TECHNICAL PAPER

ETSI LC3PLUS HIGH RESOLUTION

SPECIFICATION FOR USE AS VENDOR SPECIFIC CODEC VIA BLUETOOTH A2DP
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CONTENT

The LC3plus High Resolution audio coding scheme is a development of Fraunhofer in cooperation with Ericsson AB, pursuant to the specification issued in clause 5.8 of the ETSI TS 103 634.

This document describes how to transmit LC3plus High Resolution bitstream via Bluetooth A2DP by way of signaling a vendor specific codec. The LC3plus High Resolution transport over Bluetooth A2DP described in this document is not part of any Bluetooth SIG specification.

Implementations of the LC3plus High Resolution audio coding scheme are not compliant with any current Bluetooth specification and can not be qualified pursuant to the Bluetooth qualification process and the LC3plus High Resolution codec is not a compliant portion according to Bluetooth SIG PCLA.

For the purpose of clarity, the LC3plus High Resolution mode is a codec defined by ETSI. It is different from the Low Complexity Communication Codec (LC3) specified by Bluetooth. The LC3plus High Resolution mode is not compatible with LC3 specified by Bluetooth.

This documentation doesn’t grant any patent licence for the use of LC3plus patent licenses for necessary patent claims for the LC3plus High Resolution codec (including those of Fraunhofer) may be obtained from the respective patent owners.

For more information regarding licensing of LC3plus, please visit:
https://www.iis.fraunhofer.de/en/ff/amm/lizenz/patent.html

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1 LC3plus HIGH RESOLUTION

1.1 DEFINITION

ETSI defines a dedicated high-resolution mode of LC3plus as defined in ETSI TS 103 634 V1.3.1 (1), clause 5.8, for the coding of audio data with very high precision. Note: the measurable distortion by means of Total Harmonic Distortion and Noise (THD+N) can be less than -130dB measured on an AudioPrecision® reference tool.

Reference implementation and test tools are available as electronic attachment of TS 103 634 (2).

1.2 FEATURES

LC3plus High Resolution has the following configuration parameters:

Parameter	Value
Sample rate	48 kHz, 96 kHz
Frame duration	10 ms, 5 ms, 2.5 ms
Bit rate	Recommended rates: 128 kbps – 672 kbps per channel Fallback rates: Down to 50% of lowest recommended rate
Sample depth	24 bit signed integer or 32 bit IEEE floating point

NOTE: The exact rates depend on sampling rate, frame duration and Max Transport Unit (MTU) of A2DP link. More details provided in Annex A.

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2 LC3plus HIGH RESOLUTION COMPLIANCE

2.1 CODEC REQUIREMENTS

The following section defines the codec support requirements to build compatible products for LC3plus High Resolution as vendor specific codec via Bluetooth.

2.1.1 Common requirements

An encoder and decoder implementation shall at least support:

Parameter	Value
Sample rate	48 kHz, 96 kHz
Number of audio channels	2 (stereo)
Sample depth input/output	24 bit signed integer or 32 bit IEEE floating point
Rate switching	as defined in clause 5.7 of [1]
High-Resolution Audio	As defined in 5.8 of [1], including all required encoder and decoding functionalities of clause 5

2.1.2 Encoder requirements

The encoder implementation shall be compliant to the definitions in TS 103 634, clause 5.8 and shall at least support the following configurations

Parameter	Value
Frame duration	10 ms and 5 ms, optionally 2.5 ms
Payload sizes in bytes per frame and channel	Any number of bytes as defined in Table 5.2 of TS 103 634 (1)

2.1.3 Decoder requirements

The decoder implementation shall be compliant to the definitions in TS 103 634, clause 5.8 and shall at least support the following configurations:

Parameter	Value
Frame duration	At least one of 10 ms or 5 ms, optionally 2.5 ms
Payload sizes in bytes per frame and channel	Any number of bytes as defined in Table 5.2 of TS 103 634 (1)
Packet Loss Concealment	Packet Loss Concealment as described in TS 103 634 clause 5.6.3 with »PLC method selection«, »MDCT frame repetition with sign scrambling« and »Time domain concealment«. Whenever PLC method »Frequency domain concealment« is chosen by the »PLC method selection«, method »MDCT frame repetition with sign scrambling« is selected instead.

2.2 CONFORMANCE REQUIREMENTS

Encoder and decoder shall pass conformance requirements defined in TS 103 634 (1), clause 7.3.5.

Test scripts are available in the software package of TS 103 634 (2).

2.3 PRECISION REQUIREMENTS

Encoder and decoder shall pass the requirements on implementation precision defined in TS 103 634 (1), clause 7.3.5.4.

The measured THD+N value shall be lower or equal, and SNR value shall be higher or equal to

- 120 dB (THD+N) / 120 dB (SNR) at 1 kHz tone
- 110 dB (THD+N) / 110 dB (SNR) as worst case value over all measured frequencies

Test scripts are available in the software package of TS 103 634 (2).

3 SIGNALING AND TRANSPORT AS VENDOR SPECIFIC CODEC VIA BLUETOOTH

3.1 BLUETOOTH CLASSIC: A2DP

3.1.1 Overview

This section defines signaling and transport of LC3plus High Resolution audio coding format over the Bluetooth Advanced Audio Profile (A2DP) (3). The format is signaled as vendor specific coding format via Fraunhofer's company ID.

3.1.2 Terms

This specification uses the following terms in accordance with A2DP (3).

RFA: reserved for future additions. Bits with this designation shall be set to zero. Receivers shall ignore these bits.

SRC: Source – A device is the **SRC** when it acts as a source of a digital audio stream that is delivered to the **SNK** of the piconet.

SNK: Sink - A device is the SNK when it acts as a sink of a digital audio stream delivered from the SRC on the same piconet.

This specification uses furthermore the following terms in accordance with TS 103 634 (1):

Frame data: encoded media for a single audio frame and a single channel, either output from the encoder or input to the decoder.

Frame data block: frame data for one or more channels for a single frame period.

FBHR: full band high resolution which stands for a sampling rate of 48 kHz, an audio bandwidth of 24 kHz and enabled High Resolution mode for LC3plus.

UBHR: ultra band high resolution which stands for a sampling rate of 96 kHz, an audio bandwidth of 48 kHz and enabled High Resolution mode for LC3plus.

NOTE: For mono input audio signals, a frame data block includes the frame data for a single audio frame, see frame data. In this case, the frame data block is identical to the frame data. For stereo and multi-channel input audio signals, a frame data block contains the frame data from all channels. Thereby, a frame data block includes the same number of frames as there are channels.

3.1.3 Codec Specific Information

Table 1 shows Codec Specific Information (CSI) Elements for LC3plus High Resolution used in the signaling procedures. For reference, see AVDTP (4) section 8.19.5.

The following section defines the field values and their requirements. Support columns in each field value show the requirements to fulfill when this codec is supported. If the packet includes improper settings, the error code shall be returned as specified in AVDTP (4) Section 5.1.3.

NOTE: In the Get All Capabilities Response of AVDTP, one or more bits may be defined/set in each field. On the other hand, in the Set Configuration Command and the Reconfigure Command of AVDTP, only one bit shall be defined/set in each field.

<table>
<tr><th>Bit 7</th><th>Bit 6</th><th>Bit 5</th><th>Bit 4</th><th>Bit 3</th><th>Bit 2</th><th>Bit 1</th><th>Bit 0</th><th></th></tr>
<tr><td colspan="8">Vendor ID</td><td>Octet 0
Octet 1
Octet 2
Octet 3</td></tr>
<tr><td colspan="8">Vendor Specific Codec ID</td><td>Octet 4
Octet 5</td></tr>
<tr><td colspan="4">Frame Durations</td><td colspan="4">RFA</td><td>Octet 6</td></tr>
<tr><td colspan="8">Audio Channel Count</td><td>Octet 7</td></tr>
<tr><td colspan="8">Sampling Frequency and Resolution</td><td>Octet 8
Octet 9</td></tr>
</table>

Table 1: Codec Specific Information Elements for LC3plus High Resolution.

3.1.3.1 Vendor ID

A2DP specifies:
A 32-bit Vendor ID shall be used. The lower 16 bits of the 32-bit Vendor ID shall contain a valid, non- reserved 16-bit Company ID as defined in Bluetooth Assigned Numbers (5). The upper 16 bits of the 32- bit Vendor ID shall be set to zero.
The LSB of the Vendor ID shall be placed in octet 0.

The Fraunhofer IIS Company ID is **0x08A9**, therefore the A2DP Vendor ID is **0x000008A9**. Table 2 shows the corresponding octet values for use in the CSI.

Octet 0	Octet 1	Octet 2	Octet 3
0xA9	0x08	0x00	0x00

Table 2: Fraunhofer Vendor ID octet values.

3.1.3.2 Vendor Specific Codec ID LC3plus High Resolution

The Codec ID for LC3plus High Resolution is 0x0001 which indicates channel controlled variable bit rate support. Table 3 shows the corresponding octet values for use in the CSI.

Octet 4	Octet 5
0x01	0x00

Table 3: LC3plus High Resolution Codec ID octet values.

3.1.3.3 Frame Durations

Table 4 shows the values of the Frame Durations field for LC3plus High Resolution. The **SRC** shall support the 10 ms and the 5 ms frame duration. The **SNK** shall support at least the 5 ms or the 10 ms frame duration. The 2.5 ms frame duration is optional in SRC and SNK.

NOTE: Section 3.1.6 further describes mandatory transport configurations, depending on the selected Frame Data Length of the audio streaming session.

Position	Frame Data Length	Support in SRC	Support in SNK
Octet 6; b7	RFA	-	-
Octet 6; b6	10 ms codec frames	M	C1
Octet 6; b5	5 ms codec frames	M	C1
Octet 6; b4	2.5 ms codec frames	O	O
C1: At least one of the values shall be supported			

Table 4: Frame Durations for LC3plus High Resolution.

3.1.3.4 Audio Channel Counts

Table 5 shows the value of the Audio Channel Counts field for LC3plus High Resolution. The **SNK** and the **SRC** shall support 2 channels (stereo). 1 channel (mono) support is optional for **SNK** and **SRC**.

Position	Audio Channel Counts	Support in SRC	Support in SNK
Octet 7; b7	1	O	O
Octet 7; b6	2	M	M
Octet 7; b5-b0	RFA	-	-

Table 5: Audio Channel Counts for LC3plus High Resolution.

3.1.3.5 Sampling Frequency and Resolution LC3plus High Resolution

Table 6 shows of value of the Sampling Frequency and Resolution field for LC3plus High Resolution. The **SNK** as well as the **SRC** shall support FBHR and UBHR.

Position	Sampling Frequency and Resolution	Support in SRC	Support in SNK
Octet 8; b7	RFA	-	-
Octet 8; b6	RFA	-	-
Octet 8; b5	RFA	-	-
Octet 8; b4	RFA	-	-
Octet 8; b3	RFA	-	-
Octet 8; b2	RFA	-	-
Octet 8; b1	RFA	-	-
Octet 8; b0	FBHR (48000 Hz, High Resolution)	M	M
Octet 9; b7	UBHR (96000 Hz, High Resolution)	M	M
Octet 9; b6-0	RFA		

NOTE: If audio content with sample rates higher than 48kHz is available, UBHR is to be selected.

Table 6: Sampling Frequency and Resolution for LC3plus High Resolution.

3.1.4 Media Packet Header Requirements

3.1.4.1 General
The media packet header, also known as Real-time Transport Protocol (RTP) header, is specified in IETF RFC 3550 (6). This payload format uses the fields of the RTP header in a manner consistent with IETF RFC 3550 (6).

3.1.4.2 Marker Bit
RFA.

3.1.4.3 Sequence Number
The RTP Sequence Number is incremented by 1 for each transmitted packet, as described in IETF RFC 3550 (6).

3.1.4.4 Timestamp
The RTP clock rate for the LC3plus High Resolution codec is defined based on the Sampling Frequency and Resolution value (see section 3.1.3.5) selected via the Set Configuration or the Reconfigure Command of AVDTP for the active stream.

Streams using the FBHR or UBHR Sampling Frequency and Resolution use an RTP clock rate of 96 000 (Hz). **Table 7** defines the Time Stamp Increment (TSI) between consecutive frame data blocks for such streams:

Parameter	Value
TSI for 2.5 ms frame duration	240
TSI for 5 ms frame duration	480
TSI for 10 ms frame duration	960

NOTE: The use of Fragmentation depends on the negotiated Frame Duration Length, as specified in Section 3.1.6.

Table 7:Time Stamp Increments per frame duration.

If a media payload consists of multiple LC3plus High Resolution frame data blocks, the Timestamp (TS) of the media packet header represent the TS of the first LC3plus High Resolution frame data block. The TS of the following LC3plus High Resolution frame data blocks shall be calculated using the TSI and the number of frame data blocks.

When an LC3plus High Resolution frame data block is fragmented into multiple media packets, all packets that make up a fragmented LC3plus High Resolution frame data block shall use the same TS.

3.1.5 Media Payload Format

LC3plus High Resolution is a single-channel codec. Any stereo or multi-channel coding is supported by operating one encoder or decoder instance per channel. An overview showing channel ordering, encoding of multiple Audio Channels, and multiplexing of the data for a stereo example is shown in **Figure 1**.

All per-Channel frame data that make up one LC3plus High Resolution frame data block must have the same length, i.e. identical bit rate. The **SRC** shall enforce that the encoding bit rate of all encoder instances is synchronized within one LC3plus High Resolution frame data block.

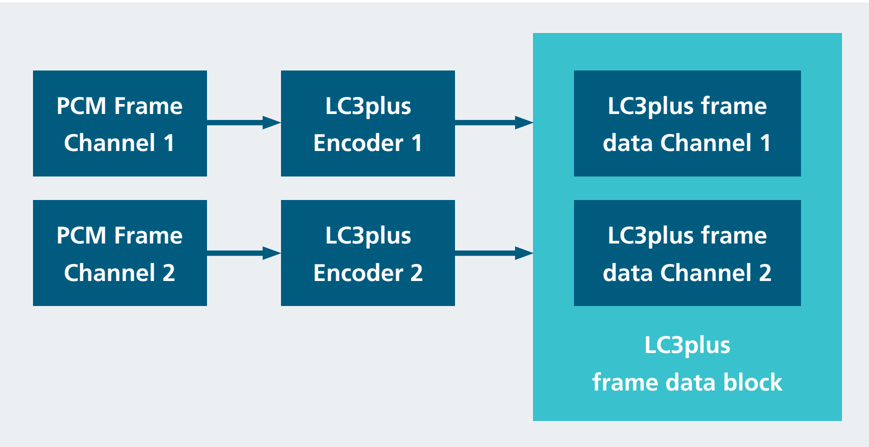

Figure 1: Multi-Channel LC3plus High Resolution frame data multiplexing order.

The media payload for LC3plus High Resolution shown in **Figure 2** always starts with an 8-bit LC3plus High Resolution specific header followed by LC3plus High Resolution frame data block(s).

If the configured MTU size for the transport channel is greater or equal to the LC3plus High Resolution frame data block size + the sum of [Media Payload header size, Content Protection header size (if Content Protection is selected), Media Packet header size], then a media payload shall contain an integral number of complete LC3plus High Resolution frame data blocks (see format (a) shown in **Figure 2**).

If the configured MTU size is less than the LC3plus High Resolution frame data block size + the sum of [Media Payload header size, Content Protection header size (if Content Protection is selected) and provided that the multiplexing service of AVDTP is not selected, the LC3plus High Resolution frame data block shall be fragmented across several media payloads (see format (b) shown in **Figure 2**). All fragmented packets, except the last one, shall have the same total data packet size. If the multiplexing service of AVDTP is selected, it is recommended not to fragment across several media payloads as AVDTP already supports fragmentation across several L2CAP packets.

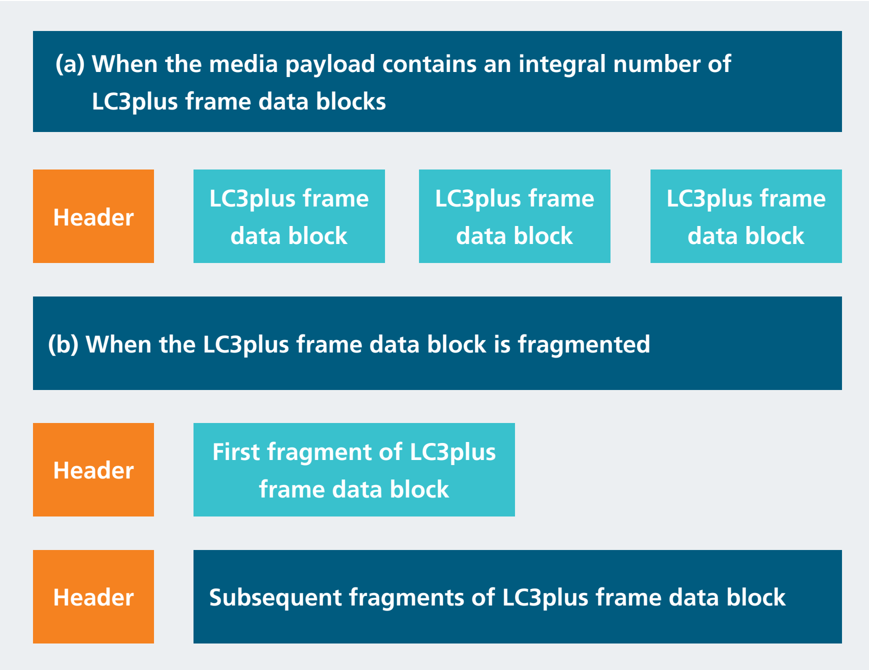

Figure 2: Media payload format of LC3plus High Resolution.

The LC3plus High Resolution media payload header is outlined in **Figure 3**.

7	6	5	4	3	2	1	0	
F	**S**	**L**	**RFA**	**Number of frame data blocks**				**Octet 0**

Figure 3: Header format of media payload for LC3plus High Resolution.

with:

- F bit – Set to 1 if the LC3plus High Resolution frame data block is fragmented, otherwise set to 0.
- S bit – Set to 1 for the starting packet of a fragmented LC3plus High Resolution frame data block, otherwise set to 0.
- L bit – Set to 1 for the last packet of a fragmented LC3plus High Resolution frame data block, otherwise set to 0.
- RFA.
- Number of frame data blocks (4 bits)
  - If the F bit is set to 0, this field indicates the number of frame data blocks contained in this packet.
  - If the F bit is set to 1, this field indicates the number of remaining fragments, including the current fragment. Thus, the last counter value shall be one. For example, if there are three fragments then the counter has value 3, 2 and 1 for subsequent fragments. This field is expressed by 4 bit UiMsbf.

When aggregating multiple LC3plus High Resolution frame data blocks into one media payload (F bit set to 0, Number of frame data blocks set larger than 1), the **SRC** must ensure that all frame data blocks within the media payload are of equal length.

As a consequence, the **SRC** must ensure that the used bit rate for all frame data transported within one media payload is identical.

The **SRC** must ensure that bit rate switches always coincide with media payload boundaries.

3.1.6 Restrictions of codec and transport configurations

A **SRC** using the 10 ms frame duration for an audio streaming session shall be prepared to use Fragmentation. Note that otherwise, it might not be possible to transmit higher bit rates supported by the LC3plus High Resolution encoder depending on the Maximum Transmission Unit (MTU) size of the L2CAP connection.

A **SRC** shall not send fragmented packets when using either the 5 ms or the 2.5 ms frame duration.

A **SNK** supporting the 10 ms frame duration shall be prepared to receive fragmented RTP packets and defragment accordingly.

A **SNK** that does not support the 10 ms frame duration may not support defragmentation.

A **SRC** may change the media payload configurations according to the codec's bit rate.

A **SNK** must be prepared to adapt to changing media payload configurations on a payload-by-payload basis.

When aggregating multiple frame data blocks into one media payload, a **SRC** shall only aggregate up to 20 ms of frame data blocks.

4 BIBLIOGRAPHY

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ANNEX A
A2DP TRANSPORT CONSIDERATIONS

Depending on the MTU of the L2CAP session, the SRC should dynamically decide how to packetize the LC3plus High Resolution frame data blocks into one or multiple media payloads.

The recommended L2CAP MTU for an LC3plus High Resolution session should correspond to the 5-Slot Packet payload size of the L2CAP connection (e.g. DH5, 2-DH5 or 3-DH5). This maximizes the efficiency of the proposed bit rate and transport configurations. This recommendation assumes sending MTU-sized L2CAP SDUs is the most efficient use of the L2CAP connection.

A SRC should constantly monitor the throughput of the L2CAP connection and update the bit rate of the encoder as well as the transport configuration accordingly. The bit rate of the encoder should be reduced in order to avoid any congestion.

Table 8 contains a recommended set of transport configurations for a specific encoder configuration (2 Channels, 10ms Frame Duration, 96 kHz) and a specific MTU (1005 in this example).

NOTE: The following examples assume that the multiplexing service of AVDTP is not selected.

NOTE: The MTU in the tables refers to the L2CAP MTU without L2CAP Header. A2DP Content Protection is assumed to be disabled. This means for an example MTU of 1005 bytes, only 993 bytes are available for the media payload, because 12 bytes are occupied by the AVDT media packet header.

Static encoder configuration			MTU	Bit rate per channel per second	Frame data blocks per media payload	Fragments per Frame data block
Audio Channel Count	**Frame Duration [ms]**	**Sampling Frequency and Resolution**				
2	10	UBHR	1005	500000	<1	2
2	10	UBHR	1005	396800	1	-
2	10	UBHR	1005	198400	2	-
2	10	UBHR	1005	149600	2	-
2	10	UBHR	1005	112000	2	-
2	10	UBHR	1005	74400	2	-

Table 8: Example transport configurations for MTU=1005, Audio Channel Count=2, Frame Duration=10 ms, Sampling Frequency and Resolution=UBHR.

The first row contains a configuration that uses the highest supported encoder bit rate to achieve maximum audio quality, i.e. frame data of 625 bytes per channel which corresponds to 500 kbps. For stereo support, the values are doubled, i.e. 1250 bytes which exceeds the number of bytes available (993), therefore the frame data block must be fragmented over two media payloads.

The configurations in the second and third row are optimized in order to fill the MTU as efficiently as possible. The configuration in the second row fills up the MTU with **one** frame data block, the third row uses **two** frame data blocks and **so on**, until the upper limit of 20ms time per packet is reached, as specified in section 3.1.6.

The fourth row contains the lowest recommended encoder bit rate, while the bottom two configurations are considered as fallback modes which should only be used temporarily under challenging channel conditions.

In comparison to the previous configuration, **Table 9** describes a set of example configurations for the 5ms frame duration. For this frame duration, up to 4 frame data blocks may be aggregated into one media payload without violating the upper 20ms boundary.

Static encoder configuration			MTU	Bit rate per channel per second	Frame data blocks per media payload	Fragments per Frame data block
Audio Channel Count	**Frame Duration [ms]**	**Sampling Frequency and Resolution**				
2	5	FBHR	1005	600000	1	-
2	5	FBHR	1005	396800	2	-
2	5	FBHR	1005	264000	3	-
2	5	FBHR	1005	198400	4	-
2	5	FBHR	1005	148800	4	-
2	5	FBHR	1005	110400	4	-
2	5	FBHR	1005	73600	4	-

Table 9: Example transport configurations for MTU=1005, Audio Channel Count=2, Frame Duration=5 ms, Sampling Frequency and Resolution=FBHR.

Table 10 also contains a set of configurations for the 5 ms frame duration, but in combination with a smaller MTU (679). As specified in section 3.1.6, fragmentation is not allowed in combination with the 5 ms frame duration. Therefore, the maximum encoder bit rate gets limited to 532800 bps per channel for this combination of MTU and encoder configuration.

Static encoder configuration			MTU	Bit rate per channel per second	Frame data blocks per media payload	Fragments per Frame data block
Audio Channel Count	Frame Duration [ms]	Sampling Frequency and Resolution				
2	5	FBHR	679	532800	1	-
2	5	FBHR	679	265600	2	-
2	5	FBHR	679	177600	3	-
2	5	FBHR	679	148800	3	-
2	5	FBHR	679	110400	4	-
2	5	FBHR	679	73600	4	-

Table 10: Example transport configurations for MTU=679, Audio Channel Count=2, Frame Duration=5 ms, Sampling Frequency and Resolution=FBHR.

ANNEX B
OVERVIEW CODEC ID

The following table lists the Codec IDs for the Fraunhofer Vendor ID (0x08A9):

Codec ID	Description
0x0001	LC3plus High Resolution Frame duration: 10ms, 5ms, 2.5ms Channel controlled variable bit rate support
0x0002	LC3plus High Resolution Frame duration: 10ms, 5ms, 2.5ms Constant bit rate support

Table 11: Overview Fraunhofer Codec ID

VERSION HISTORY

V.1.0.4 2023-04 Editorial changes in disclaimer
V.1.0.3 2023-03 Corrected wording in 4 for Static encoder configuration
V.1.0.2 2023-02 Corrected wording in 2.3 for THD+N/SNR thresholds
V.1.0.1 2022-06 Corrected wording in 3.1.4.4 for calculation of next TS
V.1.0.0 2021-11 Initial Release

ABOUT FRAUNHOFER IIS

Fraunhofer IIS, based in Erlangen, Germany, is the largest institute within Fraunhofer-Gesellschaft, Europe's leading application-oriented research organization.

For over 30 years, the institute's Audio and Media Technologies division has been shaping the globally deployed standards and technologies in the fields of audio coding and moving picture production. Fraunhofer IIS systems and tools help create, transmit and provide excellent audio and video content as well as enable high-quality real-time communication. Today, almost all computers, mobile phones and consumer electronic devices are equipped with technologies from Erlangen and are used by billions of people around the world every day.

It all started with the creation of mp3, then evolved with the co-development of AAC and HE-AAC. Now the fourth generation of best-in-class audio technologies – MPEG-H Audio, EVS, LC3/LC3plus and xHE-AAC – elevates the media experience to new heights. In terms of audio signal processing, Symphoria and the Sonamic product family provide enveloping and enhanced sound in cars, while the upHear product family dramatically improves 3D audio playback or recording quality of professional and consumer devices. Fraunhofer technologies also power digital radio: first and foremost in the form of the ContentServer, combining audio encoding, multimedia data management and multiplexing. In the field of moving picture technologies, establishing the Digital Cinema Initiative test plan boosted the creation of professional tools for digital film and media production, such as easyDCP, Realception and JPEG XS.

The interdisciplinary team transforms science into best-in-class applications with new functionalities for end users as well as optimum efficiency, reducing transmission costs while increasing reliability. Always taking into account the demands of the market, Fraunhofer IIS develops technology that makes memorable moments.

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