| Center Downmix Level | Definition |
|---|---|
| 0.707 (–3 dB) (default) | The C channel is attenuated 3 dB and sent to the L and R channels. |
| 0.595 (–4.5 dB) | The C channel is attenuated 4.5 dB and sent to the L and R channels. |
| 0.5 (–6 dB) | The C channel is attenuated 6 dB and sent to the L and R channels. |

### 6.1.9 Surround Downmix Level (surmixlev)

When the encoded audio has one or more surround channels but the consumer does not have surround speakers, the `surmixlev` parameter indicates the nominal downmix level for the surround channels with respect to the L and R channels. Dolby Digital decoders use the `surmixlev` parameter during downmixing in Lo/Ro mode when extended BSI parameters are not active. Table 6-7 provides definitions for all surround downmix level settings.

**Table 6-7** Surround Downmix Level (surmixlev)

| Surround Downmix Level | Definition |
|---|---|
| 0.707 (–3 dB) (default) | The Ls and Rs channels are each attenuated 3 dB and sent to the L and R channels, respectively. |
| 0.5 (–6 dB) | The Ls and Rs channels are each attenuated 6 dB and sent to the L and R channels, respectively. |
| 0 (–∞ dB) | The surround channels are discarded. |

### 6.1.10 Dolby Surround Mode (dsurmod)

This parameter indicates to a decoder whether the two-channel encoded bitstream contains a Dolby Surround (Lt/Rt) program that requires Dolby Pro Logic® decoding. A decoder can use the `dsurmod` flag to switch on Dolby Pro Logic decoding automatically as required. Table 6-8 provides definitions for all Dolby Surround mode settings.

**Table 6-8** Dolby Surround Mode (dsurmod)

| Dolby Surround Mode | Definition |
|---|---|
| Not Dolby Surround | The bitstream contains information not encoded in Dolby Surround. |
| Dolby Surround | The bitstream contains information encoded in Dolby Surround. After Dolby Digital or Dolby Digital Plus decoding, the PCM audio is processed by the Dolby Pro Logic matrix decoder. |
| Not indicated | There is no indication. |

### 6.1.11 Audio Production Information Exists (audprodie)

The audio production information exists parameter (`audprodie`) indicates whether the mixing level (see Section 6.1.12) and room type (see Section 6.1.13) values are valid. If they are valid (with an `audprodie` value of yes), then a receiver or amplifier can use these values. If they are not valid (with an `audprodie` value of no), then the values are ignored. In practice, only high-end consumer equipment implements these features. Dolby Digital Plus encoders send this information only if the settings deviate from the default. Thus, control of this parameter is for Dolby Digital only, and not provided to the user for Dolby Digital Plus.

**Table 6-9** Audio Production Information Exists (audprodie)

| Audio Production Information Exists | Definition |
|---|---|
| Yes | Mixing level and room type parameters are valid. |
| No | Mixing level and room type parameters are invalid (ignored). |

### 6.1.12 Mixing Level (mixlevel)

The mixing level parameter (`mixlevel`) describes the peak sound pressure level (SPL) used during the final mixing session at the studio or on the dubbing stage. The parameter allows an amplifier to set its volume control such that the SPL in the replay environment matches that of the mixing room. This control operates in addition to the dialogue level control, and is best thought of as the final volume setting on the consumer's equipment.

This value is determined by measuring the SPL of pink noise at studio reference level and then adding the amount of digital headroom above that level. For example, if 85 dB equates to a reference level of –20 dBFS, the mixing level is 105 dB (85 + 20).

**Table 6-10** Mixing Level (mixlevel)

| Parameter | Value Range |
|---|---|
| `mixlevel` | 80 to 111 dB, in 1 dB increments |

### 6.1.13 Room Type (roomtyp)

The room type parameter (`roomtyp`) describes the equalization used during the final mixing session at the studio or on the dubbing stage. A large room is a dubbing stage with the industry standard X-curve equalization; a small room has flat equalization. This parameter allows an amplifier to be set to the same equalization as heard in the final mixing environment.

**Table 6-11** Room Type (roomtyp)

| Room Type | Definition |
|---|---|
| Large | Standard X-curve equalization |
| Small | Flat equalization |
| Not indicated | |

### 6.1.14 Copyright Bit (copyrightb)

The copyright bit parameter (`copyrightb`) indicates whether the encoded Dolby Digital bitstream is copyright protected. This parameter has no effect on Dolby Digital decoders, and its purpose is to provide information only.

**Table 6-12** Copyright Bit (copyrightb)

| Copyright Bit | Definition |
|---|---|
| Yes | Copyright protected bitstream |
| No | Not copyright protected |

### 6.1.15 Original Bitstream (origbs)

The original bitstream parameter (`origbs`) indicates whether the encoded Dolby Digital bitstream is the master version or a copy. It has no effect on Dolby Digital decoders, and its purpose is to provide information only.

**Table 6-13** Original Bitstream (origbs)

| Original Bitstream | Definition |
|---|---|
| Yes | Original bitstream. |
| No | Bitstream is a copy. |

### 6.1.16 Extended Bitstream Information Exists (xbsie)

The extended bitstream information exists parameter (`xbsie`) indicates whether the encoded Dolby Digital bitstream contains extended bitstream parameters. If `xbsie` is set to a value of yes, then the extended bitstream information parameters are included in the bitstream. (See the parameter list in Table 1-4 and descriptions in Section 6.2.)

**Table 6-14** Extended Bitstream Information Exists (xbsie)

| Extended Bitstream Information Exists | Definition |
|---|---|
| Yes | Bitstream includes extended bitstream information parameters. |
| No | Extended bitstream information parameters are not included. |

## 6.2 Dolby Digital Extended Bitstream Information Parameters

This section defines each Dolby Digital extended bitstream parameter, as listed in Table 1-4.

In response to requests from content producers, Dolby Laboratories modified the definitions of several metadata parameters from their original definition as described in the first revision of the A/52 standard. The revised definitions allow more information to be carried about the audio program and also more choices for stereo downmixing. When the metadata parameters carried in Dolby Digital were first described, they were generically called bitstream information, or BSI. We refer to the additional parameter definitions as extended BSI.

Because the revised definitions affect metadata parameters not used by consumer decoders, all decoders must be compatible with the revised bitstream. Newer decoders that are programmed to detect and decode the new extended BSI parameters must be able to implement the new features that extended BSI provides. The original definition of the extended BSI parameters is available in Annex D of A52B.

### 6.2.1 Preferred Stereo Downmix Mode (dmixmod)

The preferred stereo downmix mode parameter (`dmixmod`) allows the producer to select either the Lt/Rt or Lo/Ro downmix in a consumer decoder with stereo outputs. Consumer receivers are able to override this selection, but this parameter provides the opportunity for a 5.1-channel soundtrack to play in Lo/Ro mode without user intervention. This is especially useful on music content. Dolby Digital decoders not capable of producing a downmix compatible with Dolby Pro Logic II will perform an Lt/Rt downmix when a Dolby Pro Logic II downmix is requested.

The preferred stereo dowmix mode settings are listed in Table 6-15.

**Table 6-15** Preferred Stereo Downmix Parameter Settings

| Preferred Stereo Downmix Mode |
|---|
| Not indicated |
| Lt/Rt preferred |
| Lo/Ro preferred |
| Lt/Rt Dolby Pro Logic II preferred |

### 6.2.2 Lt/Rt Center Downmix Level (ltrtcmixlev)

The Lt/Rt C downmix level parameter (`ltrtcmixlev`) indicates the level shift applied to the C channel when adding to the L and R outputs as a result of downmixing to an Lt/Rt output. Its operation is similar to the C downmix level in the universal metadata.

The Lt/Rt C downmix level settings are listed in Table 6-16.

**Table 6-16** Lt/Rt Center Downmix Level Parameter Settings

| Lt/Rt Center Downmix Level |
|---|
| 1.414 (+3 dB) |
| 1.189 (+1.5 dB) |
| 1 (0 dB) |
| 0.841 (–1.5 dB) |
| 0.707 (–3 dB) |
| 0.595 (–4.5 dB) |
| 0.5 (–6 dB) |
| 0 (–999 dB) |

### 6.2.3 Lt/Rt Surround Downmix Level (ltrtsurmixlev)

The Lt/Rt surround downmix level parameter (`ltrtsurmixlev`) indicates the level shift applied to the surround channels when downmixing to an Lt/Rt output. Its operation is similar to the surround downmix level in the universal metadata.

The Lt/Rt surround downmix level settings are listed in Table 6-17.

**Table 6-17** Lt/Rt Surround Downmix Level Parameter Settings

| Lt/Rt Surround Downmix Level |
|---|
| 0.841 (–1.5 dB) |
| 0.707 (–3 dB) |
| 0.595 (–4.5 dB) |
| 0.5 (–6 dB) |
| 0 (–999 dB) |

### 6.2.4 Lo/Ro Center Downmix Level (lorocmixlev)

The Lo/Ro C downmix level parameter (`lorocmixlev`) indicates the level shift applied to the C channel when adding to the L and R outputs as a result of downmixing to an Lo/Ro output. When extended BSI parameters are active, this parameter replaces the C downmix level parameter in the universal parameters.

The Lo/Ro C downmix level settings are listed in Table 6-18.

**Table 6-18** Lo/Ro Center Downmix Level Parameter Settings

| Lo/Ro Center Downmix Level |
|---|
| 1.414 (+3 dB) |
| 1.189 (+1.5 dB) |
| 1 (0 dB) |
| 0.841 (–1.5 dB) |
| 0.707 (–3 dB) |
| 0.595 (–4.5 dB) |
| 0.5 (–6 dB) |
| 0 (–999 dB) |

### 6.2.5 Lo/Ro Surround Downmix Level (lorosurmixlev)

The Lo/Ro surround downmix level parameter (`lorosurmixlev`) indicates the level shift applied to the surround channels when downmixing to an Lo/Ro output. When extended BSI parameters are active, this parameter replaces the surround downmix level parameter in the universal parameters.

The Lo/Ro surround downmix level settings are listed in Table 6-19.

**Table 6-19** Lo/Ro Surround Downmix Level Parameter Settings

| Lo/Ro Surround Downmix Level |
|---|
| 0.841 (–1.5 dB) |
| 0.707 (–3 dB) |
| 0.595 (–4.5 dB) |
| 0.5 (–6 dB) |
| 0 (–999 dB) |

### 6.2.6 Dolby Digital Surround EX Mode (dsurexmod)

The Dolby Digital Surround EX™ mode parameter (`dsurexmod`) is used to identify the encoded audio as material encoded in Dolby Digital Surround EX. This parameter is used only if the encoded audio has two surround channels. An amplifier or receiver with Dolby Digital Surround EX decoding can use this parameter as a flag to switch the decoding on or off automatically. The behavior is similar to that of the Dolby Surround mode parameter.

The Dolby Digital Surround EX Mode settings are listed in Table 6-20.

**Table 6-20** Dolby Digital Surround EX mode Parameter Settings

| Dolby Digital Surround EX Mode |
|---|
| Encoded in Dolby Digital Surround EX |
| Not encoded in Dolby Digital Surround EX |
| Not indicated |

### 6.2.7 A/D Converter Type (adconvtyp)

The A/D converter type parameter (`adconvtyp`) allows audio previously passed through a particular A/D conversion stage to be marked as such, so that a decoder may apply the complementary D/A process.

**Table 6-21** A/D Converter Type (adconvtyp)

| A/D Converter Type |
|---|
| Standard |
| HDCD |

## 6.3 Dolby Digital Plus Bitstream Parameters

Dolby Digital Plus bitstreams consist of Dolby Digital and extended BSI parameters, as well as three unique Dolby Digital Plus parameters, as part of the main bitstream (see Table 1-5). This section defines the five unique Dolby Digital Plus bitstream parameters currently supported.

**Note:** The bitstream parameters `audprodie` and `xbsie` are not listed as Dolby Digital Plus parameters in Table 1-5. The bitstream parameter `xbsie` is removed, while the `audprodie` parameter is hidden from the user by the behavior of the encoder and decoder. For information about these parameters, refer to Section 6.1.11 and Section 6.1.16.

Inspection of the A/52B standard yields more new bitstream parameters not described in this section, nor listed in Table 1-5. The parameters listed in this section are supported by encoders and decoders currently in use. As encoders or decoders utilize more of the new parameters, this section will be updated to reflect those changes. For information on Dolby Digital Plus parameters that are not listed, see Annex E of the A/52B standard.

This section defines only the new Dolby Digital Plus bitstream parameters currently being supported (not the shared parameters with Dolby Digital). The complete set of Dolby Digital Plus bitstream parameters is listed in Table 1-5.

### 6.3.1 Bitstream Type (strmtyp)

A Dolby Digital Plus professional encoder that implements bitstream mixing identifies the main and associated audio bitstreams as independent or dependent substreams. (See Section 6.3.2 for information about the substream ID.)

**Table 6-22** Bitstream Type (strmtyp)

| strmtype | Description |
|---|---|
| Independent | The frames comprise an independent bitstream or substream. The audio program may be decoded independently of any other substreams that might exist in the bitstream. |
| Dependent | The frames comprise a dependent substream. The audio program must be decoded in conjunction with the independent substream with which it is associated. |

### 6.3.2 Substream ID (substreamid)

All Dolby Digital Plus bitstreams (.ec3) must contain an independent substream assigned substream ID 0. The independent substream assigned substream ID 0 must be the first substream present in the bitstream. Each independent substream may have up to eight dependent substreams associated with it.

**Table 6-23** Substream ID (substreamid)

| Bitstream Type | Substream ID | Description |
|---|---|---|
| Independent | 0–7 | ID 0 = Main audio program<br>ID 1–7 = Associated audio program |
| Dependent | 0–7 | ID 0–7 = Associated audio program |

Dolby Digital Plus bitstreams may contain up to seven additional independent substreams, assigned substream ID 1–7.

Independent substream IDs must be assigned sequentially in the order the independent substreams are present in the bitstream.

Dependent substreams must immediately follow the independent substream with which they are associated. Dependent substreams are assigned substream IDs 0–7, which must be assigned sequentially according to the order the dependent substreams are present in the bitstream.

The substream identification parameter can be used, in conjunction with additional bitstream metadata, to enable carriage of a single audio program of more than 5.1 channels, multiple programs or a mixture of programs with up to 5.1 channels, and programs with greater than 5.1 channels.

If a Dolby Digital bitstream is present in the Dolby Digital Plus bitstream, then the Dolby Digital bitstream must be treated as an independent substream, assigned substream ID 0.

### 6.3.3 LFE Downmix Level Code Exists (lfelevcode)

The parameter `lfelevcode` indicates whether the LFE downmix level code is provided in the bitstream. This parameter is provided only if `lfeon` is enabled (see Section 6.1.3).

**Table 6-24** LFE Downmix Level Code Exists (lfelevcode)

| LFE Downmix Level Code Exists | Description | LFE Enabled (lfeon) |
|---|---|---|
| Yes | LFE downmix level code is provided in the bitstream. | Yes |
| No | LFE downmix level code is not provided in the bitstream. | No |

### 6.3.4 LFE Downmix Level Code (lfemixlevcod)

LFE downmixing allows a one- or two-channel system without a subwoofer to reproduce the LFE channel. In current Dolby Digital decoders with 2/0 or 1/0 output mode selected and LFE output disabled, the LFE channel audio is dropped and not reproduced.

Dolby Digital Plus allows LFE channel reproduction by downmixing the LFE content into the available channels. This five-bit parameter (`lfemixlevcod`) corresponds to the LFE mix level (`lfemixlev`).

**Table 6-25** LFE Downmix Level Code

| Parameter | Value Range |
|---|---|
| `lfemixlevcod` | 0–31: +10 to –21 dB, in 1 dB increments<br>(0 = +10 dB, 31 = –21 dB) |

The LFE mix level is related to the LFE downmix level code by the following equation:

`lfemixlev` = (10 – `lfemixlevcod`) dB

### 6.3.5 Source Sample Rate Code (sourcefscod)

This one-bit parameter (`sourcefscod`) indicates whether the source material was sampled at twice the sampling indicated by `fscod`. The decoder may decide to upsample to compensate for this. The only sampling rates designated for upsampling are 44.1 and 48 kHz.

**Table 6-26** Source Sample Rate Code (sourcefscod)

| Source Sample Rate Code | Description |
|---|---|
| 1 | Source material was sampled at twice the sampling rate indicated by `fscod`. |
| 0 | Source material was not sampled at twice the sampling rate. |

`fscod` = Encoded sampling rate

# Dolby Digital Plus Bitstream Mixing Metadata Parameters

This section describes the control parameters to configure the mixing of associated audio services in an advanced Dolby® Digital Plus decoder.

Bitstream mixing metadata parameters define how associated audio is mixed with the main audio substream. The mixing metadata parameters must always be carried in the additional associated audio substreams instead of the main audio substream. Any mixing metadata configured in the main audio program are ignored when the associated audio substream contains mixing metadata. The mixing metadata parameter is listed in Table 7-1. For more details, refer to ETSI TS 102 366.

## 7.1 External Program Scale Factor (extpgmscl)

The external program scale factor parameter (`extpgmscl`) is applied to the external audio program (typically, the main audio service) for mixing with the current audio.

**Table 7-1** External Program Scale Factor (extpgmscl)

| Parameter | Control Value Range |
| --- | --- |
| `extpgmscl` | +12 to –50 dB, in 1 dB increments |

## 7.2 External Program Channel Scale Factors

The external program channel scale factors set the scale value for the respective channel of the external audio program during mixing. The total gain applied to the individual channel of the external program is the sum of the scale factors indicated by the external program scale factor (see Section 7.1) and the external channel scale factors listed in Table 7-2.

**Table 7-2** External Program Channel Scale Factors

| Parameter | External Program Channel Name | Control Settings |
| --- | --- | --- |
| `extpgmlscl` | L channel | Scale factor settings for each channel:<br>–1 to –28 dB, and mute |
| `extpgmcscl` | C channel | |
| `extpgmrscl` | R channel | |
| `extpgmlsscl` | Ls channel | |
| `extpgmrsscl` | Rs channel | |
| `extpgmlfescl` | LFE channel | |

## 7.3 Downmix Scale Factor (dmixscl)

The downmix scale factor parameter (`dmixscl`) specifies a scale value to be applied to an external program previously downmixed before individual channel scale factors could be applied. If the `extpgmscl` field is present in the bitstream (see Section 7.1), the total gain applied to the external program is the sum of the scale factors indicated by the `extpgmscl` and `dmixscl` fields.

**Table 7-3** Downmix Scale Factor (dmixscl)

| Parameter | Control Settings |
|---|---|
| `dmixscl` | –1 to –28 dB, and mute |

## 7.4 Panning Mean Angle (panmean)

The panning mean angle parameter (`panmean`) defines the position of the mono associated audio stream in the surround field. The parameter specifies the location of the mono stream in steps of 1.5°, from 0° to 358.5°.

**Table 7-4** Panning Mean Angle (panmean)

| Parameter | Control Settings |
|---|---|
| `panmean` | 0° to 358.5°, in 1.5° steps |