**3GPP TSG-SA4 Meeting #134 *S4-251899***

**Dallas, United States, 17th Nov 2025 - 21st Nov 2025**

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| *CR-Form-v12.4* | | | | | | | | |
| **CHANGE REQUEST** | | | | | | | | |
|  | | | | | | | | |
|  | **26.253** | **CR** | **0027** | **rev** | **-** | **Current version:** | **18.6.0** |  |
|  | | | | | | | | |
| *For* [***HE******LP***](http://www.3gpp.org/3G_Specs/CRs.htm#_blank)*on using this form: comprehensive instructions can be found at* [*https://www.3gpp.org/Change-Requests*](https://www.3gpp.org/Change-Requests)*.* | | | | | | | | |
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| ***Proposed change affects:*** | UICC apps |  | ME | **x** | Radio Access Network |  | Core Network | **x** |

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| ***Title:*** | Correcting references to fixed-point specification | | | | | | | | | |
|  |  | | | | | | | | | |
| ***Source to WG:*** | Dolby Laboratories Inc., Ericsson LM, Fraunhofer IIS, Huawei Technologies Co Ltd., Nokia, NTT, Orange, Panasonic Holdings Corporation, Philips International B.V., Qualcomm Incorporated, VoiceAge Corporation | | | | | | | | | |
| ***Source to TSG:*** | S4 | | | | | | | | | |
|  |  | | | | | | | | | |
| ***Work item code:*** | IVAS\_Codec | | | | |  | ***Date:*** | | | 2025-11-11 |
|  |  | | | |  | |  | | |  |
| ***Category:*** | **F** |  | | | | | ***Release:*** | | | Rel-18 |
|  | *Use one of the following categories:* ***F*** *(correction)* ***A*** *(mirror corresponding to a change in an earlier release)* ***B*** *(addition of feature),* ***C*** *(functional modification of feature)* ***D*** *(editorial modification)*  Detailed explanations of the above categories can be found in 3GPP [TR 21.900](http://www.3gpp.org/ftp/Specs/html-info/21900.htm). | | | | | | | | *Use one of the following releases: Rel-8 (Release 8) Rel-9 (Release 9) Rel-10 (Release 10) Rel-11 (Release 11) … Rel-17 (Release 17) Rel-18 (Release 18) Rel-19 (Release 19)  Rel-20 (Release 20)* | |
|  |  | | | | | | | | | |
| ***Reason for change:*** | | The algorithmic description includes some references to fixed-point code and floating-point code where only the floating-point code (TS 26.258) should be referenced. | | | | | | | | |
|  | |  | | | | | | | | |
| ***Summary of change:*** | | Remove references to non-existing fixed-point code or replace with reference to floating-point code, as needed. Editorial corrections following from the corrections of references. | | | | | | | | |
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| ***Consequences if not approved:*** | | Incorrect references to specification that does not exist will remain, which can potentially lead to confusion. | | | | | | | | |
|  | |  | | | | | | | | |
| ***Clauses affected:*** | | 2, 4.1, 4.5.4, 5.3.3.3.6.4.3, 6.6.6.2, 7.6.8 | | | | | | | | |
|  | |  | | | | | | | | |
|  | | **Y** | **N** |  | | | |  | | |
| ***Other specs*** | |  | **x** | Other core specifications | | | | TS/TR ... CR ... | | |
| ***affected:*** | |  | **x** | Test specifications | | | | TS/TR ... CR ... | | |
| ***(show related CRs)*** | |  | **x** | O&M Specifications | | | | TS/TR ... CR ... | | |
|  | |  | | | | | | | | |
| ***Other comments:*** | |  | | | | | | | | |
|  | |  | | | | | | | | |
| ***This CR's revision history:*** | |  | | | | | | | | |

CHANGE 1

# 2 References

[1] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".

[2] 3GPP TS 26.441: "Codec for Enhanced Voice Services (EVS); General Overview".

[3] 3GPP TS 26.445: "Codec for Enhanced Voice Services (EVS); Detailed Algorithmic Description".

[4] 3GPP TS 26.447: "Codec for Enhanced Voice Services (EVS); Error concealment of lost packets".

[5] 3GPP TS 26.448: "Codec for Enhanced Voice Services (EVS); Jitter Buffer Management"

[6] 3GPP TS 26.250: "Codec for Immersive Voice and Audio Services (IVAS); General overview".

[7] (void)

[8] 3GPP TS 26.252: "Codec for Immersive Voice and Audio Services (IVAS); Test Sequences".

[9] 3GPP TS 26.254: "Codec for Immersive Voice and Audio Services (IVAS); Rendering".

[10] 3GPP TS 26.255: "Codec for Immersive Voice and Audio Services (IVAS); Error concealment of lost packets".

[11] 3GPP TS 26.256: "Codec for Immersive Voice and Audio Services (IVAS); Jitter Buffer Management".

[12] 3GPP TS 26.258: "Codec for Immersive Voice and Audio Services (IVAS); C code (floating point)".

CHANGE 2

## 4.1 Introduction

The present document is a detailed algorithmic description of the Immersive Voice and Audio Services (IVAS) coder. The IVAS coder is a framework for low-delay speech- and audio coding and rendering targeting stereo or immersive audio communication. It comprises:

- encoder,

- decoder, and

- renderer.

The procedure of this document is mandatory for implementation in all network entities and User Equipment (UE)s supporting the IVAS coder.

The present document does not describe the C code of the IVAS coder. In the case of discrepancy between the algorithmic description in the present document and its C code specification contained in [12], the C code specification prevails.

CHANGE 3

### 4.5.4 Interface for external rendering

IVAS renderer and its interface provide support to IVAS codec design constraints. The details of the rendering library API are provided in [12] for the floating-point code. The details of the rendering library API are provided in [9].

CHANGE 4

5.3.3.3.6.4.3 2-Stage Split/AVQ Quantizer

The 2-stage Split/AVQ quantizer is used at bitrates above 64 kbps and in any bitrate if the core sampling rate is 16 kHz. It consists of two stages with separate vector quantizers. The first stage vector quantizer is a stochastic split VQ which performs a fixed rate quantization. After the first stage, a residual vector is calculated, and the residual vector is further quantized using an Algebraic Vector Quantizer (AVQ) in the second stage which performs a variable rate quantization. In the joint encoding mode, the first stage is skipped for the side SNS parameters, and the residual is set to zero which results in being directly quantized by the second stage AVQ only, as is depicted in Figure 5.3‑39.



Figure 5.3‑39: SNS parameter VQ encoding for long blocks in both channels

The 1st stage split VQ uses off-line trained stochastic codebooks of dimension 8 with 32 code vectors per codebook. Codebook selection depends on the split number (first split, second split) and the transform block size, see [12] for the definition of the codebooks. To reduce the overall size of the needed codebooks, dependency on the core sampling rate is removed by normalizing the input vectors to the first stage VQ. Normalization is done by subtracting a mean value from each scale parameter in the target vector:

CHANGE 5

#### 6.6.6.2 Codec reconfiguration in bitrate switching

Bitrate switching in ISM format mainly cause reconfiguration of the underlying SCE core-decoders as the bitrate might influence selection of the core-decoder tools to be used. If a bitrate switch causes the ISM mode to change from DiscISM to ParamISM, or the other way around, more reconfiguration is needed as the number of audio channels – and thus also the number of SCE core-decoder instances – changes. Note that this can only happen if the number of objects is 3 or 4, namely when switching from a bitrate that is lower than 48 kbps to a bitrate of 48 kbps or higher (or the other way around). The performed reconfigurations are:

- Re-setting the core-decoder bitrates according to clause 5.6.2.3.1 step 1 and 2. This results in an initial equal bitrate distribution between the SCEs. The remaining steps are skipped.

- Reconfiguration of the SCE core-decoders. This can include deallocation of old SCE core-decoders and/or setting up new ones in case the bitrate switch causes the ISM mode to change. This also includes reconfiguration and initializations of the HP20 output filters. For 3 or 4 objects, the number of transport channels can change when switching from a DiscISM bitrate to a ParamISM bitrate or vice versa. Memories of the high pass filters for persisting channels are not filled with zeros but carried over for signal continuity.

- Selecting the renderer if the ISM mode changes based on [12], clause 5.1. Depending on the output config, changing the renderer can result in reconfiguration of HRTF data or the respective reverb module.

- Reconfiguring the JBM. This is dependent on whether the renderer changes or not, see 6.6.8.

- Initializing or deallocating buffers and structures needed especially for ParamISM if the ISM mode changes. This includes initializing the ParamISM renderer structures and initializing DirAC-related structures if switching to a ParamISM bitrate. Then, also the prototype matrix and the interpolator states are initialized, see clause.

- Reconfiguration of the CLDFB instances to match the current number of transport channels and the core-decoder setup.

CHANGE 6

### 7.6.8 Interface for Split rendering

Split renderer and its interface provide support to ISAR codec design constraints. The details of the split rendering library API are provided in [12] for the floating-point code. The details of the split rendering library API are provided in [9].

END OF CHANGES