Fraunhofer IIS

TECHNICAL PAPER

ETSI LC3PLUS

SPECIFICATION FOR USE AS VENDOR SPECIFIC CODEC VIA BLUETOOTH A2DP AND BLUETOOTH LE AUDIO
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The LC3plus audio coding scheme is a development of Fraunhofer in cooperation with Ericsson AB, pursuant to the specification issued in the ETSI TS 103 634.

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

Implementations of the LC3plus 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 codec is not a compliant portion according to Bluetooth SIG PCLA.

For the purpose of clarity, LC3plus is a codec defined by ETSI. It is different from the Low Complexity Communication Codec (LC3) specified by Bluetooth. LC3plus 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 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

1 LC3plus

1.1 DEFINITION

LC3plus is an audio coding scheme standardized by ETSI in TS 103 634 (1). This document exposes two of its operating modes, see Table 1. They produce different bitstreams that are not compatible with each other. Reference implementation and test tools are available as electronic attachment of TS 103 634 (2).

Operating mode	Scope	Defined in TS 103 634 (1)
Normal, hrmode 0	16, 24 and 32 kHz, 5 ms frame duration, lossy coding from 32 kbit/s, over LE Audio only	Table 5.1
Lossless CBR, hrmode 2	44.1, 48, 96 and 192 kHz, 16 and 24 bit, switching per frame between lossless and High Resolution coding to stay within a bit rate limit	clause 5.10, Table 5.2a

Table 1: Operating modes of LC3plus as exposed by this document.

Lossless CBR mode is the lossless mode of (1), clause 5.10, with a bit rate limit. The encoder switches dynamically, frame by frame, between lossless coding and lossy High Resolution coding, depending on the bits a frame needs and the limit ((1), clause 5.10.1). Sections 3.1.5.8 and 3.2.7 state the limit for each transport. The signaling of section 3 carries the operating mode as the value of the hrmode parameter with which the encoder and the decoder of (2) are initialized, 0 for Normal mode and 2 for Lossless CBR mode; an encoder and a decoder **shall** be initialized with that value. The bitstream does not carry it. High Resolution alone is specified under Codec ID 0x0001 (20), (21).

1.2 FEATURES

Table 2 states the features of the codec.

Property	Value
Sampling frequencies	8, 16, 24, 32, 44.1, 48, 96 and 192 kHz, depending on the operating mode. Table 1 states which of them this document exposes.
Frame durations	1.25, 2.5, 5, 7.5 and 10 ms in every operating mode, except 1.25 ms at 8 kHz. This document carries 1.25 ms over neither transport; sections 3.1.3.3 and 3.2.3.1 state the frame durations it carries.

Property	Value
Audio channels	Each channel is coded separately.
Sample resolution	16 and 24 bit integer in Lossless CBR mode, which supports no other resolution ((1), clause 5.10.2.1.1). The Normal mode bitstream carries no sample resolution; this document uses 16 bit at the encoder input and the decoder output in that mode.
Audio bandwidth	An encoder **may** limit the audio bandwidth in Normal mode ((1), clause 5.2.6); the bitstream carries it, so it needs no signaling.
Bit rates	32 kbit/s to 3267.2 kbit/s per channel in the range this document exposes, depending on operating mode, sampling frequency, frame duration, sample resolution and transport.
Error concealment	Part of the standard and mandatory for decoders ((1), clause 5.6).
Error protection	Optional channel coder with forward error correction, not exposed by this document.

Table 2: Features of LC3plus.

1.3 CONVENTIONS

The words "shall", "should" and "may" carry the meaning A2DP (3), section 1.5.1 gives them. In the support columns **M** means mandatory to support, **O** optional to support, and a dash no requirement. A label such as C.1 names a condition that section 3.2.8 states; **Excluded** means that the configuration is not used where the condition fails.

1.4 TYPICAL APPLICATIONS (INFORMATIVE)

Table 3 names, for typical applications, the LC3plus settings of section 3 that serve them; where an application profile applies, its mandatory settings stay LC3 (section 3.2.11).

Application	Setting	Bit rate per channel	Role	Section
Music	48_1_LC3plus_Lossless 96_1_LC3plus_Lossless 441_2_LC3plus_Lossless	128 to 176.4 kbit/s	Unicast Client to Sink ASE	3.2.8.2
Music with the Hi-Res Audio Wireless logo of the Japan Audio Society (22)	A2DP at 96000 Hz and 24 bit, bit rate limit of 563 octets or more per frame at 10 ms	450.4 kbit/s or more at 10 ms	SRC to SNK	3.1.6.1
Gaming	32_3_LC3plus_Normal_LowLatency 48_5_LC3plus_Lossless_LowLatency	76.8 to 192 kbit/s	Unicast Client to Sink ASE	3.2.9, Annex B.1
Microphone and return channel	16_3_LC3plus_Normal_LowLatency 24_3_LC3plus_Normal_LowLatency 32_3_LC3plus_Normal_LowLatency	38.4 to 76.8 kbit/s	Source ASE to Unicast Client	3.2.9
Hearing aids	16_3_LC3plus_Normal_LowLatency, in addition to the LC3 settings of the profile; the only setting that fits a pair with one stream per ear at a flush timeout of 1	38.4 kbit/s	Unicast Client to the Sink ASE of each hearing aid	3.2.11
Broadcast	32_3_LC3plus_Normal_Broadcast 48_1_LC3plus_Lossless_Broadcast 48_5_LC3plus_Lossless_Broadcast_LowLatency	76.8 to 192 kbit/s	Broadcast Source to Broadcast Sink	3.2.10.3
TV sound	48_1_LC3plus_Lossless 48_5_LC3plus_Lossless	128 to 192 kbit/s	Unicast Client or Broadcast Source to the sinks	3.2.8.2, 3.2.10.2
Multi-channel	32_3_LC3plus_Normal_LowLatency in two stereo streams of one CIG 48_1_LC3plus_Lossless in a BIG of six BISes at RTN 0	76.8 to 128 kbit/s	Unicast Client or Broadcast Source to the sinks	Annex B.1, Annex B.2

Table 3: Typical applications and the LC3plus settings that serve them, informative. The role column names the sending role first, then the receiving role; the section column names the section that carries the settings.

2 LC3plus COMPLIANCE

2.1 CODEC REQUIREMENTS

2.1.1 Common requirements

An implementation of LC3plus **shall** conform to ETSI TS 103 634 (1) for every operating mode, sampling frequency and frame duration it exposes in the signaling of section 3.

At 44100 Hz a frame holds as many samples as at 48000 Hz ((1), clause 5.2.2); its period is 10.884, 8.163, 5.442, 2.721 and 1.361 ms for the nominal 10, 7.5, 5, 2.5 and 1.25 ms. An implementation **shall** derive timing from that period, and a decoder **shall** take the sampling frequency from the signaling.

2.1.2 Encoder requirements

An encoder **shall** support the configurations of Table 4.

Parameter	Value
Operating mode	At least one of Normal mode (hrmode 0, over LE Audio only) and Lossless CBR mode (hrmode 2), initialized with the hrmode value of the signaling (sections 3.1.3.8 and 3.2.4.3).
Frame duration	Normal mode: 5 ms. Lossless CBR mode: 10 ms; 7.5 ms, 5 ms and 2.5 ms optional, 2.5 ms over A2DP only, 5 ms mandatory for a SRC over A2DP (section 3.1.3.3).
Sample resolution	Lossless CBR mode: 16 bit and 24 bit (sections 3.1.3.6 and 3.2.3.3).
Rate switching	As defined in clause 5.7 of (1).
Lossless CBR mode	Operation with a bit rate limit and the padding option of clause 5.10.2.1.17 of (1).

Table 4: Encoder requirements.

Under the bit rate limit a frame on the lossless path can need fewer octets than the limit. The padding option of clause 5.10.2.1.17 of (1) fills such a frame with octets of the value 0x07 up to the limit, so that every frame has the same length and the bit rate stays constant. The decoder removes these octets before it decodes the frame, as clause 5.10.2.2.1 of (1) states, and they do not change the decoded audio. A source **may** use the option, send the frame without padding, or fill it with the same pattern to any length in between. It **shall not** send a frame on the lossless path shorter than the lowest value of section 3.3.2 for the combination. Without padding the frames of the audio channels of one frame period can differ in length, each at or below the limit. Sections 3.1.5.1 and 3.2.7 state what each transport adds.

An encoder **shall not** use the signal-shifting option of clause 5.10.2.1.2 of (1). An encoder in Lossless CBR mode **shall not** use the LFE mode of clause 5.3.15 of (1). An encoder **shall** keep the codec frame of every audio channel at or below the bit rate limit, also where one encoder instance codes two audio channels.

2.1.3 Decoder requirements

A decoder **shall** support the configurations of Table 5.

Parameter	Value
Operating mode	Every one the decoder exposes, initialized with the hrmode value of the signaling (sections 3.1.3.8 and 3.2.4.3).
Sampling frequency, frame duration	Every one the decoder exposes.
Rate switching	As defined in clause 5.7 of (1).
Packet loss concealment	As defined in (1), clause 5.6 and, in Lossless CBR mode, clause 5.8.2.
Lossless CBR mode	Switching per frame between the lossless, the high-resolution and the verbatim path of clause 5.10.1 of (1); removal of the padding pattern (clause 5.10.2.2.1); 16 and 24 bit.

Table 5: Decoder requirements.

A decoder **shall** implement the lossless path bit-exact, as clause 5.10.2.2 of (1) requires.

2.2 CONFORMANCE REQUIREMENTS

Encoder and decoder **shall** pass the conformance tests of TS 103 634 (1), clause 7.3, for the operating modes they expose, with the test scripts of (2); in Lossless CBR mode the tests with a bit rate limit.

2.3 PRECISION REQUIREMENTS

An encoder and a decoder of Lossless CBR mode **shall** each pass the precision tests of TS 103 634 (1), clause 7.3.5.4, run in Lossless CBR mode at 24 bit with the padding option enabled, at every sampling frequency and frame duration they expose, with the bit rate limits of Table 6, against the fixed point reference executable of (2). The THD+N value **shall** be at most –120 dB and the SNR at least 120 dB at the 1 kHz tone, and at most –110 dB and at least 110 dB at every measured frequency.

Sampling frequency	10 ms	7.5 ms	5 ms	2.5 ms
44.1 kHz	625 (459.4)	510 (499.8)	408 (599.8)	228 (670.3)
48 and 96 kHz	625 (500)	468 (499.2)	375 (600)	210 (672)
192 kHz	2040 (1632)	1530 (1632)	1020 (1632)	510 (1632)

Table 6: Bit rate limits of the precision tests, octets per codec frame, with the bit rate in kbit/s in parentheses.

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 the LC3plus audio coding format over the Bluetooth Advanced Audio Distribution Profile (A2DP) (3). The format is signaled as vendor specific coding format via Fraunhofer's Company ID, in a Media Codec Capabilities service capability with Media Type Audio and Media Codec Type 0xFF (Assigned Numbers (5), section 6.5.1).

Section 3.1.5.8 states the range of bit rates over A2DP, section 3.3.2 the operating points, section 3.1.6.1 the bit rates for the Hi-Res Audio Wireless logo and Annex A.1 typical configurations.

3.1.2 Terms

This document uses the following terms in accordance with A2DP (3) and AVDTP (4).

RFA: reserved for future additions. Bits with this designation **shall** be set to zero, and receivers **shall** ignore them. A code point with this designation **shall not** be sent, and a receiver **shall** reject the structure that carries it; A2DP (3), section 1.5.2 only recommends the rejection. Section 3.2 writes RFU, reserved for future use, with the same requirements.

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 document also uses 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.

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 7 shows the Codec Specific Information (CSI) Elements for LC3plus used in the signaling procedures; for reference see AVDTP (4), section 8.21.5. The CSI **shall** be 12 octets long, octet 0 to octet 11, so the Length Of Service Capability is 14. A receiver **shall** ignore octets beyond octet 11. The following sections define the field values and their requirements. Support columns in each field value show the requirements to fulfill when this codec is supported. In the Get Capabilities and Get All Capabilities responses, at least one bit **shall** be set in each of octet 6 to octet 9 and in octet 11. A device **shall** announce every value this document makes mandatory for its role, and a SNK

also at least one of 10 ms and 5 ms. A device **shall** support every combination of the values it announces in octet 6 to octet 9 and in octet 11. In the Set Configuration and Reconfigure commands exactly one bit **shall** be set in each of octet 6 to octet 9 and in octet 11; octet 10 follows section 3.1.3.7. If a Set Configuration or a Reconfigure command includes improper settings, the error code **shall** be returned as specified in A2DP (3), section 5.1.3. The improper settings are at least those of Table 8. A set RFA bit and a set feature bit are not improper settings.

Octet	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
Octet 0 to 3: Vendor ID, 0xA9 0x08 0x00 0x00								
Octet 4 to 5: Vendor Specific Codec ID, 0x03 0x00								
6	RFA	10 ms	5 ms	2.5 ms	7.5 ms	RFA	RFA	RFA
7	1 ch	2 ch	RFA	RFA	RFA	RFA	RFA	RFA
8	44.1 kHz	48 kHz	96 kHz	192 kHz	RFA	RFA	RFA	RFA
9	16 bit	24 bit	RFA	RFA	RFA	RFA	RFA	RFA
10	Version				Features			
11	RFA	RFA	RFA	RFA	RFA	Lossless CBR	RFA	RFA

NOTE: This document uses Vendor Specific Codec ID 0x0003. Codec ID 0x0001 identifies LC3plus High Resolution as specified in (20).

Table 7: Codec Specific Information Elements for LC3plus.

Improper setting	Error code	Defined in
CSI shorter than 12 octets.	0x11, BAD_LENGTH	AVDTP (4)
Octet 0 to 5 carry a Vendor ID or Vendor Specific Codec ID the receiver does not support.	0xC2, NOT_SUPPORTED_CODEC_TYPE	A2DP (3)
Octet 8 carries none or more than one of its defined bits.	0xC3, INVALID_SAMPLING_FREQUENCY	A2DP (3)
Octet 8 carries a sampling frequency the receiver does not support.	0xC4, NOT_SUPPORTED_SAMPLING_FREQUENCY	A2DP (3)
Octet 6, octet 7, octet 9 or octet 11 carries none or more than one of its defined bits.	0xE2, INVALID_CODEC_PARAMETER	A2DP (3)

Improper setting	Error code	Defined in
Octet 6, octet 7, octet 9 or octet 11 carries a value the receiver does not support.	0xE3, NOT_SUPPORTED_CODEC_PARAMETER	A2DP (3)
Octet 10 carries the version 0x0.	0xE2, INVALID_CODEC_PARAMETER	A2DP (3)
Octet 10 carries a version above the announced one.	0xE3, NOT_SUPPORTED_CODEC_PARAMETER	A2DP (3)

Table 8: Error codes for the improper settings of section 3.1.3.

Over A2DP, LC3plus **shall** be used in Lossless CBR mode only (octet 11, b2). No field of the CSI carries a bit rate limit; the SRC selects it inside the range of section 3.1.5.8.

3.1.3.1 Vendor ID

The Vendor ID is a 32 bit field. The lower 16 bits **shall** contain a valid, non-reserved Company ID as assigned by the Bluetooth SIG in Assigned Numbers (5), section 7.1, the upper 16 bits **shall** be zero, and the least significant octet **shall** be placed in octet 0. The Company ID of Fraunhofer IIS is 0x08A9, so octets 0 to 3 **shall** carry the values 0xA9, 0x08, 0x00, 0x00.

3.1.3.2 Vendor Specific Codec ID

The Vendor Specific Codec ID of LC3plus as defined in this document is **0x0003**, with the least significant octet in octet 4, so octets 4 and 5 **shall** carry the values 0x03, 0x00. Annex C lists all Vendor Specific Codec IDs administered by Fraunhofer IIS. A receiver **shall** check the Vendor Specific Codec ID before it evaluates the remaining octets and **shall** take the field layout from that ID.

NOTE: Under Codec ID 0x0001, octet 8 and octet 9 form one Sampling Frequency and Resolution field, and octet 9 bit 7 means 96000 Hz High Resolution. A parser for 0x0001 applied to this layout misreads a 16 bit configuration of this document as 96000 Hz High Resolution.

3.1.3.3 Frame durations

Table 9 shows the values of the Frame Durations field. The **SRC shall** support the 10 ms and the 5 ms frame duration, and the **SNK shall** support at least one of the two; 7.5 ms and 2.5 ms are optional in both. At 2.5 ms a SRC **should not** aggregate frame data blocks, and a device **should** announce and select 2.5 ms only while its own controller serves no other link. At 2.5 ms a stereo SRC sends one three-slot baseband packet per frame period at the base value of section 3.3.2, which with its reply slot takes 1600 slots per second, 1470 at 44100 Hz.

Position	Frame Duration	Support in SRC	Support in SNK
Octet 6; b7	RFA	-	-
Octet 6; b6	10 ms codec frames	M	C.4

NOTE: 10, 5 and 2.5 ms keep the bits of Codec ID 0x0001; 7.5 ms takes the next free bit.

Position	Frame Duration	Support in SRC	Support in SNK
Octet 6; b5	5 ms codec frames	M	C.4
Octet 6; b4	2.5 ms codec frames	O	O
Octet 6; b3	7.5 ms codec frames	O	O
Octet 6; b2	RFA, reserved for 1.25 ms	-	-
Octet 6; b1-b0	RFA	-	-

Table 9: Frame Durations for LC3plus. Section 3.2.8 states the condition label C.4.

3.1.3.4 Audio channel counts

Table 10 shows the values of the Audio Channel Counts field. The **SNK** and the **SRC shall** support 2 channels (stereo); 1 channel (mono) is optional in both. At the Audio Channel Count 2 the channel of channel counter 1 **shall** be the Front Left channel and the channel of channel counter 2 the Front Right channel (Audio Locations 0x00000001 and 0x00000002 of Assigned Numbers (5), section 6.12.1). At the Audio Channel Count 1 the single channel carries no location.

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 10: Audio Channel Counts for LC3plus.

3.1.3.5 Sampling frequencies

Table 11 shows the values of the Sampling Frequencies field. A **SRC shall** support all four sampling frequencies. A **SNK shall** support 44100, 48000 and 96000 Hz; 192000 Hz is optional for it. A SRC that sends the Set Configuration or the Reconfigure command **should** select the lowest frequency the SNK announced that is at or above the sampling frequency of its source material, and the highest one the SNK announced where the material lies above all of them. A SNK that configures the stream does not know the source material and **should** select 96000 Hz and 24 bit. A SRC **shall** resample source material of another sampling frequency to the selected one before it encodes.

Position	Sampling Frequency	Support in SRC	Support in SNK
Octet 8; b7	44100 Hz	M	M
Octet 8; b6	48000 Hz	M	M
Octet 8; b5	96000 Hz	M	M
Octet 8; b4	192000 Hz	M	O
Octet 8; b3-b0	RFA	-	-

Table 11: Sampling Frequencies for LC3plus over A2DP.

3.1.3.6 Sample resolution

Table 12 shows the values of the Sample Resolution field. A **SRC** and a **SNK shall** support 16 bit and 24 bit. The field states the resolution of the coded audio, which is part of the bitstream definition in Lossless CBR mode ((1), clause 5.10.1), not the sample format at the audio interface of an implementation, which can also be 32 bit floating point.

Position	Sample Resolution	Support in SRC	Support in SNK
Octet 9; b7	16 bit	M	M
Octet 9; b6	24 bit	M	M
Octet 9; b5-b0	RFA	-	-

Table 12: Sample Resolution for LC3plus.

3.1.3.7 Version and features

Table 13 shows the values of octet 10. Bits 7 to 4 carry the major version of this document, 0x1 for version 1, and bits 3 to 0 one bit per feature added after version 1, all RFA in version 1. A device **shall** announce the highest major version it supports. A device reading a Get Capabilities or Get All Capabilities response **shall** accept any version other than 0x0, **shall** ignore feature bits it does not know and **shall not** select a configuration that needs a version above the one the peer announced. In the Set Configuration and Reconfigure commands octet 10 carries the version selected for the stream, at most the one the receiving device announced.

Position	Meaning	Support in SRC	Support in SNK
Octet 10; b7-b4	Major version, 0x1 for version 1	M	M
Octet 10; b3-b0	Features, all RFA in version 1	-	-

Table 13: Version and features.

3.1.3.8 Operating mode

Table 14 shows the values of the Operating Mode field. Bit n stands for the value n of the hrmode parameter of (2). A **SRC** and a **SNK shall** initialize the encoder and the decoder with the value the Set Configuration or Reconfigure command selects. This document carries no Normal mode over A2DP; b0 is reserved for it.

Position	Operating Mode	Support in SRC	Support in SNK
Octet 11; b7-b3	RFA	-	-
Octet 11; b2	Lossless CBR mode, hrmode 2	M	M
Octet 11; b1	RFA	-	-
Octet 11; b0	RFA, reserved for Normal mode, hrmode 0	-	-

Table 14: Operating Mode for LC3plus.

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 its fields as RFC 3550 (6) and TS 103 634 (1), clause B.2.2 do, with the requirements of this section.

3.1.4.2 Marker bit

A SRC **shall** set the marker bit to 0, and a SNK **shall** ignore it.

3.1.4.3 Sequence number

The RTP Sequence Number **shall** be incremented by 1 per transmitted packet, as IETF RFC 3550 (6) describes.

3.1.4.4 Timestamp

The RTP clock rate follows TS 103 634 (1), clause B.2.2.4, which covers neither 7.5 ms nor 192000 Hz; this document extends its rule to both: a SRC **shall** use 44100 Hz at the sampling frequency 44100 Hz and 96000 Hz at 48000 Hz, 96000 Hz and 192000 Hz, and a SNK **shall** derive the clock rate from the sampling frequency of octet 8 by the same rule. The Time Stamp Increment (TSI) between consecutive frame data blocks is the number of clock ticks of one frame period, see Table 15.

Sampling frequency	2.5 ms	5 ms	7.5 ms	10 ms
44100 Hz	120	240	360	480
48000 Hz	240	480	720	960
96000 Hz	240	480	720	960
192000 Hz	240	480	720	960

Table 15: Time Stamp Increments per sampling frequency and frame duration. Columns name the nominal frame duration; at 44100 Hz the frame period is 2.721, 5.442, 8.163 and 10.884 ms (section 2.1.1).

The Timestamp (TS) of a media payload is the sampling time of the first sample of its first frame data block, and while the stream stays started it **shall** advance by the TSI for every block the SRC encodes, transmitted or not. All media packets of one fragmented media payload **shall** carry the same TS, and a SNK **shall** derive the TS of the following blocks from the Table of Contents of section 3.1.5.5.

3.1.4.5 Payload type and SSRC

A SRC **shall** select the Payload Type from the dynamic range 96 to 127 of IETF RFC 3551 (17), section 3, one for each combination of sampling frequency, frame duration and Audio Channel Count it sends on a transport channel, as clauses B.2.2.4 and B.3.2 of (1) require, and **shall** set the padding bit, the extension bit and the contributing source (CSRC) count to zero. A SNK **shall** accept any Payload Type and SSRC. A Reconfigure command can change the RTP clock rate (section 3.1.4.4), so a SNK **shall not** relate a Timestamp sent before it to one sent after it.

3.1.5 Media payload format

3.1.5.1 General

LC3plus codes each channel separately, as TS 103 634 (1), clause 5.2.1 states. Figure 1 shows the channel ordering and the multiplexing of the frame data for a stereo example, where Channel 1 and Channel 2 are the channels of channel counter 1 and 2.

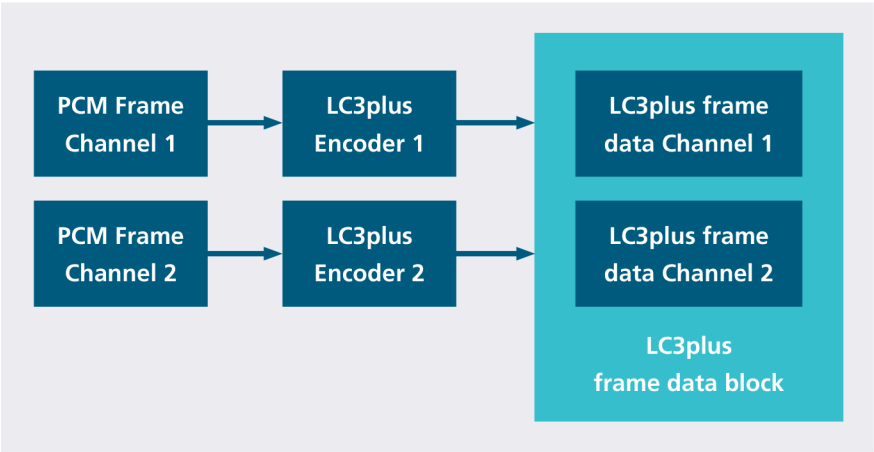


Figure 1: Multi-channel LC3plus frame data multiplexing order.

This document uses the RTP payload format of TS 103 634 (1), Annex B, with the restrictions and additions of sections 3.1.4 and 3.1.5. Sections 3.1.4 and 3.1.5 define the media packet header and the media payload of Codec ID 0x0003 completely; a later revision of Annex B of (1) does not change them. Each frame data carries its own length, so a SRC **may** send a frame on the lossless path shorter than the bit rate limit, filled as section 2.1.2 states.

NOTE: Codec ID 0x0001 has another media payload format: its one octet header carries the number of frame data blocks, there is no payload header with FDLR, no Table of Contents and all frame data of one media payload are of equal length. A parser for 0x0001 does not read the format of this document.

3.1.5.2 Payload structure

Octet 0 of the media payload of every media packet, the first octet after the 12 octet media packet header and after a content protection header where one is selected, **shall** be a fragmentation header. The octets following it in the first media packet of a media payload are the payload of TS 103 634 (1), clause B.2.4: a payload header (PH) of one octet, a Table of Contents of one Frame Type Description (FTD) per frame data, and the frame data in FTD order. Figure 2 shows the two formats.

Octet	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	Bit 0
0	F	S	L	RFA	Remaining fragments			
1	FDLR							
2	FC		FDI		RFA			H
3	FDL1							
4	(FDL2)							
5	(FDL3)							

Table 16: Header format of the media payload for LC3plus. Octet 0 is the fragmentation header of every media packet, octet 1 the payload header and octet 2 and following the first Frame Type Description of the first media packet.

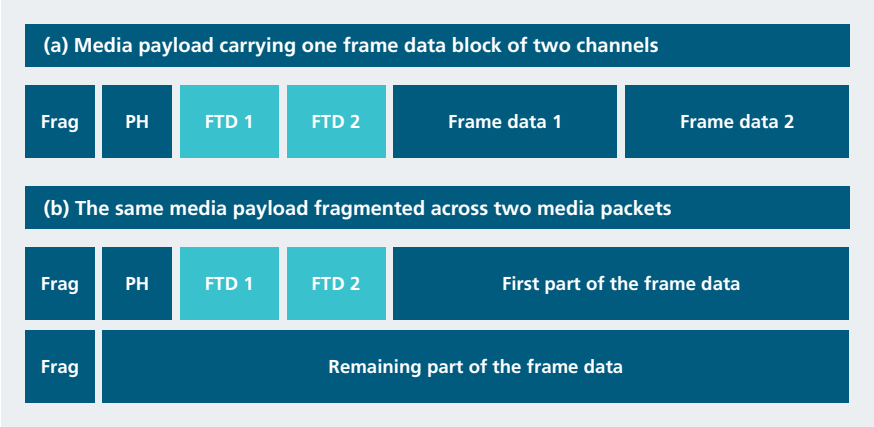


Figure 2: Media payload format of LC3plus. Frag is the fragmentation header, PH the payload header, FTD a Frame Type Description. Section 3.1.5.7 states where the cut of format (b) can fall.

3.1.5.3 Fragmentation header

The fragmentation header is octet 0 of the media payload of every media packet; its format is the first row of Table 16.

- **F bit:** a SRC **shall** set it to 1 if the media payload is fragmented, to 0 otherwise.
- **S bit:** a SRC **shall** set it to 1 for the first packet of a fragmented media payload, to 0 otherwise.
- **L bit:** a SRC **shall** set it to 1 for the last packet of such a payload, to 0 otherwise.
- **RFA.**
- **Remaining fragments** (4 bits, unsigned integer, most significant bit first): with F = 1, a SRC **shall** carry the number of remaining fragments including the current one, so the last value is 1; with F = 0 the field is RFA.

3.1.5.4 Payload header

The payload header is octet 1 of the first media packet of a media payload and carries the Frame Data Length Request (FDLR) of clause B.2.5 of (1). A SRC **shall** set it to NO_REQ, the value 0x00, and a SNK **shall** ignore it.

3.1.5.5 Table of Contents

The Table of Contents follows the payload header, one Frame Type Description per frame data in the media payload. An FTD **shall** be 2, 3 or 4 octets long, in the format of the rows for octet 2 and following of Table 16. Its fields are:

- **FC** (2 bits): Frame and Channel indicator, see Table 17. It states what the next FTD describes and where the Table of Contents ends. The channel counter counts the channels of one frame data block from 1.
- **FDI** (2 bits): Frame Duration Index, see Table 18. A SRC **shall** set it to the frame duration of the configuration. Value 3 is 7.5 ms, which Table B.3 of (1) marks reserved.
- **RFA** (3 bits): the BandWidth and Resolution index of Table B.4 of (1), which has no code point for Lossless CBR mode. A SNK **shall** take the sampling frequency, the sample resolution and the operating mode from octets 8, 9 and 11 of the Codec Specific Information.
- **H** (1 bit): redundancy indicator of clause B.2.6 of (1). A SRC **shall** set it to 0.

- **FDL** (8, 16 or 24 bits): Frame Data Length, the number of octets of the corresponding frame data, see Table 19.

Value	Binary	Meaning	TS next	CC next
0	00	This is the last FTD. The frame data follow.	-	-
1	01	The next FTD describes the next channel of the same frame data block.	+0	+1
2	10	The next FTD describes the first channel of the next frame data block.	+TSI	= 1
3	11	RFA	-	-

Table 17: FC encoding.

Value	Binary	Frame duration
0	00	2.5 ms
1	01	5 ms
2	10	10 ms
3	11	7.5 ms

Table 18: FDI encoding.

The FDL values 0, 1 and 2 carry a frame data length of zero. A SRC **shall not** send them. A SNK receiving one **shall** read no frame data for that FTD, **shall** treat the frame as lost and **shall** invoke packet loss concealment per clause 5.6 of (1).

FDL1	FDL2	FDL3	FTD size	Frame data length
0 to 2	not used	not used	2 octets	0 (not sent here)
3 to 19	not used	not used	2 octets	RFA
20 to 254	not used	not used	2 octets	FDL1, that is 20 to 254
255	0 to 254	not used	3 octets	255 + FDL2, that is 255 to 509
255	255	0 to 255	4 octets	510 + FDL3, that is 510 to 765

Table 19: FDL encoding. All values are decimal.

3.1.5.6 Forming the payload

The FTDs **shall** be ordered as clause B.2.7 of (1) requires: the channels of the oldest frame data block first in increasing channel order, then those of the next block, and so on to the newest, and the frame data **shall** follow the last FTD in that order. A SNK **shall** read the Table of Contents until an FTD whose FC is 0 and **shall** take the frame data from the octets that follow. After defragmentation, a SNK **shall** discard a media payload

- that carries more FTDs than the aggregation limit of section 3.1.5.8 for the selected frame duration times the Audio Channel Count, or no FTD whose FC is 0,
- whose number of FTDs per frame data block differs from the Audio Channel Count,
- whose frame data lengths and header sizes do not sum to the length of the payload,
- in which an FTD carries FC = 3 or an FDL from 3 to 19,
- in which an FTD carries an FDI that does not match the configured frame duration, or the H bit set to 1.

A SNK **shall** invoke packet loss concealment for every frame period missing between two media payloads it accepts while the stream stays started; the Timestamp of the second gives their number.

3.1.5.7 Fragmentation

If the configured MTU of the transport channel is at least the size of the media packet, the media payload **shall** be carried in one media packet with the F bit set to 0, as in format (a) of Figure 2. Otherwise, and provided that the multiplexing service of AVDTP (4), section 7.5 is not selected, the media payload **shall** be fragmented across several media packets, as in format (b): one octet stream, starting with the payload header, cut at any octet boundary into fragments that each carry their own fragmentation header. A SNK **shall** defragment the media payload before parsing it. All fragments except the last **shall** have the same size, and a media payload **shall not** be fragmented into more than 15 media packets. With the multiplexing service selected, a SRC **shall** select a bit rate whose media payload fits the MTU, unless the Adaptation Layer Fragmentation of AVDTP (4), section 7.5.3 is configured. A SNK **shall** discard an incomplete set of fragments. A set is incomplete when a media packet with the S bit set or an unfragmented media packet arrives while a set is open, when a fragment arrives without a preceding S bit, or when the Remaining fragments field does not decrement by one or the Timestamp changes. A SRC at the 10 ms frame duration **shall** be prepared to fragment; a SNK supporting it **shall** be prepared to defragment. A SRC **shall not** fragment at 7.5 ms, 5 ms or 2.5 ms and **shall** there select a bit rate whose media payload fits the configured MTU.

3.1.5.8 Restrictions

The FDL field carries a frame data length from 20 to 765 octets, as Table 19 states. The lower end is the smallest payload a decoder accepts per TS 103 634 (1), Table 5.2. A SRC **shall** select a bit rate limit whose octet count lies between the lowest value that section 3.3.2 states for the combination and the smaller of 765 octets and the maximum that section 3.3.2 states for it at the selected sample resolution.

NOTE: 765 octets per audio channel correspond to 612 kbit/s at 10 ms, 816 kbit/s at 7.5 ms, 1224 kbit/s at 5 ms and 2448 kbit/s at 2.5 ms (562.3 and 749.7 kbit/s at 44100 Hz at 10 and 7.5 ms).

A SRC **shall not** aggregate more than two frame data blocks into one media payload at the 10 ms and the 7.5 ms frame duration, more than four at 5 ms and more than eight at 2.5 ms. A SNK **shall** accept this aggregation at the Audio Channel Count selected for the stream and **shall** adapt to the number of frame data blocks and the length of every frame data, payload by payload, as the Table of Contents states them.

3.1.6 Restrictions of codec and transport configuration

A SRC **should** select a bit rate the transport sustains; Annex A.1 gives figures. Except at 2.5 ms, a SRC **should** aggregate as many frame data blocks into one media payload as section 3.1.5.8 allows while it still fits one media packet of the configured MTU, and one block where even one does not fit. The MTU in which a SRC sends media packets is the one the SNK announces in its L2CAP Configuration Request, which a SRC can only lower in its response ((9), Vol 3 Part A, section 5.1). A SNK **should** announce at least the user payload of a five-slot baseband packet of the connection less the four octets of the L2CAP header, giving a recommended MTU of 335, 675 and 1017 octets for DH5, 2-DH5 and 3-DH5, and a SRC **should not** lower it. Of the MTU, the media packet header takes 12 octets, the fragmentation header and the payload header one octet each, and every Frame Type Description 2 to 4 octets. This document neither requires nor excludes a content protection method. The figures of this section and of Annex A are computed without it.

3.1.6.1 Bit rate limit

The maximum bit rate of Lossless CBR mode is `(bits_per_audio_sample + 1)` × sampling frequency per channel ((1), clause 5.10.1), 816 kbit/s at 48000 Hz and 16 bit; Table 39 gives the octet counts. These can exceed the 765 octets of the FDL field and the throughput of A2DP; section 3.1.5.8 bounds the limit.

The Japan Audio Society (JAS) awards its Hi-Res Audio Wireless logo to products whose codec meets the conditions of (22), and LC3plus is certified for it. The certification covers the High Resolution mode of (1), clause 5.8 at 96000 Hz, 24 bit and at least 450 kbit/s per audio channel at the 10 ms and the 7.5 ms frame duration, 470 kbit/s at 5 ms and 485 kbit/s at 2.5 ms; Fraunhofer IIS provides its compliance document and self-certificate for the Hi-Res Audio Wireless logo on request. A SRC that claims the Hi-Res Audio Wireless logo for a stream **shall** select 96000 Hz and 24 bit, in place of the selection of section 3.1.3.5, and **shall** operate with a bit rate limit of at least 563 octets (450.4 kbit/s) at 10 ms, 422 octets (450.1 kbit/s) at 7.5 ms, 294 octets (470.4 kbit/s) at 5 ms and 152 octets (486.4 kbit/s) at 2.5 ms per audio channel where the transport sustains it, an adapting one bounding L0 by the value for its frame duration (section 3.3.3).

A SRC **shall** hand its encoder the samples at the Sample Resolution selected for the stream. Where its material carries fewer bits than that resolution, the SRC **shall** place them in the most significant bits of each sample; where it carries more, the SRC **shall** reduce it to that resolution before the encoder. A SRC that sends the Set Configuration or the Reconfigure command **should** select 24 bit where its material carries more than 16 bits.

3.1.7 Delay Reporting

A SNK and a SRC exposing this codec **shall** support the Delay Reporting service capability of AVDTP (4), section 8.21.9, and the device that initiates the stream configuration **shall** configure it where the other device announces it. A SNK **shall** report, as AVDTP (4), sections 5.5 and 8.19.1 describe, a delay that covers the processing of its decoder, the operation range of its receive buffer and the time of its output path, and **shall** report a new value after a Reconfigure command that changes the frame duration or the sampling frequency. A SRC **shall** accept the report even if it does not use the value.

3.2 BLUETOOTH LE AUDIO

3.2.1 Overview

This section defines signaling and transport of LC3plus over the Basic Audio Profile (7). The coding format is signaled as vendor specific coding format via Fraunhofer's Company ID. The requirements apply alike to Unicast Client, Unicast Server, Broadcast Source and Broadcast Sink unless stated otherwise.

3.2.2 Codec_ID field

Table 20 defines the Codec_ID field for LC3plus. Octets 1 and 2 carry the Company ID, octets 3 and 4 the Vendor Specific Codec ID, each least significant octet first.

Octet	Value	Meaning
0	0xFF	Vendor specific Coding_Format
1	0xA9	Company ID of Fraunhofer IIS, 0x08A9
2	0x08	Company ID of Fraunhofer IIS, 0x08A9
3	0x03	LC3plus Vendor Specific Codec ID, 0x0003
4	0x00	LC3plus Vendor Specific Codec ID, 0x0003

NOTE: This document uses Vendor Specific Codec ID 0x0003. Codec ID 0x0001 identifies LC3plus High Resolution as specified in (21).

Table 20: Codec_ID field for LC3plus. The five octets form the Codec_ID parameter of (11) and of (10) and the Codec_ID of a BASE subgroup, (7), Table 3.15. The Coding_Format value 0xFF is the Vendor Specific entry of (5), section 2.11.

3.2.3 Codec_Specific_Capabilities LTV requirements

A Unicast Server or Broadcast Sink exposing LC3plus **shall** populate the Codec_Specific_Capabilities field of a PAC record as this section states, in the format of the Published Audio Capabilities Service (11), Tables 3.2 and 3.4. A Broadcast Source has no PAC record. This document's vendor specific LTV structures use the types 0xF1 to 0xF4, 0xF8, 0xF9 and 0xFB. The types 0xF5, 0xF6, 0xF7 and 0xFA stay free, and a device **shall not** send them under the Codec_ID of Table 20.

The following LTV structures **shall** be present:

- **LC3plus_Supported_Operating_Modes**, section 3.2.3.6.
- **Supported_Sampling_Frequencies**, type 0x01 of (8), see section 3.2.3.2.
- **LC3plus_Supported_Frame_Durations**, section 3.2.3.1.
- **LC3plus_Version**, section 3.2.3.5.
- **LC3plus_Supported_Octets_Per_Codec_Frame**, section 3.2.3.4, one structure per frame duration announced in LC3plus_Supported_Frame_Durations and none for another.
- **LC3plus_Supported_Sample_Resolutions**, section 3.2.3.3, if the record announces Lossless CBR mode.

The following LTV structures **may** be present:

- **Supported_Audio_Channel_Counts**, type 0x03 of (8). Absence **shall** be read as one audio channel.
- **Supported_Max_Codec_Frames_Per_SDU**, type 0x05 of (8). Absence **shall** be read as one codec frame per audio channel per SDU.

A client **shall** ignore a PAC record that lacks one of the mandatory structures. A PAC record **shall not** carry the types 0x02 and 0x04 of (8), and a receiver **shall** ignore them where they appear.

3.2.3.1 LC3plus_Supported_Frame_Durations

Parameter	Size (octets)	Value
Length	1	0x03
Type	1	0xF1
Value	2	Bitfield, see Table 22

Table 21: LC3plus_Supported_Frame_Durations LTV structure.

Bit	Meaning
0	10 ms frame duration
1	7.5 ms frame duration
2	5 ms frame duration
4	RFU, reserved for 1.25 ms
3, 5 to 15	RFU

Table 22: Bitfield of LC3plus_Supported_Frame_Durations. In each bit 0b1 means supported and 0b0 means not supported.

At least one bit of the value field of Table 21 **shall** be set. Table 22 gives the meaning of each bit. Over LE Audio the shortest ISO interval is 5 ms ((9), Vol 6 Part B, section 4.5.13.2, and Vol 4 Part E, section 7.7.65.27), so this document carries 2.5 ms over A2DP only.

3.2.3.2 Supported_Sampling_Frequencies

The Supported_Sampling_Frequencies LTV structure, type 0x01 of (8), carries a 16 bit value of which at least one bit **shall** be set. Bits that Table 23 does not list for the operating mode of the record **shall** be zero and are read as zero. A client **shall** ignore a PAC record that sets no bit Table 23 lists for its operating mode.

Bit	Sampling frequency	Operating mode
2	16000 Hz	Normal, hrmode 0
4	24000 Hz	Normal, hrmode 0
5	32000 Hz	Normal, hrmode 0
6	44100 Hz	Lossless CBR, hrmode 2
7	48000 Hz	Lossless CBR, hrmode 2
9	96000 Hz	Lossless CBR, hrmode 2
11	192000 Hz	Lossless CBR, hrmode 2
0, 1, 3, 8, 10, 12	8000, 11025, 22050, 88200, 176400 and 384000 Hz	not used
13 to 15	RFU	RFU

Table 23: Sampling frequencies of each operating mode.

3.2.3.3 LC3plus_Supported_Sample_Resolutions

Parameter	Size (octets)	Value
Length	1	0x02
Type	1	0xF8
Value	1	Bit 0: 16 bit. Bit 1: 24 bit. Bit 2 to 7: RFU.

Table 24: LC3plus_Supported_Sample_Resolutions LTV structure.

A device that supports Lossless CBR mode in a role **shall** support 16 bit and 24 bit in that role and **shall** set bit 0 and bit 1 of Table 24 in every PAC record of that mode. The Normal mode bitstream carries no sample resolution (16 bit, section 1.2), so a PAC record announcing Normal mode **shall not** carry the structure, and a receiver **shall** ignore it there.

3.2.3.4 LC3plus_Supported_Octets_Per_Codec_Frame

Table 25 gives the type values, Table 26 the format.

Structure	Type	Required when
LC3plus_Supported_Octets_Per_Codec_Frame_10ms	0xF2	10 ms is announced
LC3plus_Supported_Octets_Per_Codec_Frame_7_5ms	0xF3	7.5 ms is announced
LC3plus_Supported_Octets_Per_Codec_Frame_5ms	0xF4	5 ms is announced

Table 25: Type values of the LC3plus_Supported_Octets_Per_Codec_Frame LTV structures and the frame duration each one belongs to.

Parameter	Size (octets)	Value
Length	1	0x05
Type	1	see Table 25
Value	4	Octet 0 to 1: minimum number of octets per codec frame. Octet 2 to 3: maximum number of octets per codec frame.

Table 26: Format of an LC3plus_Supported_Octets_Per_Codec_Frame LTV structure. The two values follow the layout that (8), section 6.12.4.4 uses for the Supported_Octets_Per_Codec_Frame structure.

The range applies to every sampling frequency and sample resolution announced in the same PAC record for that frame duration, and a device **shall** support it for each of them. A device that supports a wider range for one sampling frequency announces it in a PAC record of its own. The minimum **shall not** be above the maximum, **shall not** be below the largest lowest value that section 3.3.2 gives for the record's combinations, and the maximum **shall not** be above the smallest of their maxima, for Lossless CBR mode the 16 bit one, to which a device **may** add the two octets that section 3.3.2 permits in that mode. A PAC record announcing Normal mode **shall** set the 5 ms bit of LC3plus_Supported_Frame_Durations alone.

3.2.3.5 LC3plus_Version

Parameter	Size (octets)	Value
Length	1	0x02
Type	1	0xF9
Value	1	Bits 7 to 4: major version, 0x1 for version 1, 0x0 RFU. Bits 3 to 0: features, all RFU in version 1.

Table 27: LC3plus_Version LTV structure.

Table 27 shows the structure. Bits 7 to 4 carry the major version of this document, 0x1 for version 1, and bits 3 to 0 one bit per feature added after version 1, all RFU in version 1. A device **shall** announce the highest major version it supports. A receiver **shall** accept any version other than 0x0, which it treats as an absent field, **shall** ignore feature bits it does not know and **shall not** select a configuration that needs a version above the one the peer announced.

3.2.3.6 LC3plus_Supported_Operating_Modes

Parameter	Size (octets)	Value
Length	1	0x02
Type	1	0xFB
Value	1	Bit 0: Normal mode, hrmode 0. Bit 2: Lossless CBR mode, hrmode 2. Bit 1 and bits 3 to 7: RFU.

Table 28: LC3plus_Supported_Operating_Modes LTV structure.

Table 28 shows the structure. Bit n of the value field stands for the value n of the hrmode parameter of (2). Exactly one bit **shall** be set. A client **shall** ignore a PAC record in whose LC3plus_Supported_Operating_Modes neither or both of bit 0 and bit 2 are set.

3.2.4 Codec_Specific_Configuration LTV requirements

A device **shall** follow this section when it initiates the Config Codec operation of (7), section 5.6.1, when it configures a broadcast Audio Stream as (7), section 6.3 defines, and, as a Unicast Server, when it writes the field itself in a Config Codec operation it initiates autonomously or in the Released operation with caching ((10), sections 5.1 and 5.9). The following LTV structures **shall** be present:

- **LC3plus_Operating_Mode**, section 3.2.4.3.
- **Sampling_Frequency**, type 0x01 of (8).
- **Octets_Per_Codec_Frame**, type 0x04 of (8).
- **LC3plus_Frame_Duration**, section 3.2.4.1.
- **LC3plus_Sample_Resolution**, section 3.2.4.2, in Lossless CBR mode.

The following LTV structures **may** be present:

- **Audio_Channel_Allocation**, type 0x03 of (8). Absence **shall** be read as one audio channel without a specified location.
- **Codec_Frame_Blocks_Per_SDU**, type 0x05 of (8). Absence **shall** be read as 0x01. A Broadcast Source **shall not** include it.

In Normal mode (16 bit, section 1.2) a device **shall not** include LC3plus_Sample_Resolution, and a receiver **shall** ignore it there.

To render audio channels at specific audio locations, Audio_Channel_Allocation **shall** be present with at least one Audio Location bit set; for a single mono audio channel with no specified audio location it **shall** be left out. In a single Audio Stream a device **shall not** configure several mono channels, several channels with the same audio location, or a mix of a mono channel and located channels. A receiver **shall** take the frame duration from LC3plus_Frame_Duration and **shall not** reject a configuration because it also carries the type 0x02 of (8). A device that writes the configuration after reading the peer's PAC records **shall** select a combination of operating mode, sampling frequency, frame duration and sample resolution the peer announced in its Codec_Specific_Capabilities, a number of audio channels it announced in Supported_Audio_Channel_Counts, no more codec frames in an SDU than its Supported_Max_Codec_Frames_Per_SDU, and an Octets_Per_Codec_Frame inside the range it announced for the frame duration. A Unicast Server that writes the configuration of its own ASE **shall** select one that its own PAC records for that direction announce, and a Broadcast Source **shall** select one of section 3.2.10.2. A client that does not encode the configuration the server wrote for a Sink ASE, or decode the one for a Source ASE, **shall** write its own with a Config Codec operation before the Config QoS operation. An encoder **may** code a channel at a Low Frequency Effects location in the LFE mode of (1), clause 5.3.15, which the encoder interface of (2) enables per channel; it applies in Normal mode and needs no signaling. An encoder **shall not** code any other channel in that mode.

3.2.4.1 LC3plus_Frame_Duration

Parameter	Size (octets)	Value
Length	1	0x02
Type	1	0xF1
Value	1	0x01 = 10 ms, 0x02 = 7.5 ms, 0x03 = 5 ms. 0x05 = RFU, reserved for 1.25 ms. All other values RFU.

Table 29: LC3plus_Frame_Duration LTV structure.

The values of Table 29 are the nominal frame durations; at 44100 Hz the frame period of section 2.1.1 applies.

3.2.4.2 LC3plus_Sample_Resolution

Table 30 gives the sample resolution in bits, with which a sink **shall** initialize its decoder.

Parameter	Size (octets)	Value
Length	1	0x02
Type	1	0xF8
Value	1	0x10 = 16 bit, 0x18 = 24 bit. All other values RFU.

Table 30: LC3plus_Sample_Resolution LTV structure.

3.2.4.3 LC3plus_Operating_Mode

Parameter	Size (octets)	Value
Length	1	0x02
Type	1	0xFB
Value	1	0x00 = Normal mode, 0x02 = Lossless CBR mode. All other values RFU.

Table 31: LC3plus_Operating_Mode LTV structure.

Table 31 gives the operating mode. The value is the hrmode parameter of (2) with which a source **shall** initialize its encoder and a sink its decoder. A sampling frequency that Table 23 does not list for that value counts as an RFU code point.

3.2.5 Handling of unknown and malformed LTV structures

All multi-octet values in the Codec_Specific_Capabilities and Codec_Specific_Configuration fields under the Codec_ID of Table 20 **shall** be transmitted least significant octet first. A receiver **shall** skip an LTV structure whose Type it does not know, **shall** continue to parse the remainder and **shall not** reject a PAC record or a configuration for that reason alone. A receiver **shall** take the meaning of an LTV type from the Codec_ID and from the field it was read from: the types 0xF1, 0xF8 and 0xFB carry different meanings in the two fields, and Codec ID 0x0001 defines 0xF1 with another bitfield. A Type **shall** appear at most once per field. A field is malformed where a structure carries a Length of 0x00, where a Length reaches past the end of the field, where a Length differs from the one that this document or (8) gives its Type, or where a Type repeats. A receiver **shall** then discard the whole field. A Unicast Server **shall** reject a Codec_Specific_Configuration it cannot accept with the Response_Code of ASCS (10), Table 5.1 that fits the case:

- **0x09 Invalid Configuration Parameter Value**, if a mandatory LTV structure of section 3.2.4 is absent, if the field is malformed, if a value is an RFU code point, or if an Audio_Channel_Allocation carries no Audio Location bit.
- **0x07 Unsupported Configuration Parameter Value**, if the value is valid but the server cannot support it.
- **0x08 Rejected Configuration Parameter Value**, if the server supports the value but rejects it.
- **0x06 Unsupported Audio Capabilities**, if the server has not exposed the requested configuration in any PAC record.

For 0x07, 0x08 and 0x09 the Reason field **shall** be 0x02, Codec_Specific_Configuration. Where more than one code applies, the server **shall** use the first of 0x09, 0x06, 0x07 and 0x08 that applies. A Broadcast Sink **shall not** synchronize to the BISes of a subgroup whose Codec_Specific_Configuration it cannot accept, and **shall not** render a BIS whose Audio_Channel_Allocation carries no Audio Location bit.

3.2.6 Metadata LTV requirements

The Metadata LTV requirements of (7), section 4.3.3 **shall** apply to LC3plus under the Codec_ID of Table 20. Section 3.2.10.4 adds requirements for a Public Broadcast Source.

3.2.7 Media payload format

This document uses the LC3 Media Packet format of (7), section 4.2: the SDU carries the codec frames of all channels of one or more frame blocks, ordered from the lowest to the highest Audio Location bit index set in Audio_Channel_Allocation, without an additional header. Every codec frame in an SDU **shall** have the same length, from which section 3.3.7 derives the frame length. A source **shall not** send a codec frame longer than Octets_Per_Codec_Frame, and in Normal mode **shall** send codec frames of that length except where section 3.3.6.2 permits a shorter SDU. In Lossless CBR mode the encoder **shall** be operated with a bit rate limit at Octets_Per_Codec_Frame, at most at the maximum of section 3.3.2 for the selected sample resolution, or at the octet count of the level in the case of section 3.3.6.2. Under that limit a frame on the lossless

path can come out shorter, as section 2.1.2 describes. The LC3 Media Packet format carries no length per codec frame, so where an SDU carries several codec frames the source fills the shorter ones to the length of the longest. On a unicast Audio Stream a source **shall** fill a codec frame at least to the lower end of the range that section 3.3.7 binds the decoder to. On a broadcast Audio Stream every codec frame **shall** reach Octets_Per_Codec_Frame. A source **shall** pad a codec frame only with the octets of value 0x07 of section 2.1.2. In Lossless CBR mode a source **shall** hand its encoder the samples at the sample resolution selected for the stream. Where its material carries fewer bits, the source **shall** place them in the most significant bits of each sample; where it carries more, it **shall** reduce it to that resolution before the encoder. A device that writes the configuration for audio it sends **should** select 24 bit where its material carries more than 16 bits.

3.2.8 Codec configuration settings

Every requirement on a configuration depends on what the device supports in its role. The condition labels of the requirement tables of this document carry the following meaning; they differ from the labels of the same name in (7):

- **C.1:** Mandatory if the device supports the Lossless CBR operating mode in that role, otherwise Excluded.
- **C.2:** Mandatory if the device supports the Normal operating mode in that role, otherwise Excluded.
- **C.3:** Mandatory if the device supports the Lossless CBR operating mode, the 7.5 ms frame duration and the sampling frequency of the configuration in that role, otherwise Excluded.
- **C.4:** Mandatory to support at least one of the 10 ms and the 5 ms frame duration.
- **C.5:** Mandatory if the device supports the Lossless CBR operating mode and the sampling frequency 96000 Hz in that role, otherwise Excluded.

Table 32, Table 33 and Table 34 carry the requirements for a unicast Audio Stream, the tables of sections 3.2.10.2 and 3.2.10.3 those for a broadcast Audio Stream. C.1, C.2, C.3 and C.5 depend only on what a device does over LE Audio under the Codec_ID of Table 20, each of the roles Unicast Server, Unicast Client, Broadcast Source and Broadcast Sink on its own: in the Sink column on what it decodes, in the Source column on what it encodes. Where a Unicast Server with a Sink ASE is also a Broadcast Sink, the Sink columns of both roles depend on what its Sink PAC records announce under that Codec_ID. A device **shall** support, for its role, every configuration that a requirement table marks C.1, C.2, C.3 or C.5 and whose condition it meets. Every configuration that a Sink column marks C.1 or C.2 carries the same label in the Source column, so a Source and a Sink of the same operating mode always share a configuration. A Unicast Server or Broadcast Sink **shall** announce, for each configuration it supports in that role, a range under section 3.2.3.4 that includes its octets per codec frame.

Configuration names follow the pattern `<frequency>_<index>_LC3plus_<mode>_<QoS>`: `<mode>` is Normal or Lossless, `<QoS>` HighReliability or LowLatency, and a broadcast setting carries Broadcast. The indices 1 to 5 at 48 and 96 kHz follow Codec ID 0x0001 (Annex C.1). A codec configuration setting leaves out the QoS part.

3.2.8.1 Normal mode, 5 ms (low latency)

The Normal mode settings run at the 5 ms frame duration only, which the LC3 settings of (7) do not offer.

Configuration	Frequency (kHz)	Frame duration (ms)	Octets per frame (kbit/s)	Sink	Source
16_3_LC3plus_Normal	16	5	24 (38.4)	O	C.2
24_3_LC3plus_Normal	24	5	36 (57.6)	O	C.2
32_3_LC3plus_Normal	32	5	48 (76.8)	C.2	C.2

Table 32: Codec configuration settings, Normal mode. Section 3.2.8 states the condition labels.

3.2.8.2 Lossless CBR mode

Table 33 shares indices with Codec ID 0x0001; Annex C.1 states what that means.

Configuration	Frequency (kHz)	Resolution (bit)	Frame duration (ms)	Octets per frame (kbit/s)	Sink	Source
48_1_LC3plus_Lossless	48	16/24	10	160 (128)	C.1	C.1
48_2_LC3plus_Lossless	48	16/24	10	310 (248)	O	O
48_3_LC3plus_Lossless	48	16/24	7.5	117 (124.8)	C.3	O
48_4_LC3plus_Lossless	48	16/24	7.5	180 (192)	O	O
48_5_LC3plus_Lossless	48	16/24	5	120 (192)	O	O
96_1_LC3plus_Lossless	96	16/24	10	190 (152)	C.5	C.1
96_2_LC3plus_Lossless	96	16/24	10	310 (248)	O	O
96_3_LC3plus_Lossless	96	16/24	7.5	141 (150.4)	C.3	O
96_4_LC3plus_Lossless	96	16/24	7.5	225 (240)	O	O
96_5_LC3plus_Lossless	96	16/24	5	120 (192)	O	O
441_1_LC3plus_Lossless	44.1	16/24	10	500 (367.5)	O	O
441_2_LC3plus_Lossless	44.1	16/24	10	240 (176.4)	C.1	C.1
441_3_LC3plus_Lossless	44.1	16/24	7.5	375 (367.5)	O	O

Configuration	Frequency (kHz)	Resolution (bit)	Frame duration (ms)	Octets per frame (kbit/s)	Sink	Source
48_6_LC3plus_Lossless	48	16/24	10	450 (360)	O	O
192_1_LC3plus_Lossless	192	16/24	10	249 (199.2)	O	O

*Table 33: Codec configuration settings, Lossless CBR mode. A device that supports a row **shall** support it in 16 bit and in 24 bit. Section 3.2.8 states the condition labels.*

3.2.9 QoS configuration settings

Configuration	SDU interval	Framing	Max SDU (kbit/s)	PHY	RTN	Max latency	Presentation delay	Sink	Source
16_3_LC3plus_Normal_HighReliability	5000	unframed	24 (38.4)	2M	13	100	40000	O	C.2
24_3_LC3plus_Normal_HighReliability	5000	unframed	36 (57.6)	2M	13	100	40000	O	C.2
32_3_LC3plus_Normal_HighReliability	5000	unframed	48 (76.8)	2M	13	100	40000	C.2	C.2
16_3_LC3plus_Normal_LowLatency	5000	unframed	24 (38.4)	2M	4	10	5000	O	O
24_3_LC3plus_Normal_LowLatency	5000	unframed	36 (57.6)	2M	4	10	5000	O	O
32_3_LC3plus_Normal_LowLatency	5000	unframed	48 (76.8)	2M	4	10	5000	C.2	C.2
48_1_LC3plus_Lossless_HighReliability	10000	unframed	160 (128)	2M	13	100	40000	C.1	C.1
48_2_LC3plus_Lossless_HighReliability	10000	unframed	310 (248)	2M	13	100	40000	O	O
48_3_LC3plus_Lossless_HighReliability	7500	unframed	117 (124.8)	2M	13	75	40000	C.3	O
48_4_LC3plus_Lossless_HighReliability	7500	unframed	180 (192)	2M	13	75	40000	O	O
96_1_LC3plus_Lossless_HighReliability	10000	unframed	190 (152)	2M	13	100	40000	C.5	C.1
96_2_LC3plus_Lossless_HighReliability	10000	unframed	310 (248)	2M	13	100	40000	O	O
96_3_LC3plus_Lossless_HighReliability	7500	unframed	141 (150.4)	2M	13	75	40000	C.3	O
96_4_LC3plus_Lossless_HighReliability	7500	unframed	225 (240)	2M	13	75	40000	O	O
441_1_LC3plus_Lossless_HighReliability	10884	framed	500 (367.5)	2M	13	100	40000	O	O

Configuration	SDU interval	Framing	Max SDU (kbit/s)	PHY	RTN	Max latency	Presentation delay	Sink	Source
441_2_LC3plus_Lossless_HighReliability	10884	framed	240 (176.4)	2M	13	100	40000	C.1	C.1
441_3_LC3plus_Lossless_HighReliability	8163	framed	375 (367.5)	2M	13	75	40000	O	O
48_6_LC3plus_Lossless_HighReliability	10000	unframed	450 (360)	2M	13	100	40000	O	O
192_1_LC3plus_Lossless_HighReliability	10000	unframed	249 (199.2)	2M	13	100	40000	O	O
48_5_LC3plus_Lossless_LowLatency	5000	unframed	120 (192)	2M	4	10	5000	O	O
96_5_LC3plus_Lossless_LowLatency	5000	unframed	120 (192)	2M	4	10	5000	O	O

Table 34: QoS configuration settings. SDU interval and presentation delay in microseconds, maximum transport latency in milliseconds, maximum SDU size in octets per audio channel with its bit rate in kbit/s in parentheses. At 44100 Hz the SDU interval is the frame period of section 2.1.1, rounded, and the bit rate counts over the frame period of section 2.1.1, unrounded. The presentation delay of the LowLatency settings applies to a Sink ASE; on a Source ASE they carry 10000 microseconds. Section 3.2.8 states the condition labels.

The QoS part HighReliability denotes the standard QoS setting, LowLatency the low latency setting. Max SDU is the value for one audio channel with one block of codec frames per SDU; other channel allocations and more blocks scale Max SDU in the manner of footnote 2 of (7), Table 5.2, and more blocks multiply the SDU interval as well. A Unicast Client that needs a shorter transport latency **may** select a lower retransmission number and a shorter maximum transport latency than Table 34 states. The settings are stated for the LE 2M PHY, and a device that supports a setting of Table 34 or of section 3.2.10.3 **shall** support the LE 2M PHY.

A Unicast Server **shall** include in the Presentation_Delay range it exposes in the Codec Configured state the presentation delay that Table 34 gives for that type of ASE to every setting it supports for the codec configuration of that ASE. For a LowLatency setting on a Sink ASE a Unicast Server **shall** support a presentation delay of 10000 microseconds and **should** support 5000 microseconds, in place of the value of Table 34. The Unicast Client writes the value in the Config QoS operation inside the range the server exposes ((7), section 5.6.2). It **should** request the lowest value this section gives its QoS setting inside that range and **shall** request, under a profile that sets its own presentation delay, the value of that profile for the settings it makes mandatory. On a Source ASE the delay of the LowLatency settings counts from the start of acquisition ((7), section 7.1.2), and the 5 ms frame duration alone takes 7.5 ms of algorithmic delay.

At 44100 Hz no ISO interval is a multiple of the SDU interval, so those settings are framed ((9), Vol 6 Part G, section 2); the controller selects the ISO interval, and Annex B.1 assumes 10 ms and 7.5 ms. In framed mode a PDU of at most 251 octets carries

NOTE: SDU_Interval and Max_Transport_Latency are set per CIG and direction, Framing per CIG ((9), Vol 4 Part E, section 7.8.97), the presentation delay per direction ((7), section 7.1.3), and all of them per BIG. A 5 ms stream shares a direction of a CIG with a 10 ms stream only with two blocks of codec frames per SDU, and a 44100 Hz setting makes every stream of its CIG framed.

a segmentation header of 2 octets per segment and a time offset of 3 octets before the segment that starts an SDU, so at most 246 octets of codec data where an SDU starts and 249 otherwise ((9), Vol 6 Part B, section 4.5.13.3, and Part G, section 2.2). RTN is a recommendation to the controller, which selects NSE, BN and FT as Annex B.1 shows, and the host **shall** be capable of requesting the values of Table 34.

3.2.10 Broadcast operation

3.2.10.1 Signaling in a Basic Audio Announcement

A Broadcast Source announces its stream configuration in the Broadcast Audio Source Endpoint (BASE) structure of (7), section 3.7.2.2. Level 1 is the BIG, level 2 a subgroup, level 3 a single BIS. Table 35 gives each parameter's LC3plus value.

Level	Parameter	Size	Value for LC3plus
1	Presentation_Delay	3	see section 3.2.10.5
1	Num_Subgroups	1	at least 1
2	Num_BIS[i]	1	at least 1
2	Codec_ID[i]	5	the value of Table 20, unchanged
2	Codec_Specific_Configuration_ Length[i]	1	16, or 13 in Normal mode, for the LTV structures below
2	Codec_Specific_Configuration [i]	varies	the LTV structures of section 3.2.4
2	Metadata_Length[i]	1	see section 3.2.6
2	Metadata[i]	varies	see sections 3.2.6 and 3.2.10.4
3	BIS_index[i[k]]	1	1 to 31
3	Codec_Specific_Configuration_ Length[i[k]]	1	6, or 0 for a single mono channel without audio location
3	Codec_Specific_Configuration [i[k]]	varies	Audio_Channel_Allocation only

Table 35: Parameters of the BASE structure of (7), Table 3.15, and the value LC3plus places there. Size in octets. A BIG carries at most 31 BISes ((9), Vol 6 Part B, section 4.4.6.2).

A Broadcast Source **shall** populate Codec_ID at level 2 with the value of Table 20. The Codec_Specific_Configuration **shall** be present at level 2, with a length other

than 0x00, and **may** be present at level 3. Each LTV structure **shall** appear at one level only: level 2 carries LC3plus_Operating_Mode, Sampling_Frequency, Octets_Per_Codec_Frame, LC3plus_Frame_Duration and, in Lossless CBR mode, LC3plus_Sample_Resolution; level 3 carries Audio_Channel_Allocation. The BASE travels in one AD structure, so it **shall not** exceed 252 octets ((9), Vol 3 Part C, section 11); with 9 octets of Metadata a subgroup of two BISes takes 49 octets of it in Lossless CBR mode and 46 in Normal mode. A Broadcast Sink that reads an incomplete BASE **shall** discard it and wait for a complete one.

3.2.10.2 Broadcast codec configuration settings

Table 36 gives the codec configuration settings of a broadcast Audio Stream, for one audio channel per BIS and one codec frame block per SDU. It is a subset of Table 32 and Table 33. 44100 Hz would need framed operation, and more than 251 octets (200.8 kbit/s at 10 ms) per codec frame a burst number above 1, so neither appears. A Broadcast Source that is to reach every Broadcast Sink of its operating mode **should** use 32_3 or 48_1, which every such sink supports.

Configuration	Frequency (kHz)	Frame duration (ms)	Octets per frame (kbit/s)	Sink	Source
16_3_LC3plus_Normal	16	5	24 (38.4)	O	C.2
24_3_LC3plus_Normal	24	5	36 (57.6)	O	C.2
32_3_LC3plus_Normal	32	5	48 (76.8)	C.2	C.2
48_1_LC3plus_Lossless	48	10	160 (128)	C.1	C.1
96_1_LC3plus_Lossless	96	10	190 (152)	C.5	C.1
48_3_LC3plus_Lossless	48	7.5	117 (124.8)	O	O
48_5_LC3plus_Lossless	48	5	120 (192)	O	O
192_1_LC3plus_Lossless	192	10	249 (199.2)	O	O

*Table 36: Broadcast codec configuration settings. A device that supports a Lossless CBR row **shall** support it in 16 and in 24 bit.*

3.2.10.3 Broadcast QoS configuration settings

Configuration	SDU interval	Max SDU (kbit/s)	RTN	Max latency	Presentation delay	Sink	Source
16_3_LC3plus_Normal_Broadcast	5000	24 (38.4)	2	60	40000	O	C.2
24_3_LC3plus_Normal_Broadcast	5000	36 (57.6)	2	60	40000	O	C.2

Configuration	SDU interval	Max SDU (kbit/s)	RTN	Max latency	Presentation delay	Sink	Source
32_3_LC3plus_Normal_Broadcast	5000	48 (76.8)	2	60	40000	C.2	C.2
48_1_LC3plus_Lossless_Broadcast	10000	160 (128)	4	65	40000	C.1	C.1
96_1_LC3plus_Lossless_Broadcast	10000	190 (152)	3	65	40000	C.5	C.1
48_3_LC3plus_Lossless_Broadcast	7500	117 (124.8)	3	50	40000	O	O
48_5_LC3plus_Lossless_Broadcast_LowLatency	5000	120 (192)	2	20	10000	O	O
192_1_LC3plus_Lossless_Broadcast	10000	249 (199.2)	2	65	40000	O	O

Table 37: Broadcast QoS configuration settings, all unframed and on the LE 2M PHY. SDU interval and presentation delay in microseconds, maximum transport latency in milliseconds, maximum SDU size in octets per audio channel with its bit rate in kbit/s in parentheses. The Sink column binds the Broadcast Sink, the Source column the Broadcast Source.

The QoS part Broadcast denotes a broadcast QoS setting, Broadcast_LowLatency the broadcast low latency setting. Max SDU is the value for one audio channel per BIS; a BIS that carries more audio channels scales it in the manner of footnote 2 of (7), Table 6.4. HCI_LE_Create_BIG of (9), Vol 4 Part E, section 7.8.103 takes these values for the whole BIG, and the host **shall** be capable of requesting the values of Table 37. Annex B.2 gives the airtime of a BIG.

3.2.10.4 Public Broadcast Profile

Bits 1 and 2 of the features field of the Public Broadcast Announcement ((12), Table 4.1) announce LC3 settings of the BIG that its advertising set points to, so a Public Broadcast Source **shall not** set them where that BIG carries only LC3plus; it **shall** set bit 1 where that BIG also carries an LC3 subgroup that meets (12), section 4.2, and bit 2 where that BIG also carries one that meets (12), section 4.3. A Broadcast Source carrying an LC3plus and an LC3 subgroup in one BIG **shall** use values of SDU_Interval, Max_Transport_Latency, RTN and Presentation_Delay that both settings permit and a Max_SDU that covers both; where no such values exist, it **shall not** carry both in one BIG and transmits two BIGs from two advertising sets where its controller supports that, and one of them otherwise. A Public Broadcast Source transmitting LC3plus **should** include a Program_Info LTV structure in the Metadata of each subgroup, so that a listener can tell the LC3plus subgroup from an LC3 subgroup, and **should** include at least one subgroup with a configuration that (7), Table 6.4 defines as mandatory for a Broadcast Sink.

A Public Broadcast Sink or Public Broadcast Assistant that performs the Basic Audio Announcement discovery of (12), sections 3.3 and 3.4, **shall not** pass over a Public

Broadcast Announcement because its bit 1 and bit 2 are both 0b0, unless it knows the sink does not support LC3plus; it reads the BASE.

3.2.10.5 Presentation delay

Presentation_Delay sits at level 1 of the BASE, one value for the whole BIG. A Broadcast Source **shall** set it to the presentation delay of the Table 37 setting it uses, and to the larger value where the subgroups of one BIG carry settings with different presentation delays and no profile of section 3.2.11 fixes the value of one of them. A Broadcast Sink that supports a setting of Table 37 **shall** be capable of rendering audio no later than its presentation delay after the SDU synchronization reference, and **shall not** synchronize to a BIS whose BASE carries a presentation delay it cannot meet.

3.2.10.6 Broadcast Assistant

The Broadcast Audio Scan Service (18) carries no Codec_ID, so a Broadcast Sink learns the coding format only from the BASE. A Broadcast Assistant that directs a sink to an LC3plus subgroup **shall**, before it initiates an Add Source or a Modify Source operation that requests synchronization to a BIS of that subgroup, compare all five octets of Codec_ID[i] and the Codec_Specific_Configuration of that subgroup against the sink's Sink PAC records, and **shall not** direct the sink to that subgroup where none of them covers it; it **may** direct such a sink to an LC3 subgroup where the BIG carries one. A Broadcast Sink directed by such an operation to a BIS it does not synchronize to under section 3.2.5 or 3.2.10.5 treats it as a synchronization that failed ((18), sections 3.1.1.4 and 3.1.1.5).

3.2.11 Coexistence with the application profiles

A device conformant to (13), (14) or (15) that supports LC3plus **shall** meet the profile's mandatory settings with LC3 and **shall not** replace them with LC3plus. The latency requirements of those profiles apply to their own settings, except that (15), section 3.7 requires a hearing aid to expose a Presentation_Delay range that includes 20 ms in addition to the presentation delays section 3.2.9 requires of it.

An Initiator of the Common Audio Profile (16) **shall not** start a unicast LC3plus stream to a Coordinated Set (19) unless every connected member it starts a stream with in that direction exposes LC3plus in a PAC record for that direction, and a Commander **shall not** direct a Coordinated Set to an LC3plus subgroup unless every connected member exposes a Sink PAC record covering its configuration. The Initiator **may** then use LC3 or another codec those members expose; the Commander does not proceed, as (16), section 7.3.1.8.1 states, and **may** direct the set to an LC3 subgroup where the BIG carries one.

NOTE: (15), Tables 3.1 and 3.2 require a hearing aid to accept a flush timeout of 1 and a burst number of 1. With a flush timeout of 1 only 16_3_LC3plus_Normal_LowLatency fits a hearing aid pair with one stream per ear, and it fills the ISO interval, 2 × 2500 microseconds. Every setting that section 3.2.8 makes mandatory fits such a pair at a flush timeout of 5, and at 4 with a burst number of 2, except 32_3_LC3plus_Normal_LowLatency, which fits at 2 only.

3.3 BIT RATE ADAPTATION

3.3.1 Overview

A source **may** adapt the bit rate over A2DP and on a unicast Audio Stream over LE Audio. A Broadcast Source **shall** select its operating point before the stream starts and **shall not** adapt it. Every LC3plus decoder derives the bit rate from the size of each frame ((1), clause 5.7), so a change needs no in-band signaling. An adapting source **shall** start a new stream on the base value of section 3.3.2 for the selected combination, raised where section 3.3.3 raises it, or on L0 where L0 lies below it or the SRC claims the Hi-Res Audio Wireless logo for the stream (sections 3.1.6.1, 3.3.3 and 3.3.6.2), and **shall not** start higher. Over A2DP a Reconfigure command that changes the sampling frequency, the frame duration, the sample resolution or the Audio Channel Count starts a new stream in this sense. A non-adapting source **shall** keep one octet count for the lifetime of the stream: over LE Audio the configured Octets_Per_Codec_Frame, in Lossless CBR mode no higher than the maximum of section 3.3.2 for the selected sample resolution; over A2DP one bit rate limit inside the range of section 3.1.5.8 that the transport sustains. An adapting source **shall** change the bit rate only between the levels of section 3.3.3; it **should** reduce it on a congested transport and **may** raise it on an uncongested one. Section 3.3.4 recommends how to detect congestion, and section 3.3.5 when to change the level. Where the ladder holds one level only, a source **shall not** adapt.

3.3.2 Operating points per combination

The tables of this section state, per audio channel and codec frame, in octets:

- **Lowest value:** the octet count the reference encoder of (2) produces at the lowest bit rate it accepts.
- **Base:** the lowest octet count of regular operation.
- **Maximum:** the highest octet count the mode allows for the combination; in Lossless CBR mode per sample resolution.

Sampling frequency	Frame duration	Lowest value	Base	Maximum
16000 Hz	5 ms	20 (32)	24 (38.4)	200 (320)
24000 Hz	5 ms	20 (32)	36 (57.6)	200 (320)
32000 Hz	5 ms	20 (32)	48 (76.8)	200 (320)

Table 38: Operating points of Normal mode, octets per codec frame and audio channel, with the bit rate in kbit/s in parentheses.

The lowest value and the maximum of Table 38 are the octet range of (1), Table 5.1. The base value is six tenths of the 10 ms value that (7), Table 3.11 gives for the same sampling frequency, a fifth more bit rate than that LC3 setting.

Sampling frequency	Frame duration	Lowest value	Base	Maximum, 16 bit	Maximum, 24 bit
44100 Hz	10 ms	84 (61.7)	156 (114.7)	1020	1500
44100 Hz	7.5 ms	63 (61.7)	117 (114.7)	765	1125
44100 Hz	5 ms	50 (73.5)	93 (136.7)	510	750
44100 Hz	2.5 ms	29 (85.3)	54 (158.8)	255	375
48000 Hz	10 ms	78 (62.4)	156 (124.8)	1020	1500
48000 Hz	7.5 ms	58 (61.9)	117 (124.8)	765	1125
48000 Hz	5 ms	46 (73.6)	93 (148.8)	510	750
48000 Hz	2.5 ms	27 (86.4)	54 (172.8)	255	375
96000 Hz	10 ms	93 (74.4)	187 (149.6)	2040	3000
96000 Hz	7.5 ms	69 (73.6)	141 (150.4)	1530	2250
96000 Hz	5 ms	54 (86.4)	109 (174.4)	1020	1500
96000 Hz	2.5 ms	31 (99.2)	62 (198.4)	510	750
192000 Hz	10 ms	124 (99.2)	249 (199.2)	4080	6000
192000 Hz	7.5 ms	93 (99.2)	189 (201.6)	3060	4500
192000 Hz	5 ms	70 (112)	141 (225.6)	2040	3000
192000 Hz	2.5 ms	39 (124.8)	78 (249.6)	1020	1500

Table 39: Operating points of Lossless CBR mode, octets per codec frame and audio channel, with the bit rate in kbit/s in parentheses for the lowest value and the base. The maximum corresponds to 749.7, 816, 1632 and 3264 kbit/s at 16 bit and 1102.5, 1200, 2400 and 4800 kbit/s at 24 bit per audio channel at 44.1, 48, 96 and 192 kHz.

NOTE: The lowest value is not half the base value in every row.

The base column follows (1), Table 5.2, with the 48000 Hz values at 44100 Hz and continued for 192000 Hz, the maximum (1), Table 5.2a. A device **may** announce the maximum plus two octets, and a client reading the record **shall** accept that upper end.

A source that drives the reference encoder of (2) by bit rate sets, on a level at the lowest value, the lowest bit rate that encoder accepts for one audio channel, in an encoder

instance per audio channel; at 44100 Hz that is the lowest bit rate of 48000 Hz, unscaled.

3.3.3 The rate ladder

A source **shall** derive the operating points of a stream from the base value `nbytes_base` of Table 38 or Table 39 as Table 40 states.

Level	Octets per frame	Meaning
L0	`nbytes_max`	Highest level. Transport dependent, as stated below.
Ln	transport dependent	Optional intermediate levels, see below.
L1	`nbytes_base`	Base level. A new stream starts here unless section 3.3.1 starts it on L0.
F1	`floor(3 x nbytes_base / 4)`	First fallback level.
F2	`floor(nbytes_base / 2)`	Second fallback level, and the lowest level of the stream.

Table 40: The rate ladder of one stream.

NOTE: In Lossless CBR mode at 48000 and 96000 Hz F2 is the fallback value `floor(nbytes_min / 2)` of TS 103 634 (1), Table 5.2, whose `nbytes_min` is the base value. At 44100 Hz `floor(nbytes_base / 2)` lies below the lowest value, and F2 is raised to it. In Normal mode TS 103 634 states no fallback value.

Every level **shall** lie between the lowest value and the maximum of Table 38 or Table 39 for the selected combination and sample resolution. Over LE Audio every level of a stream on a Sink ASE **shall** also lie inside the range that section 3.3.7 gives for the sink. A level below the lower end of these bounds **shall** be raised to it, and where that makes two levels equal, the stream has one level fewer. The levels of a stream are ordered by octet count; one level up is the next higher octet count.

Over A2DP, L0 is the smallest of the maximum of the combination, the 765 octets of section 3.1.5.8, the largest frame data block the configured MTU carries under section 3.1.5.7 divided by the number of audio channels, at 2.5 ms also the largest frame data block that one media packet in a three-slot baseband packet of the connection carries divided by the number of audio channels (section 3.1.3.3), and, on a stream for which the SRC claims the Hi-Res Audio Wireless logo under section 3.1.6.1, the value stated there for its frame duration. A source **should** insert intermediate levels between L1 and L0, one per packetization that fills the MTU less the overhead of section 3.1.6: one frame data block per media payload, then, except at 2.5 ms, two and more aggregated blocks, then, at 10 ms only, one block per media payload fragmented over two and more media packets.

Over LE Audio L0 is the configured Octets_Per_Codec_Frame, at most the maximum of the combination and sample resolution, and a stream has those of L0, L1, F1 and F2 that lie at or below it. With the settings of Table 32 L0 equals the base value, so the level starts there and never rises above it. A source on a Sink ASE intending to adapt

by the shorter SDU of section 3.3.6.2 **shall** configure Octets_Per_Codec_Frame at the highest octet count it intends to send, which **shall not** exceed the maximum of the combination.

3.3.4 Detecting a congested transport

This section and section 3.3.5 are recommendations: a source that adapts the bit rate **should** follow them and **shall not** infer congestion from the audio content.

3.3.4.1 A2DP

The evaluation interval over A2DP is the interval at which a source reads the transmit queue fill level; it need not be the period of the task that encodes audio. A source **should** set it to 20 milliseconds and keep it constant for as long as the configuration stays.

Quantity	What it reports	Use
Transmit queue fill level	Number of media packets queued for the transport and not taken by it yet	Primary indication.
AVDTP Receiver Report	Fraction Lost, Cumulative Number of Packets Lost and Inter-arrival Jitter, reported by the SNK	Optional. Needs the AVDTP Reporting service and a second transport session.

Table 41: Congestion indications available to an A2DP source.

A source **should** use the transmit queue fill level of Table 41, read once per evaluation interval before encoding; a source that encodes in the controller uses the queue of the controller. The AVDTP Receiver Report and further indications **may** be used in addition.

3.3.4.2 LE Audio

The isochronous transport exposes counters per stream, read with the HCI command LE Read ISO Link Quality, (9), Vol 4 Part E, section 7.8.116, which a controller supports optionally. They are 32 bit and only ever increase, so a source evaluates the difference between two reads modulo the 32 bit counter range.

Counter	Available on	Use for a source
TX_UnACKed_Packets	CIS	Primary. A PDU sent at least once was not acknowledged by its flush point.
TX_Flushed_Packets	CIS	Primary. A payload was not sent by its flush point.
Retransmitted_Packets	CIS	Early. Rises before an SDU is lost.
TX_Last_Subevent_Packets	CIS, Peripheral only	Early. A PDU was sent in the last subevent.

Table 42: Counters of LE Read ISO Link Quality that a source can use. The command reports three further counters, CRC_Error_Packets, RX_Unreceived_Packets and Duplicate_Packets, which apply to the receive direction.

A source on a unicast Audio Stream **should** use TX_UnACKed_Packets and TX_Flushed_Packets of Table 42, read once per counter read interval of Table 43. Its congestion indication is the sum of the two increments of one read interval. A source **may** use Retransmitted_Packets and TX_Last_Subevent_Packets as early indications. A source whose encoder runs in the controller reads these counters there. A Broadcast Source has no indication, because the command applies to a BIS only on a Synchronized Receiver.

3.3.5 The control loop

A source that adapts the bit rate **should** evaluate the congestion indication of section 3.3.4 once per evaluation interval and **should** apply the parameters of Table 43 and the transitions of Table 44. Over A2DP the evaluation interval is that of section 3.3.4.1, over LE Audio the counter read interval.

Parameter	Value	Unit	Applies to
Reduction threshold	5	queued media packets	A2DP
Increase threshold	2	queued media packets	A2DP
Counter read interval	500	milliseconds	LE Audio, Unicast
Reduction threshold	1	unacknowledged or flushed packets per read interval	LE Audio, Unicast
Increase threshold	0	unacknowledged or flushed packets per read interval	LE Audio, Unicast
Observation window	4	evaluation intervals	A2DP and LE Audio, Unicast
Hold time before an increase that is not from a fallback level	10	seconds	A2DP and LE Audio, Unicast
Hold time before an increase from a fallback level	2	seconds	A2DP and LE Audio, Unicast
Hold time before a reduction into or inside the fallback range	40	milliseconds	A2DP
Hold time before a reduction into or inside the fallback range	500	milliseconds	LE Audio, Unicast

Table 43: Parameters of the control loop.

A source **may** double the hold time before an increase for every reduction already made on this stream, up to a factor of four. All hold times of a stream run on one clock, which restarts when the operating point changes or when the conditions for leaving a level are met but the stream has no further level in that direction, and not otherwise.

The indication is still rising where the value of the current evaluation interval is above that of the previous one; the first value of a stream is not rising. While the observation window holds fewer values than it has intervals, the condition on the whole window is not met. The observation window keeps its values when the operating point changes.

State	Operating point	Leaving upwards	Leaving downwards
Regular	L0 or an intermediate level	Indication at or below the increase threshold and hold time before an increase not from a fallback level elapsed: one level up	Indication above the reduction threshold: to L1
Base	L1	as above	Indication above the reduction threshold, hold time before a reduction elapsed, and indication still rising or, throughout the observation window, above that threshold: to F1
Fallback	F1, then F2	Indication at or below the increase threshold and hold time before an increase from a fallback level elapsed: one level up	as above, down to F2 at most

Table 44: States of the control loop and every transition between them.

A new stream starts in the state Base, or in the state Regular where it starts on L0 above L1, with an empty observation window. Where L0 lies below L1, L0 takes the place of L1 in Table 44, and a step down from it leads to the next lower level of the stream.

3.3.6 Applying a change to a running stream

A source **shall not** use padding as a reaction to a congested transport; padding keeps the octet count on air. Lowering the bit rate limit of the encoder to a level's octet count is not padding in that sense.

3.3.6.1 A2DP

Over A2DP a source changes the bit rate without renegotiation; the Table of Contents carries the length of every frame data. The bounds of sections 3.1.5.8 and 3.1.6 apply to every level. A source **should** prefer, among levels of equal octet count, the packetization with the smaller delay, and **should not** change the packetization more often than the octet count.

3.3.6.2 LE Audio

Over LE Audio Octets_Per_Codec_Frame and Max_SDU cannot be changed while the stream runs. An adapting source **shall** adapt by sending codec frames shorter than Octets_Per_Codec_Frame, and with them an SDU shorter than Max_SDU, where the conditions below are met. Where they are not met, a source on a Sink ASE **shall** renegotiate instead, and a source on a Source ASE **shall not** adapt.

A shorter SDU. The isochronous adaptation layer accepts an SDU of any length up to Max_SDU ((9), Vol 6 Part G, section 1). A source **shall not** submit an SDU above Max_SDU. All of the following conditions **shall** be met:

- The source **shall** send an SDU length that is a multiple of the number of codec frames the SDU carries, which is Codec_Frame_Blocks_Per_SDU times the number of audio channels of the Audio_Channel_Allocation of section 3.2.4.
- The device that configures the stream **shall** set Max_SDU to Octets_Per_Codec_Frame times the number of codec frames the SDU carries and **shall** choose Octets_Per_Codec_Frame so that this product does not exceed 4095.
- The source **shall** operate the encoder for all codec frames of an SDU with the octet count of one level as its bit rate limit; in Lossless CBR mode a codec frame reaches at most that count, and section 3.2.7 states how far a shorter frame is filled.

Apart from Max_SDU, the QoS parameters are not a condition for a shorter SDU. The device that writes the Config QoS operation takes an SDU interval of Codec_Frame_Blocks_Per_SDU times the frame period of section 2.1.1, requests framed operation where a stream of its CIG runs at 44100 Hz and unframed operation otherwise, and **should** take the retransmission number, maximum transport latency and presentation delay that section 3.2.9 gives for that frame duration in the HighReliability or the LowLatency setting, within the values the server exposes in the Codec Configured state.

A Source ASE. On a Source ASE the Unicast Server sends. It cannot renegotiate, because (10), Table 3.2 gives the Config QoS and the Enable operation to the Unicast Client alone. L0 follows section 3.3.3, whichever role wrote Octets_Per_Codec_Frame. The floor is the lowest value of Table 38 or Table 39 for the configured combination.

Renegotiation. On a Sink ASE the source disables the ASE, terminates every CIS and removes the CIG, which is the only way to change Max_SDU ((9), Vol 6 Part B, section 4.5.14.3), and runs the Config Codec, Config QoS and Enable operations again. Renegotiation interrupts every stream of the group and **shall** be used only where a condition above is not met, or where the change is not a reaction to the transport but a change of content.

3.3.7 Shorter SDU and codec frame length

A decoder **shall** accept a codec frame of any octet count inside the range of its stream and **shall not** require re-initialization where the octet count changes from one codec frame to the next. On a Sink ASE and on a broadcast Audio Stream the range runs from the lowest minimum to the highest maximum that the

LC3plus_Supported_Octets_Per_Codec_Frame structures for the frame duration in use announce in the sink's Sink PAC records whose values cover the configuration. On a Source ASE the range runs from the floor of section 3.3.6.2 up to the configured Octets_Per_Codec_Frame. A decoder **shall** accept an SDU shorter than the length the Codec_Specific_Configuration gives it, and **shall** take the codec frame length from the SDU length divided by the number of codec frames the configuration gives the SDU. A decoder **shall** discard an SDU whose length is not a multiple of that number or whose derived frame length lies outside the range of its stream, and **shall** conceal the codec frames of an SDU it discards or does not receive.

4 BIBLIOGRAPHY

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ANNEX A (INFORMATIVE)

A.1 A2DP TRANSPORT CONSIDERATIONS

The throughput available to A2DP depends on the Bluetooth packet type, the link quality and the activity of other links on the same controller. The values of Table 45 are not guaranteed by any specification.

Condition	Sustained throughput	Stereo, 10 ms	Per channel	Stereo, 44.1 kHz	Per channel
Conservative	650 kbit/s	812	406	884	442
Typical	990 kbit/s	1237	618	1346	673
Good conditions	1200 kbit/s	1500	750	1632	816

Table 45: Throughput of A2DP, in octets of transport payload per frame period. The 10 ms columns apply to 48000, 96000 and 192000 Hz, the 44.1 kHz columns to the frame period of 10.884 ms of section 2.1.1.

The figures are transport octets, not frame data; the overhead of section 3.1.6 is 18 to 22 octets for one frame data block of a stereo stream in one media packet. A frame data block of two audio channels fits one media packet of the 3-DH5 MTU of 1017 octets up to 498 octets (398.4 kbit/s at 10 ms) per channel, of the 2-DH5 MTU of 675 octets up to 327 octets (261.6 kbit/s at 10 ms) and of the DH5 MTU of 335 octets up to 158 octets (126.4 kbit/s at 10 ms). Above that, at 10 ms only, the media payload is fragmented (section 3.1.5.7). Table 46 shows typical configurations.

Sampling frequency	Frame duration	Octets per frame and channel	Bit rate per channel	Blocks per payload	Media packets
44100 Hz	10 ms	160	117.6 kbit/s	2	1
48000 Hz	10 ms	160	128 kbit/s	2	1
48000 Hz	10 ms	310	248 kbit/s	1	1
48000 Hz	10 ms	563	450.4 kbit/s	1	2
48000 Hz	7.5 ms	117	124.8 kbit/s	2	1
96000 Hz	10 ms	190	152 kbit/s	2	1
96000 Hz	10 ms	310	248 kbit/s	1	1
96000 Hz	5 ms	120	192 kbit/s	4	1
192000 Hz	10 ms	563	450.4 kbit/s	1	2

Table 46: Typical configurations over A2DP in Lossless CBR mode, for two audio channels at the MTU of 1017 octets of a 3-DH5 connection. The last two columns follow the aggregation rule of section 3.1.6.

ANNEX B (INFORMATIVE)

B.1 LE AUDIO TRANSPORT CONSIDERATIONS

The airtime of one CIS subevent on the LE 2M PHY follows from

```
MPT       = 44 + 4 x (N + MIC)
SE_Length = MPT_C + T_IFS_CIS + MPT_P + T_MSS_CIS
```

where N is the payload size in octets and MIC is 4 on an encrypted CIS or BIS with a payload and 0 otherwise. The figures of this document use MIC equal to 4 where the payload is not empty; the device that sends no audio sends only empty PDUs, so its MPT is 44 microseconds. T_IFS_CIS and T_MSS_CIS default to 150 microseconds each ((9), Vol 6 Part B, sections 4.1.1 and 4.1.3); a BIS has only T_MSS_150, fixed at 150 microseconds. The subevents of one CIS fit into the ISO interval: NSE times Sub_Interval is at most the ISO interval, and SE_Length at most Sub_Interval ((9), Vol 6 Part B, section 4.5.13.2); several CISes of one CIG share the interval ((9), Vol 6 Part B, section 4.5.14.2). An SDU larger than Max_PDU, at most 251 octets, is split across several PDUs, which raises BN and the airtime. Table 47 gives the resulting ceiling for one CIS.

ISO interval	Largest SDU, no retransmission	Per channel, stereo in one stream (kbit/s)	Net rate	Largest SDU with 47 percent margin
10 ms	1757	878 (702.4)	1405.6 kbit/s	920
7.5 ms	1266	633 (675.2)	1350.4 kbit/s	690
5 ms	844	422 (675.2)	1350.4 kbit/s	460

Table 47: Airtime ceiling of LE Audio on the LE 2M PHY, one CIS that carries audio in one direction, with 150 microseconds of frame space. The airtime is NSE times one subevent length; Max_PDU is the SDU size divided by BN and rounded up.

The last column states the largest SDU that leaves 47 percent of the interval unused, an example of a margin for the ACL connection that the controller of a sink also serves. Max_SDU is a 12 bit field, so stereo in one stream caps at 2047 octets (1637.6 kbit/s at 10 ms) per channel, and it is fixed for the lifetime of a stream (section 3.3.6.2).

The controller selects NSE, BN and FT for the RTN of section 3.2.9 so that

```
NSE x FT - floor(NSE / BN) x (BN - 1) >= RTN + 1
```

((9), Vol 4 Part E, section 7.8.97), and a setting fits only where such values also fit the airtime into the ISO interval ((9), Vol 6 Part B, sections 4.5.13.2 and 4.5.14.2). Table 48 gives one solution per setting.

Configuration	SDU, two channels (kbit/s)	BN	NSE	FT	Airtime per event	Airtime share	Flush span
16_3_LC3plus_Normal_HighReliability	48 (76.8)	1	2	7	1192 µs	24 %	30 ms
24_3_LC3plus_Normal_HighReliability	72 (115.2)	1	2	7	1384 µs	28 %	30 ms
32_3_LC3plus_Normal_HighReliability	96 (153.6)	1	2	7	1576 µs	32 %	30 ms
16_3_LC3plus_Normal_LowLatency	48 (76.8)	1	3	2	1788 µs	36 %	5 ms
24_3_LC3plus_Normal_LowLatency	72 (115.2)	1	3	2	2076 µs	42 %	5 ms
32_3_LC3plus_Normal_LowLatency	96 (153.6)	1	3	2	2364 µs	47 %	5 ms
48_1_LC3plus_Lossless_HighReliability	320 (256)	2	4	4	4176 µs	42 %	30 ms
48_2_LC3plus_Lossless_HighReliability	620 (496)	3	6	3	7392 µs	74 %	20 ms
48_3_LC3plus_Lossless_HighReliability	234 (249.6)	1	2	7	2680 µs	36 %	45 ms
48_4_LC3plus_Lossless_HighReliability	360 (384)	2	4	4	4496 µs	60 %	22.5 ms
96_1_LC3plus_Lossless_HighReliability	380 (304)	2	4	4	4656 µs	47 %	30 ms
96_2_LC3plus_Lossless_HighReliability	620 (496)	3	6	3	7392 µs	74 %	20 ms
96_3_LC3plus_Lossless_HighReliability	282 (300.8)	2	4	4	3872 µs	52 %	22.5 ms
96_4_LC3plus_Lossless_HighReliability	450 (480)	2	4	4	5216 µs	70 %	22.5 ms
441_1_LC3plus_Lossless_HighReliability	1000 (735)	5	6	3	7344 µs	73 %	20 ms
441_2_LC3plus_Lossless_HighReliability	480 (352.8)	2	4	4	5536 µs	55 %	30 ms
441_3_LC3plus_Lossless_HighReliability	750 (735)	4	5	4	5880 µs	78 %	22.5 ms
48_6_LC3plus_Lossless_HighReliability	900 (720)	4	5	4	6520 µs	65 %	30 ms
192_1_LC3plus_Lossless_HighReliability	498 (398.4)	2	4	4	5600 µs	56 %	30 ms

Configuration	SDU, two channels (kbit/s)	BN	NSE	FT	Airtime per event	Airtime share	Flush span
48_5_LC3plus_Lossless_LowLatency	240 (384)	2	4	2	3536 µs	71 %	5 ms
96_5_LC3plus_Lossless_LowLatency	240 (384)	2	4	2	3536 µs	71 %	5 ms

Table 48: Airtime of the settings of Table 34 for both audio channels in one stream on the LE 2M PHY, informative. The bit rate in kbit/s in parentheses is that of the whole SDU, both audio channels. Airtime per event is NSE times one subevent; the PDU payload is the SDU size divided by BN and rounded up, plus 5 octets in framed mode. The airtime share is the part of the ISO interval one CIS event occupies, the flush span (FT - 1) times the ISO interval. BN, NSE and FT follow the rule stated below this table; they are informative, not a requirement.

Each row takes the smallest airtime and, for that airtime, the smallest FT among the values of BN, NSE and FT that satisfy the inequality, fit the ISO interval, keep the transport latency within the maximum transport latency of Table 34 and have `NSE >= 2 x BN`, so that every PDU of an event has one spare subevent. For 441_1_LC3plus_Lossless_HighReliability, 441_3_LC3plus_Lossless_HighReliability and 48_6_LC3plus_Lossless_HighReliability there are no such values; their rows take the smallest airtime with `NSE >= BN + 1` instead and leave no margin for a second stream.

Of these settings, nine leave room for a second stream of two audio channels with the same BN, NSE and FT, at 24 to 47 percent of the ISO interval each. For 48_5_LC3plus_Lossless_LowLatency and 96_5_LC3plus_Lossless_LowLatency a larger FT exceeds the maximum transport latency of 10 ms, so a group with two of their streams needs fewer octets per codec frame or a lower retransmission number. For 441_1_LC3plus_Lossless_HighReliability, 441_3_LC3plus_Lossless_HighReliability and 48_6_LC3plus_Lossless_HighReliability no larger FT helps, because their smallest airtime with `NSE >= BN`, at any FT within the maximum transport latency, is more than half of the ISO interval; a group with two of their streams needs fewer octets per codec frame. For the other seven settings a group with two streams needs a larger FT with a smaller NSE, or fewer octets per codec frame.

A pair of earbuds uses two CISes of one CIG, one audio channel each. Table 49 applies the same rule, with the two CISes following one another in the ISO interval.

Configuration	SDU, one channel (kbit/s)	BN	NSE	FT	Airtime per event	Airtime share	Flush span
16_3_LC3plus_Normal_HighReliability	24 (38.4)	1	2	7	2000 µs	40 %	30 ms
24_3_LC3plus_Normal_HighReliability	36 (57.6)	1	2	7	2192 µs	44 %	30 ms
32_3_LC3plus_Normal_HighReliability	48 (76.8)	1	2	7	2384 µs	48 %	30 ms

Configuration	SDU, one channel (kbit/s)	BN	NSE	FT	Airtime per event	Airtime share	Flush span
16_3_LC3plus_Normal_LowLatency	24 (38.4)	1	3	2	3000 µs	60 %	5 ms
24_3_LC3plus_Normal_LowLatency	36 (57.6)	1	3	2	3288 µs	66 %	5 ms
32_3_LC3plus_Normal_LowLatency	48 (76.8)	1	3	2	3576 µs	72 %	5 ms
48_1_LC3plus_Lossless_HighReliability	160 (128)	1	2	7	4176 µs	42 %	60 ms
48_2_LC3plus_Lossless_HighReliability	310 (248)	2	4	4	8192 µs	82 %	30 ms
48_3_LC3plus_Lossless_HighReliability	117 (124.8)	1	2	7	3488 µs	47 %	45 ms
48_4_LC3plus_Lossless_HighReliability	180 (192)	1	2	7	4496 µs	60 %	45 ms
96_1_LC3plus_Lossless_HighReliability	190 (152)	1	2	7	4656 µs	47 %	60 ms
96_2_LC3plus_Lossless_HighReliability	310 (248)	2	4	4	8192 µs	82 %	30 ms
96_3_LC3plus_Lossless_HighReliability	141 (150.4)	1	2	7	3872 µs	52 %	45 ms
96_4_LC3plus_Lossless_HighReliability	225 (240)	1	2	7	5216 µs	70 %	45 ms
441_1_LC3plus_Lossless_HighReliability	500 (367.5)	3	4	4	8736 µs	87 %	30 ms
441_2_LC3plus_Lossless_HighReliability	240 (176.4)	1	2	7	5536 µs	55 %	60 ms
441_3_LC3plus_Lossless_HighReliability	375 (367.5)	2	3	5	7056 µs	94 %	30 ms
48_6_LC3plus_Lossless_HighReliability	450 (360)	2	3	5	7824 µs	78 %	40 ms
192_1_LC3plus_Lossless_HighReliability	249 (199.2)	1	2	7	5600 µs	56 %	60 ms

Configuration	SDU, one channel (kbit/s)	BN	NSE	FT	Airtime per event	Airtime share	Flush span
48_5_LC3plus_Lossless_LowLatency	120 (192)	2	3	2	3864 µs	77 %	5 ms
96_5_LC3plus_Lossless_LowLatency	120 (192)	2	3	2	3864 µs	77 %	5 ms

Table 49: Airtime of the settings of Table 34 for a pair of earbuds on the LE 2M PHY, informative: two CISes of one CIG, one audio channel each, same settings, airtime per ISO event. The bit rate in kbit/s in parentheses is that of one audio channel. Airtime per event is two times NSE times one subevent, the two CISes following one another; the PDU payload is the SDU size divided by BN and rounded up, plus 5 octets in framed mode. The airtime share is the part of the ISO interval both CIS events occupy, the flush span (FT - 1) times the ISO interval. BN, NSE and FT follow the rule stated below Table 48 and are informative, not a requirement.

For 441_1_LC3plus_Lossless_HighReliability, 441_3_LC3plus_Lossless_HighReliability, 96_5_LC3plus_Lossless_LowLatency, 48_5_LC3plus_Lossless_LowLatency and 48_6_LC3plus_Lossless_HighReliability there are no such values either; their rows take the smallest airtime with `NSE >= BN + 1`, 77 to 94 percent of the ISO interval, with less than one spare subevent per PDU.

With the same formulas, a pair of earbuds with one CIS each carries 48_5_LC3plus_Lossless_LowLatency at RTN 4 with BN 2 and at RTN 3 at most with BN 1, and a headset with stereo 48_5_LC3plus_Lossless_LowLatency and a 32_3_LC3plus_Normal_LowLatency microphone in one CIS with BN 1 and NSE 3 fills 94 percent of the ISO interval.

B.2 BROADCAST AIRTIME

The airtime a BIG occupies follows from the packet time MPT of Annex B.1 and the number of subevents:

```
Airtime = Num_BIS x NSE x (MPT + T_MSS_150)
```

A BIG event that also carries a BIG Control PDU needs one further subevent of 242 microseconds: the packet time of the longest BIG Control PDU, BIG_CHANNEL_MAP_IND (8 octets), and the MIC, plus the closing T_MSS_150 ((9), Vol 6 Part B, sections 2.6.3.1, 4.4.6.4 and 4.4.6.7). Table 50 gives the largest octets per codec frame per channel that fit an ISO interval of 10 ms with that subevent, which it reserves in every event, because a BIG Control PDU, for a channel map update or a termination, can fall into any event and BIG_Control_Offset follows from the fixed BIG parameters.

Num_BIS	RTN 0	RTN 1	RTN 2	RTN 3	RTN 4
1	1757 (1405.6)	1004 (803.2)	654 (523.2)	502 (401.6)	382 (305.6)
2	1004 (803.2)	502 (401.6)	300 (240)	251 (200.8)	191 (152.8)
4	502 (401.6)	251 (200.8)	150 (120)	99 (79.2)	69 (55.2)
6	300 (240)	150 (120)	83 (66.4)	49 (39.2)	28 (22.4)
8	251 (200.8)	99 (79.2)	49 (39.2)	23 (18.4)	8 (6.4)

Table 50: Largest number of octets per codec frame per channel that fits an ISO interval of 10 ms together with a control subevent, with the bit rate in kbit/s in parentheses, one audio channel per BIS, LE 2M PHY, BN times (RTN + 1) subevents. BN of a BIS is at most 7 ((9), Vol 6 Part B, section 4.4.6.3), which caps the first cell at 7 PDUs of 251 octets.

With the formulas of this annex, a BIG with a stereo LC3 subgroup and a stereo 48_1_LC3plus_Lossless subgroup carries RTN 1, and two BIGs carry RTN 2 with a stereo 24_2 or 48_2 LC3 subgroup and RTN 1 with a stereo 48_6 LC3 subgroup; six BISes of 160 octets (128 kbit/s at 10 ms) carry RTN 0. Four BISes of a Lossless CBR setting carry RTN 1 at most.

ANNEX C (NORMATIVE)

C.1 OVERVIEW FRAUNHOFER CODEC IDs

Table 51 lists the Vendor Specific Codec IDs administered by Fraunhofer IIS.

Codec ID	Description
0x0001	LC3plus High Resolution. Frame durations 10 ms, 5 ms and 2.5 ms over A2DP, with 7.5 ms also over LE Audio. Channel controlled variable bit rate over A2DP, variable bit rate over LE Audio. Specified for A2DP and for LE Audio in separate documents.
0x0002	LC3plus High Resolution. Frame durations 10 ms, 5 ms, 2.5 ms. Constant bit rate support. Listed in Annex B of (20), specified in no document of this series.
0x0003	LC3plus. Normal mode, hrmode 0, at 16, 24 and 32 kHz with the 5 ms frame duration, over LE Audio only. Lossless CBR mode, hrmode 2, at 44.1, 48, 96 and 192 kHz with the 10 ms, 7.5 ms and 5 ms frame duration over both transports, and 2.5 ms over A2DP. Specified in this document for A2DP and for LE Audio.

Table 51: Vendor Specific Codec IDs administered by Fraunhofer IIS under Vendor ID 0x000008A9 and Company ID 0x08A9.

Codec IDs 0x0001 and 0x0002 remain valid; this document does not supersede them. A device **may** expose 0x0001 and 0x0003 at the same time, on separate stream endpoints over A2DP or in separate PAC records over LE Audio. The Codec IDs do not

share a format, and a device **shall not** conclude from a shared configuration index that a peer which decodes it under one Codec_ID also decodes it under another. The configuration indices 48_1 to 48_5 and 96_1 to 96_5 carry the same frame duration and octet count under both, but Codec ID 0x0001 codes them in the High Resolution mode of (1), clause 5.8, and this document in Lossless CBR mode, which (1), clause 5.10.1 states to be not compatible with that mode.

Differences to Codec ID 0x0001 ((20), (21)):

- **Codec Specific Information over A2DP.** 12 octets against 10; octet 8 and octet 9 are two fields here and one Sampling Frequency and Resolution field there; octet 10, Version and features, and octet 11, Operating Mode, are new.
- **Frame durations over A2DP.** The same grades for 10 ms and 5 ms; octet 6 bit 3 carries 7.5 ms here, which is RFA there.
- **Sampling frequencies over A2DP.** All four mandatory in a SRC, 44100 Hz, 48000 Hz and 96000 Hz mandatory and 192000 Hz optional in a SNK here; 48000 Hz and 96000 Hz mandatory there.
- **Operating mode.** Octet 11 over A2DP and the type 0xFB over LE Audio carry the value of the hrmode parameter of (2) here. 0x0001 signals no operating mode; it carries High Resolution mode, hrmode 1, only.
- **Sample resolution.** A field of its own over A2DP and an LTV structure over LE Audio here; in Lossless CBR mode both roles support 16 bit and 24 bit. 0x0001 signals no sample resolution and requires 24 bit signed integer or 32 bit floating point at the codec interface ((20), section 2.1.1).
- **Media packet header and payload over A2DP.** A Payload Type per combination of sampling frequency, frame duration and Audio Channel Count (section 3.1.4.5), a payload header and a Table of Contents (sections 3.1.5.2 to 3.1.5.5) here.
- **Delay Reporting.** Mandatory for SNK and SRC here, not mentioned there.
- **Codec_ID over LE Audio.** Octet 4 is RFU there ((21), Table 5) and 0x00 here.
- **LTV types over LE Audio.** This document uses the types 0xF8, 0xF9 and 0xFB, which 0x0001 does not define. The types 0xF1 to 0xF4 exist under both; bits 0 to 2 and the values 0x01 to 0x03 of 0xF1 and the octet ranges of 0xF2 to 0xF4 carry the same meaning. Bit 3 and bits 8 to 11 of 0xF1 and the type 0xF5 are free here, and the value 0x04 of 0xF1 is RFU.
- **Requirement grades over LE Audio.** Here the conditions of section 3.2.8 bind what each role encodes or decodes under Codec ID 0x0003 over LE Audio. In Lossless CBR mode the encoding role carries 48_1, 96_1 and, on a unicast Audio Stream, 441_2; the decoding role carries 48_1, 96_1 where it supports 96000 Hz, and on a unicast Audio Stream 441_2, 48_3 where it supports 7.5 ms, and 96_3 where it supports 96000 Hz and 7.5 ms. 0x0001 makes 96_1 mandatory for both roles and 48_1, 48_3 and 96_3 where a device supports their sampling frequency and frame duration ((21), Tables 12 and 13), and in broadcast 48_1 mandatory and 96_1 optional ((21), Tables 14, 15 and 17).
- **QoS values.** For 48_5 and 96_5 0x0001 recommends a maximum transport latency of 5 ms and a presentation delay of 40000 microseconds, this document 10 ms and 5000 microseconds on a Sink ASE, 10000 being mandatory for a sink that supports

the setting; on a Source ASE 10000; for broadcast 96_1 0x0001 recommends a retransmission number of 4, this document 3.

- **Shorter SDU over LE Audio.** 0x0001 requires a decoder to accept an SDU of a new size ((21), section 4.2); this document states the derivation of the frame length and the range in section 3.3.7.
- **Concealment method.** Over A2DP 0x0001 requires MDCT frame repetition with sign scrambling where the method selection chooses frequency domain concealment ((20), section 2.1.3); this document follows clauses 5.6 and 5.8.2 of (1) and adds no rule of its own.

VERSION HISTORY

V.1.0.0 2026-11 Initial release. Covers LC3plus under Vendor Specific Codec ID 0x0003 in one document: Lossless CBR mode over A2DP and over LE Audio, Normal mode over LE Audio.

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 35 years, the institute's Audio and Media Technologies division has been shaping the globally deployed standards and technologies in the fields of audio and moving picture production. Starting with the creation of mp3 and continuing with the co-development of AAC, almost all consumer electronic devices, computers and mobile phones are equipped with systems and technologies from us today. Meanwhile, a new generation of best-in-class media technologies – such as MPEG-H Audio, xHE-AAC, EVS, LC3/LC3plus, Symphoria, Sonamic and upHear – is elevating the user experience to new heights. Always taking into account the demands of the market, Fraunhofer IIS develops technology that makes memorable moments.

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.

We hear the bigger picture.

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