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**Information technology — MPEG  
audio technologies —**

Part 4:  
**Dynamic Range Control**

**AMENDMENT 1: Parametric DRC, gain  
mapping and equalization tools**

*Technologies de l'information — Technologies audio MPEG —*

*Partie 4: Contrôle de gamme dynamique*

*AMENDEMENT 1: Outils de DRC paramétrique, de mappage des gains  
et d'égalisation*

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This document was prepared by Technical Committee ISO/IEC JTC 1, *Information technology, SC 29, Coding of audio, picture, multimedia and hypermedia information*.

A list of all parts in the ISO/IEC 23003 series can be found on the ISO website.

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# Information technology — MPEG audio technologies —

## Part 4: Dynamic Range Control

### AMENDMENT 1: Parametric DRC, gain mapping and equalization tools

*Page vi, Introduction*

Add the following at the end of the Introduction:

Loudness normalization is fully integrated with DRC and peak control to avoid clipping. A metadata-controlled equalization tool is provided to compensate for playback scenarios that impact the spectral balance, such as downmix or DRC. Furthermore, the DRC tool supports metadata-based loudness equalization to compensate the effect of playback level changes on the spectral balance.

*Page 2, Clause 4*

Insert the following new definitions and maintain the alphabetical order:

`mod` modulo operator:  $(x \bmod y) = x - y \cdot \text{floor}(x/y)$

`sizeof(x)` size operator that returns the bit size of a field

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*Page 3, Clause 5*

Replace the first paragraph:

The technology described in this part of ISO/IEC 23003 is called DRC tool. It provides efficient control of dynamic range, loudness, and clipping based on metadata generated at the encoder. The decoder can choose to selectively apply the metadata to the audio signal to achieve a desired result. Metadata for dynamic range compression consists of encoded time-varying gain values that can be applied to the audio signal. Hence, the main blocks of the DRC tool include a DRC gain encoder, a DRC gain decoder, a DRC gain modification block, and a DRC gain application block. These blocks are exercised on a frame-by-frame basis during audio processing. Various DRC configurations can be conveyed in a separate bitstream element, such as configurations for a downmix or combined DRCs. The DRC set selection block decides based on the playback scenario and the applicable DRC configurations which DRC gains to apply to the audio signal. Moreover, the DRC tool supports loudness normalization based on loudness metadata.

With:

The technology described in this document is called the “DRC tool”. It provides efficient control of dynamic range, loudness, and clipping based on metadata generated at the encoder. The decoder can choose to selectively apply the metadata to the audio signal to achieve a desired result. Metadata for dynamic range compression consists of encoded time-varying gain values that can be applied to the audio signal. Hence, the main blocks of the DRC tool include a DRC gain encoder, a DRC gain decoder, a DRC gain modification block, and a DRC gain application block. These blocks are exercised on a frame-by-frame basis during audio processing. In addition to encoded time-varying gain values, the DRC

gain decoder can also receive parametric DRC metadata for generation of time-varying gain values at the decoder. Various DRC configurations can be conveyed in a separate bitstream element, such as configurations for a downmix or combined DRCs. The DRC set selection block decides based on the playback scenario and the applicable DRC configurations which DRC gains to apply to the audio signal. Moreover, the DRC tool supports loudness normalization based on loudness metadata.

Page 3, Clause 5

Add the following at the end of the clause:

The DRC tool provides support for loudness equalization, or sometimes called “loudness compensation”, that can be applied to compensate for the effect of the playback level on the spectral balance. For this purpose, time-varying loudness information can be recovered from DRC gain sequences to dynamically control the compensation module. While the compensation module is out of scope, the interface describes in which frequency ranges the loudness information should be applied.

A flexible tool for generic metadata-controlled equalization is provided. The tool can be used to reach the desired spectral balance of the reproduced audio signal depending on a wide variety of playback scenarios, such as downmix, DRC, or playback room size. It can operate in the sub-band domain of an audio decoder and in the time domain.

Page 4, 6.1.1

Replace the following list after Table 1:

The static payload is divided into five logical blocks:

- channelLayout()
- downmixInstructions ()
- drcCoefficientsBasic(), drcCoefficientsUniDrc()
- drcInstructionsBasic(), drcInstructionUniDrc()
- loudnessInfo()

With:

The static payload is divided into six logical blocks:

- channelLayout();
- downmixInstructions(), downmixInstructionsV1();
- drcCoefficientsBasic(), drcCoefficientsUniDrc(), drcCoefficientsUniDrcV1();
- drcInstructionsBasic(), drcInstructionUniDrc(), drcInstructionUniDrcV1();
- loudnessInfo(), loudnessInfoV1();
- loudEqInstructions().

Page 4, 6.1.1

Replace the last two paragraphs:

uniDrcConfig() contains all blocks except for the loudnessInfo() blocks which are bundled in loudnessInfoSet(). The last part of the uniDrcConfig() payload can include future extension payloads. In the event that a *uniDrcConfigExtType* value is received that is not equal to UNIDRCCONFEXT\_TERM, the DRC tool parser shall read and discard the bits (otherBit) of the extension payload. Similarly, the last part of the loudnessInfoSet() payload can include future extension payloads. In the event that a *loudnessInfoSetExtType* value is received that is not equal to UNIDRCLOUDEXT\_TERM, the DRC tool parser shall read and discard the bits (otherBit) of the extension payload.

The top level fields of uniDrcConfig() include the audio sample rate, which is a fundamental parameter for the decoding process (if not present, the audio sample rate is inherited from the employed audio codec). Moreover, the top level fields of uniDrcConfig() include the number of instances of each of the logical blocks, except for the channelLayout() block which appears only once. The top level fields of loudnessInfoSet() only include the number of loudnessInfo() blocks. The five logical blocks are described in the following.

With:

uniDrcConfig() contains all blocks except for the loudnessInfo() blocks which are bundled in loudnessInfoSet(). The last part of the uniDrcConfig() payload can include future extension payloads. In the event that a *uniDrcConfigExtType* value is received that is not equal to UNIDRCCONFEXT\_TERM, the DRC tool parser shall read and discard the bits (otherBit) of the extension payload. Similarly, the last part of the loudnessInfoSet() payload can include future extension payloads. In the event that a *loudnessInfoSetExtType* value is received that is not equal to UNIDRCLOUDEXT\_TERM, the DRC tool parser shall read and discard the bits (otherBit) of the extension payload. Each extension payload type in uniDrcConfig() or loudnessInfoSet() shall not appear more than once in the bitstream if not stated otherwise. An extension payload of type UNIDRCCONFEXT\_V1 shall precede an extension payload of type UNIDRCCONFEXT\_PARAM\_DRC in the bitstream if both payloads are present. Note that for ISO/IEC 14496-12, configuration extension payloads are provided according to Table AMD1.26.

The top level fields of uniDrcConfig() include the audio sample rate, which is a fundamental parameter for the decoding process (if not present, the audio sample rate is inherited from the employed audio codec). Moreover, the top level fields of uniDrcConfig() include the number of instances of each of the logical blocks, except for the channelLayout() block which appears only once. The top level fields of loudnessInfoSet() only include the number of loudnessInfo() blocks. The six logical blocks are described in the following.

Page 5, 6.1.2.2

Replace:

### 6.1.2.2 downmixInstructions()

This block includes a unique non-zero downmix identifier (*downmixId*) that can be used externally to refer to this downmix. The *targetChannelCount* specifies the number of channels after downmixing to the target layout. It may also contain downmix coefficients, unless they are specified elsewhere. For use cases where the base audio signal represents objects or other audio content, the *downmixId* can be used to refer to a specific target channel configuration of a present rendering engine.

With:

### 6.1.2.2 downmixInstructions() and downmixInstructionsV1()

This block includes a unique non-zero downmix identifier (*downmixId*) that can be used externally to refer to this downmix. The *targetChannelCount* specifies the number of channels after downmixing to the target layout. It may also contain downmix coefficients, unless they are specified elsewhere. For