**3GPP TSG-SA WG4 Meeting #129-eS4-241442**

**e-Meeting, 19 – 23 August 2024**

|  |  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- | --- |
| *CR-Form-v12.3* | | | | | | | | |
| **CHANGE REQUEST** | | | | | | | | |
|  | | | | | | | | |
|  | **26.114** | **CR** | **0571** | **rev** | **-** | **Current version:** | **18.7.0** |  |
|  | | | | | | | | |
| *For* [***HE******LP***](http://www.3gpp.org/3G_Specs/CRs.htm#_blank)*on using this form: comprehensive instructions can be found at* [*http://www.3gpp.org/Change-Requests*](http://www.3gpp.org/Change-Requests)*.* | | | | | | | | |
|  | | | | | | | | |

|  |  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- | --- |
| ***Proposed change affects:*** | UICC apps |  | ME | **X** | Radio Access Network |  | Core Network |  |

|  |  |  |  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- |
|  | | | | | | | | | | |
| ***Title:*** | Clarifications on SDP settings at DCMTSI client. | | | | | | | | | |
|  |  | | | | | | | | | |
| ***Source to WG:*** | NTT | | | | | | | | | |
| ***Source to TSG:*** | S4 | | | | | | | | | |
|  |  | | | | | | | | | |
| ***Work item code:*** | 5G\_MEDIA\_MTSI\_ext, TEI19 | | | | |  | ***Date:*** | | | 2024-08-09 |
|  |  | | | |  | |  | | |  |
| ***Category:*** | F |  | | | | | ***Release:*** | | | Rel-19 |
|  | *Use one of the following categories:* ***F*** *(correction)* ***A*** *(mirror corresponding to a change in an earlier release)* ***B*** *(addition of feature),* ***C*** *(functional modification of feature)* ***D*** *(editorial modification)*  Detailed explanations of the above categories can be found in 3GPP [TR 21.900](http://www.3gpp.org/ftp/Specs/html-info/21900.htm). | | | | | | | | *Use one of the following releases: Rel-8 (Release 8) Rel-9 (Release 9) Rel-10 (Release 10) Rel-11 (Release 11) … Rel-17 (Release 17) Rel-18 (Release 18) Rel-19 (Release 19)  Rel-20 (Release 20)* | |
|  |  | | | | | | | | | |
| ***Reason for change:*** | | IETF RFC 8864 defines the data channles using TCP/DTLS/SCTP in addition to UDP/DTLS/SCTP for the "proto" value of SDP (m=application for data channel). Since the the DCMTSI can supports only UDP/DTLS/SCTP according to the current versions of TS 26.114, it should be clarified that the DCMTSI client uses UDP/DTLS/STCP as a 3GPP profile in order to avoid misimplementation of signalling.  As for the "fmt" value of SDP (m=application for data channel), IETF RFC 8864 states " WebRTC application will only use the "m=" line format "webrtc-datachannel" and will not use other formats in the "m=" line for other protocols such as T.38 [T38]. [RFC8841]", and TS 26.114 annex A (Examples of SDP offers and answers) uses this "webrtc-datachannel". To ensure the interoperability between MTSI and 3GPP network, it is proposed to clarify that "webrtc-datachannel" is set to the "fmt" value of data channel. | | | | | | | | |
|  | |  | | | | | | | | |
| ***Summary of change:*** | | Added IETF RTF 8841 as a referemce for WebRTC data channel.  Added the description for the format of SDP "m=" line for data channel. | | | | | | | | |
|  | |  | | | | | | | | |
| ***Consequences if not approved:*** | | Misalignment with IETF-defined protocol specifications and unclear specification leading misimplementation of SDP generations at DCMTSI and lack of interoperability. | | | | | | | | |
|  | |  | | | | | | | | |
| ***Clauses affected:*** | | 2, 6.10.1 | | | | | | | | |
|  | |  | | | | | | | | |
|  | | **Y** | **N** |  | | | |  | | |
| ***Other specs*** | |  | **X** | Other core specifications | | | | TS/TR ... CR ... | | |
| ***affected:*** | |  | **X** | Test specifications | | | | TS/TR ... CR ... | | |
| ***(show related CRs)*** | |  | **X** | O&M Specifications | | | | TS/TR ... CR ... | | |
|  | |  | | | | | | | | |
| ***Other comments:*** | |  | | | | | | | | |
|  | |  | | | | | | | | |
| ***This CR's revision history:*** | |  | | | | | | | | |

\* \* \* First Change \* \* \* \*

# 2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non‑specific.

- For a specific reference, subsequent revisions do not apply.

- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

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

[2] 3GPP TS 22.173: "IP Multimedia Core Network Subsystem (IMS) Multimedia Telephony Service and supplementary services; Stage 1".

[3] 3GPP TS 26.235: "Packet switched conversational multimedia applications; Default codecs".

[4] 3GPP TS 26.236: "Packet switched conversational multimedia applications; Transport protocols".

[5] 3GPP TR 26.914: "Multimedia telephony over IP Multimedia Subsystem (IMS); Optimization opportunities".

[6] 3GPP TR 22.973: "IMS Multimedia Telephony service; and supplementary services".

[7] 3GPP TS 24.229: "IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3".

[8] IETF RFC 4566 (2006): "SDP: Session Description Protocol", M. Handley, V. Jacobson and C. Perkins.

[9] IETF RFC 3550 (2003): "RTP: A Transport Protocol for Real-Time Applications", H. Schulzrinne, S. Casner, R. Frederick and V. Jacobson.

[10] IETF RFC 3551 (2003): "RTP Profile for Audio and Video Conferences with Minimal Control", H. Schulzrinne and S. Casner.

[11] 3GPP TS 26.071: "Mandatory Speech Codec speech processing functions; AMR Speech CODEC; General description".

[12] 3GPP TS 26.090: "Mandatory Speech Codec speech processing functions; Adaptive Multi-Rate (AMR) speech codec; Transcoding functions".

[13] 3GPP TS 26.073: "ANSI C code for the Adaptive Multi Rate (AMR) speech codec".

[14] 3GPP TS 26.104: "ANSI‑C code for the floating-point Adaptive Multi Rate (AMR) speech codec".

[15] 3GPP TS 26.093: "Mandatory speech codec speech processing functions; Adaptive Multi-Rate (AMR) speech codec; Source controlled rate operation".

[16] 3GPP TS 26.103: "Speech codec list for GSM and UMTS".

[17] 3GPP TS 26.171: "Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; General description".

[18] 3GPP TS 26.190: "Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Transcoding functions".

[19] 3GPP TS 26.173: "ANCI-C code for the Adaptive Multi Rate - Wideband (AMR-WB) speech codec".

[20] 3GPP TS 26.204: "Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; ANSI-C code".

[21] 3GPP TS 26.193: "Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Source controlled rate operation".

[22] Void.

[23] Void.

[24] Recommendation ITU-T H.264 (04/2017): "Advanced video coding for generic audiovisual services" | ISO/IEC 14496-10:2014: "Information technology – Coding of audio-visual objects – Part 10: Advanced Video Coding".

[25] IETF RFC 6184 (2011): "RTP Payload Format for H.264 Video", Y.-K. Wang, R. Even, T. Kristensen, R. Jesup.

[26] ITU-T Recommendation T.140 (02/1998): "Protocol for multimedia application text conversation".

[27] ITU-T Recommendation T.140 (02/2000): "Protocol for multimedia application text conversation - Addendum 1".

[28] IETF RFC 4867 (2007): "RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs", J. Sjoberg, M. Westerlund, A. Lakaniemi and Q. Xie.

[29] Void

[30] Void.

[31] IETF RFC 4103 (2005): "RTP Payload for Text Conversation", G. Hellstrom and P. Jones.

[32] Void.

[33] 3GPP TR 25.993: "Typical examples of Radio Access Bearers (RABs) and Radio Bearers (RBs) supported by Universal Terrestrial Radio Access (UTRA)".

[34] 3GPP TS 22.105: "Services and service capabilities".

[35] 3GPP TS 26.131: "Terminal acoustic characteristics for telephony; Requirements".

[36] 3GPP TS 26.132: "Speech and video telephony terminal acoustic test specification".

[37] 3GPP TS 28.062: "Inband Tandem Free Operation (TFO) of speech codecs; Service description; Stage 3".

[38] 3GPP TS 23.153: "Out of band transcoder control; Stage 2".

[39] IETF RFC 0768 (1980): "User Datagram Protocol", J. Postel.

[40] IETF RFC 4585 (2006): "Extended RTP Profile for Real-time Transport Control Protocol (RTCP) - Based Feedback (RTP/AVPF)", J. Ott, S. Wenger, N. Sato, C. Burmeister and J. Rey.

[41] RTP Tools: <http://www.cs.columbia.edu/IRT/software/rtptools/>.

[42] IETF RFC 3556 (2003): "Session Description Protocol (SDP) Bandwidth Modifiers for RTP Control Protocol (RTCP) Bandwidth", S. Casner.

[43] IETF RFC 5104 (2008): "Codec Control Messages in the RTP Audio-Visual Profile with Feedback (AVPF)", S. Wenger, U. Chandra, M. Westerlund and B. Burman.

[44] Void.

[45] 3GPP TS 26.111: "Codec for circuit switched multimedia telephony service; Modifications to H.324".

[46] 3GPP TS 23.172: "Technical realization of Circuit Switched (CS) multimedia service; UDI/RDI fallback and service modification; Stage 2".

[47] 3GPP TS 23.002: "Network Architecture".

[48] IETF RFC 3388 (2002): "Grouping of Media Lines in the Session Description Protocol (SDP)", G. Camarillo, G. Eriksson, J. Holler and H. Schulzrinne.

[49] IETF RFC 4102 (2005): "Registration of the text/red MIME Sub-Type", P. Jones.

[50] ITU-T H.248 (06/2000): "Packages for text conversation, fax and call discrimination".

[51] ETSI EG 202 320, v1.2.1 (2005-10): "Human Factors (HF); Duplex Universal Speech and Text (DUST) communications".

[52] 3GPP TS 26.226: "Cellular text telephone modem; General description".

[53] IETF RFC 4504 (2006): "SIP Telephony Device Requirements and Configuration", H. Sinnreich, Ed., S. Lass and C. Stredicke.

[54] ITU-T Recommendation V.151 (05/2006): "Procedures for end-to-end connection of analogue PSTN text telephones over an IP network utilizing text relay".

[55] ITU-T Recommendation V.152 (09/2010): "Procedures for supporting Voice Band Data over IP networks".

[56] IETF RFC 3448 (2003): "TCP Friendly Rate Control (TFRC): Protocol Specification", M. Handley, S. Floyd, J. Padhye and J. Widmer.

[57] 3GPP TS 24.173: "IMS Multimedia Telephony Communication Service and Supplementary Services".

[58] IETF RFC 3264 (2002): "An Offer/Answer Model with the Session Description Protocol (SDP)", J. Rosenberg and H. Schulzrinne.

[59] 3GPP TS 26.141: "IP Multimedia System (IMS) Messaging and Presence; Media formats and codecs".

[60] 3GPP TS 26.234: "Transparent end-to-end Packet-switched Streaming Service; Protocols and codecs".

[61] IETF RFC 4733 (2006): "RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals", H. Schulzrinne and T.Taylor.

[62] 3GPP TS 23.014: "Support of Dual Tone Multi-Frequency (DTMF) signalling".

[63] ETSI ES 201 235-2, v1.2.1: "Specification of Dual Tone Multi-Frequency (DTMF); Transmitters and Receivers; Part 2: Transmitters".

[64] 3GPP TS 23.107: "Quality of Service (QoS) concept and architecture".

[65] 3GPP TS 29.163: "Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks".

[66] Void.

[67] OMA-ERELD-DM-V1\_2-20070209-A: "Enabler Release Definition for OMA Device Management, Approved Version 1.2".

[68] Void.

[69] IETF RFC 5939 (2010): "Session Description Protocol (SDP) Capability Negotiation", F. Andreasen.

[70] Void

[71] IETF RFC 1952 (May 1996): "GZIP file format specification version 4.3", P. Deutsch.

[72] IETF RFC 2326 (1998): "Real Time Streaming Protocol (RTSP)".

[73] IETF RFC 2616 (June 1999): "Hypertext Transfer Protocol -- HTTP/1.1".

[74] 3GPP TS 26.346 "Multimedia Broadcast/Multicast Service (MBMS); Protocols and codecs".

[75] Void

[76] IETF RFC 6236 (2011): "Negotiation of Generic Image Attributes in the Session Description Protocol (SDP)", I. Johansson and K. Jung.

[77] ITU-T G.711 (11/1988): "Pulse code modulation (PCM) of voice frequencies".

[78] ITU-T G.722 (09/2012): "7 kHz audio-coding within 64 kbit/s".

[79] IETF RFC 4821 (2007): "Packetization Layer Path MTU Discovery".

[80] 3GPP TS 23.003: "Numbering, addressing and identification".

[81] IETF RFC 4796 (2007): "The Session Description Protocol (SDP) Content Attribute", J. Hautakorpi and G. Camarillo.

[82] 3GPP TS 24.247: "Messaging service using the IP Multimedia (IM) Core Network (CN) subsystem".

[83] IETF RFC 3168 (2001): "The Addition of Explicit Congestion Notification (ECN) to IP", K. Ramakrishnan, S. Floyd and D. Black.

[84] IETF RFC 6679 (2012): "Explicit Congestion Notification (ECN) for RTP over UDP", M. Westerlund, et. al.

[85] 3GPP TS 36.300: "Evolved Universal Terrestrial Radio Access (E-UTRA) and Evolved Universal Terrestrial Radio Access Network (E-UTRAN); Overall description".

[86] Void

[87] IETF RFC 5506 (2009): "Support for Reduced-Size Real-Time Transport Control Protocol (RTCP): Opportunities and Consequences".

[88] IETF RFC 3611 (2003): "RTP Control Protocol Extended Reports (RTCP XR) ", T. Friedman, R. Caceres and A. Clark.

[89] 3GPP TS 25.401: "UTRAN overall description".

[90] 3GPP TS 23.203: "Policy and charging control architecture".

[91] ITU-T Recommendation T.4 (07/2003): "Standardization of Group 3 facsimile terminals for document transmission".

[92] ITU-T Recommendation T.30 (09/2005): "Procedures for document facsimile transmission in the general switched telephone network".

[93] ITU-T Recommendation T.38 (09/2010): "Procedures for real-time Group 3 facsimile communication over IP networks".

[94] IETF RFC 3362 (2002): "Real-time Facsimile (T.38) - image/t38 MIME Sub-type Registration".

[95] IETF RFC 5285 (2008): "A General Mechanism for RTP Header Extensions", D. Singer, H. Desineni.

[96] IETF RFC 5168 (2008): "XML Schema for Media Control", O. Levin, R. Even and P. Hagendorf.

[97] 3GPP2 C.S0055-A, version 1.0: "Packet Switched Video Telephony Service (PSVT/MCS)".

[98] ETSI TS 181 005, v3.3.1: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Service and Capability Requirements".

[99] 3GPP2 C.S0014-E, version 1.0: "Enhanced Variable Rate Codec (EVRC)".

[100] ITU-T Recommendation G.729 (06/2012): "Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)".

[101] ITU-T Recommendation G.729.1 (05/2006): "G.729-based embedded variable bit-rate coder: An 8-32 kbit/s scalable wideband coder bitstream interoperable with G.729".

[102] 3GPP2 C.S0076, version 1.0: "Discontinuous Transmission (DTX) of Speech in cdma2000 Systems".

[103] IETF RFC 5188 (2008):"RTP Payload Format for the Enhanced Variable Rate Wideband Codec (EVRC-WB) and the Media Subtype Updates for EVRC-B Codec".

[104] IETF RFC 4749 (2006): "RTP Payload Format for the G.729.1 Audio Codec".

[105] IETF RFC 5459 (2009): "G.729.1 RTP Payload Format Update: Discontinuous Transmission (DTX) Support".

[106] IETF RFC 4788 (2007): "Enhancements to RTP Payload Formats for EVRC Family Codecs".

[107] IETF RFC 4855 (2007): "Media Type Registration of RTP Payload Formats".

[108] ITU-T Recommendation P.10 (07/2006): "Vocabulary and effects of transmission parameters on customer opinion of transmission quality".

[109] ETSI TS 103 737, v1.1.2: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for narrowband wireless terminals (handset and headset) from a QoS perspective as perceived by the user".

[110] ETSI TS 103 738, v1.1.2: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for narrowband wireless terminals (handsfree) from a QoS perspective as perceived by the user".

[111] ETSI TS 103 739, v1.1.2: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for wideband wireless terminals (handset and headset) from a QoS perspective as perceived by the user".

[112] ETSI TS 103 740, v1.1.2: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for wideband wireless terminals (handsfree) from a QoS perspective as perceived by the user".

[113] ETSI TS 202 737, v1.3.2: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for narrowband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user".

[114] ETSI TS 202 738, v1.3.2: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for narrowband VoIP loudspeaking and handsfree terminals from a QoS perspective as perceived by the user".

[115] ETSI TS 202 739, v1.3.2: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for wideband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user ".

[116] ETSI TS 202 740, v1.3.2: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for wideband VoIP loudspeaking and handsfree terminals from a QoS perspective as perceived by the user ".

[117] ETSI EN 300 175-8, v2.5.1: "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 8: Speech and audio coding and transmission".

[118] ETSI TS 300 176-2, v2.2.1: "Digital Enhanced Cordless Telecommunications (DECT); Test specification; Part 2: Audio and speech".

[119] Recommendation ITU-T H.265: "High efficiency video coding" | ISO/IEC 23008-2:2020: "High Efficiency Coding and Media Delivery in Heterogeneous Environments – Part 2: High Efficiency Video Coding". [120] IETF RFC 7798 (2016): "RTP Payload Format for High Efficiency Video Coding (HEVC)", Y.-K. Wang, Y. Sanchez, T. Schierl, S. Wenger, M. M. Hannuksela.

[121] 3GPP TS 26.441: "Codec for Enhanced Voice Services (EVS); General Overview".

[122] 3GPP TS 26.442: "Codec for Enhanced Voice Services (EVS); ANSI C code (fixed-point)".

[123] 3GPP TS 26.443: "Codec for Enhanced Voice Services (EVS); ANSI C code (floating-point)".

[124] 3GPP TS 26.444: "Codec for Enhanced Voice Services (EVS); Test Sequences".

[125] 3GPP TS 26.445: "Codec for Enhanced Voice Services (EVS); Detailed Algorithmic Description".

[126] 3GPP TS 26.446: "Codec for Enhanced Voice Services (EVS); AMR-WB Backward Compatible Functions".

[127] 3GPP TS 26.447: "Codec for Enhanced Voice Services (EVS); Error Concealment of Lost Packets".

[128] 3GPP TS 26.448: "Codec for Enhanced Voice Services (EVS); Jitter Buffer Management".

[129] 3GPP TS 26.449: "Codec for Enhanced Voice Services (EVS); Comfort Noise Generation (CNG) Aspects".

[130] 3GPP TS 26.450: "Codec for Enhanced Voice Services (EVS); Discontinuous Transmission (DTX)".

[131] 3GPP TS 26.451: "Codec for Enhanced Voice Services (EVS); Voice Activity Detection (VAD)".

[132] 3GPP TS 45.003: "Radio Access Network; Channel coding".

[133] 3GPP TS 23.216: "Single Radio Voice Call Continuity (SRVCC); Stage2".

[134] 3GPP TS 23.237: "IP Multimedia Subsystem (IMS) Service Continuity; Stage2".

[135] ITU-T Recommendation H.224 (01/05): "A real time control protocol for simplex applications using the H.221 LSD/HSD/MLP channels ".

[136] ITU-T Recommendation H.224 (2005): Corrigendum 1 (08/07).

[137] ITU-T Recommendation H.281 (11/94): Transmission of non-telephone signals "A far end camera control protocol for videoconferences using H.224".

[138] ITU-T Recommendation H.323 (12/2009): "Packet-based multimedia communications systems".

[139] IETF RFC 4573 (2006): "MIME Type Registration for RTP Payload Format for H.224".

[140] IETF RFC 4588 (2006): "RTP Retransmission Payload Format", J. Rey, D. Leon, A. Miyazaki, V. Varsa and R. Hakenberg.

[141] IETF RFC 8627 (2019): "RTP Payload Format for Flexible Forward Error Correction (FEC)".

[142] TR 26.922: "Video Telephony Robustness Improvements Extensions (VTRI\_EXT): Performance Evaluation".

[143] IETF RFC 5956 (2010): "Forward Error Correction Grouping Semantics in the Session Description Protocol", A. Cengiz.

[144] 3GPP TR 26.924: "Multimedia telephony over IP Multimedia Subsystem (IMS); Study on improved end-to-end Quality of Service (QoS) handling for Multimedia Telephony Service for IMS (MTSI)".

[145] Void

[146] Void.

[147] 3GPP TS 24.147: "Conferencing Using IP Multimedia Core Network; Stage 3".

[148] IETF RFC 4575 (2006): "A Session Initiation Protocol (SIP) Event Package for Conference State".

[149] IETF RFC 4582 (2006): "The Binary Floor Control Protocol (BFCP)".

[150] IETF RFC 4583 (2006): "Session Description Protocol (SDP) Format for Binary Floor Control (BFCP) Streams".

[151] Void.

[152] 3GPP TR 26.980: "Multimedia telephony over IP Multimedia Subsystem (IMS); Media handling aspects of multi-stream multiparty conferencing for Multimedia Telephony Service for IMS (MTSI)".

[153] IETF RFC 5234 (2008): "Augmented BNF for Syntax Specifications: ABNF", D. Crocker and P. Overell.

[154] IETF RFC 8853 (2021): "Using Simulcast in Session Description Protocol (SDP) and RTP Sessions"

[155] IETF RFC 8851 (2021): "RTP Payload Format Restrictions"

[156] IETF RFC 7728 (2016): "RTP Stream Pause and Resume".

[157] 3GPP TS 36.321: "Evolved Universal Terrestrial Radio Access (E-UTRA); Medium Access Control (MAC) protocol specification".

[158] 3GPP TS 25.331: "Radio Resource Control (RRC); Protocol specification".

[159] "Mobile Location Protocol (MLP)", Open Mobile Alliance, OMA-LIF-MLP-V3\_1, Approved Version 3.1 – 20 Sep 2011.

[160] 3GPP TS 36.331: "Evolved Universal Terrestrial Radio Access (E-UTRA); Radio Resource Control (RRC); Protocol specification".

[161] 3GPP TS 27.007: " Technical Specification Group Core Network and Terminals; AT command set for User Equipment (UE)".

[162] Void

[163] 3GPP TS 38.331: "NR; Radio Resource Control (RRC); Protocol Specification".

[164] 3GPP TS 38.300: "NR; NR and NG-RAN Overall Description; Stage 2".

[165] 3GPP TS 26.452: "Codec for Enhanced Voice Services (EVS); ANSI C code; Alternative fixed-point using updated basic operators".

[166] 3GPP TS 38.321: "NR; Medium Access Control (MAC) protocol specification".

[167] 3GPP TS 23.228: "IP Multimedia Subsystem (IMS); Stage 2".

[168] 3GPP TR 26.952: "Codec for Enhanced Voice Services (EVS); Performance characterization".

[169] 3GPP TR 26.959: "Study on enhanced Voice over LTE (VoLTE) performance".

[170] 3GPP TS 36.323: "Evolved Universal Terrestrial Radio Access (E-UTRA); Packet Data Convergence Protocol (PDCP) specification".

[171] 3GPP TS 37.324: "Evolved Universal Terrestrial Radio Access (E-UTRA) and NR; Service Data Adaptation Protocol (SDAP) specification".

[172] IETF RFC 8864 (2021): "Negotiation Data Channels Using the Session Description Protocol (SDP)"

[173] IETF RFC 4960 (2007): "Stream Control Transmission Protocol"

[174] IETF RFC 8261 (2017): "Datagram Transport Layer Security (DTLS) Encapsulation of SCTP Packets"

[175] IETF RFC 8831 (2021): "WebRTC Data Channels".

[176] 3GPP TS 23.501: "System Architecture for the 5G System; Stage 2".

[177] IETF RFC 5688 (2010): "A Session Initiation Protocol (SIP) Media Feature Tag for MIME Application Subtypes".

[178] 3GPP TS 28.405; "Management of Quality of Experience (QoE) measurement collection; Control and configuration"

[179] ISO/IEC 23090-2:2019: " Information technology -- Coded representation of immersive media -- Part 2: Omnidirectional media format".

[180] 3GPP TS 26.118: "3GPP Virtual reality profiles for streaming applications".

[181] ITU-T Recommendation G.1028 (06/2019): "End-to-end quality of service for voice over 4G mobile networks".

[182] 3GPP TS 24.526: "User Equipment (UE) policies for 5G System (5GS); Stage 3".

[183] ISO/IEC 23090-14: Information technology — Coded representation of immersive media — Part 14: Scene Description for MPEG Media.

[184] IETF RFC 8839 (2021): "Session Description Protocol (SDP) Offer/Answer Procedures for Interactive Connectivity Establishment (ICE)".

[185] IETF RFC 9071(2021): "RTP-Mixer Formatting of Multiparty Real-Time Text".

[186] 3GPP TS 26.250: "Codec for Immersive Voice and Audio Services – General Overview".

[187] 3GPP TS 26.252: "Codec for Immersive Voice and Audio Services – Test Sequences".

[188] 3GPP TS 26.253: "Codec for Immersive Voice and Audio Services – Detailed Algorithmic Description incl. RTP payload format and SDP parameter definitions".

[189] 3GPP TS 26.254: "Codec for Immersive Voice and Audio Services – Rendering".

[190] 3GPP TS 26.255: "Codec for Immersive Voice and Audio Services – Error Concealment of Lost Packets".

[191] 3GPP TS 26.256: "Codec for Immersive Voice and Audio Services – Jitter Buffer Management".

[192] 3GPP TS 26.258: "Codec for Immersive Voice and Audio Services – C code (floating-point)".

[193] 3GPP TS 26.261: "Terminal audio quality performance requirements for immersive audio services".

[194] IETF RFC 8841 (2021): "Session Description Protocol (SDP) Offer/Answer Procedures for Stream Control Transmission Protocol (SCTP) over Datagram Transport Layer Security (DTLS) Transport".

\* \* \* Next Change \* \* \* \*

#### 6.2.10.1 General

Support of data channel media is optional for an MTSI client and an MTSI client in terminal. For brevity, an MTSI client supporting data channel is henceforth denoted as a DCMTSI client or DCMTSI client in terminal, respectively.

To indicate support for the procedures in this clause, a DCMTSI client shall when including media feature tags as specified in TS 24.229 [7] include a +sip.app-subtype media feature tag, as specified by IETF RFC 5688 [177], with a value of "webrtc-datachannel" (the application media format used by IETF RFC 8864 [172]), regardless of data channel media being part of the SDP or not.

One or more data channel SDP media descriptions formatted according to IETF RFC 8864 [172] may be added to the SDP, alongside other SDP media descriptions such as e.g. speech, video, and text. The protocol identifier (proto value) and media format (fmt value) of a data channel SDP media description shall be set to "UDP/DTLS/SCTP" and "webrtc-datachannel" defined in IETF RFC 8841 [194], respectively.

A data channel SDP media description shall not be placed before the first SDP speech media description. SDP examples are provided in Annex A.17.

If data channels are used in a session, the session setup shall determine the applicable bandwidth limit(s) as defined in clause 6.2.5.

Multiple data channels may be mapped to a single data channel SDP media description, each with a corresponding "a=dcmap" SDP attribute and stream IDs that are unique within that media description. There is no limit to the number of data channels in an SDP media description, but the aggregate of all defined data channels shall keep within the set bandwidth limit and care should be taken to avoid excessive SDP size. If the session is re-negotiated to include a changed number of data channels in an SDP media description, the bandwidth limit may either be kept constant, changing the share of bandwidth available to each individual data channel, or the bandwidth limit may be changed to accommodate the changed number of data channels, keeping individual data channel bandwidth shares. Regardless of what approach is used when changing number of used data channels in a media description, the aggregate of all defined data channels shall keep within the re-negotiated bandwidth limit.

If there is a need to use data channels with either different transport IP addresses, different UDP ports, or different SCTP ports, separate data channel SDP media descriptions shall be used, as IP address, UDP port and SCTP port are all constant per SDP media description. Multiple SCTP associations for a single channel, commonly denoted as "multi-homing", defined in IETF RFC 4960 [173] for reasons of redundancy and basically using one destination transport address at a time, is not described for use with WebRTC data channel and shall therefore not be used in this specification.

NOTE 1: The main reasons to not specify multi-homing are because it cannot use the needed separation of signalling paths for redundancy purposes in the applicable usage scenarios, and it is also not considered feasible when using SCTP on top of DTLS.

To ease data channel media implementation and ease interworking with WebRTC data channels, DCMTSI clients shall support ICE Lite and may support full ICE [184], for data channel media. DCMTSI clients supporting full ICE shall only use host candidate addresses. SDP "a=candidate" line host address information shall match corresponding SDP "c=" and "m=" line information.

NOTE 2: In typical IMS deployments, it is expected that DCMTSI clients have no need to use STUN or TURN servers with ICE. This is in line with what constitutes an ICE Lite agent.

A "data channel application" consists of an HTML web page including JavaScript(s), and optionally image(s) and style sheet(s). A "bootstrap data channel" is henceforth defined as a data channel used to retrieve data channel application(s) for a DCMTSI client in terminal, with a data channel stream ID below 1000, and using the HTTP [73] protocol as data channel subprotocol. The data channel application accessible at the HTTP root ("/") URL through a bootstrap data channel describes the graphical user interface and the logic needed to handle any further data channel usage beyond the bootstrap data channel itself. The meaning of the "authority" (host) part of the URL and consequently the "Host" HTTP header are not defined, shall be ignored on reception, and shall be set to the empty value by a DCMTSI client in terminal.

NOTE 3: Data channel stream IDs below 1000 may use a well-defined subprotocol for other features than retrieving data channel application(s). For example, the “mpeg-sd” subprotocol can be used for a data channel stream ID below 1000 for scene description-based overlays as specified in Annex Y.6.9.

The data channel application is created prior to the DCMTSI call where it is intended to be used, by means left out of scope for this specification. The data channel application workflow is depicted by Figure 6.2.10.1-1 below.



Figure 6.2.10.1-1: Data Channel Workflow

NOTE 4: A Data Channel Server in this specification can be further decomposed into a number of functional entities including DC Signalling Function, Media Function (or MRF) and DC Application Server as specified in Annex AC of [167].

The data channel application is, referring to the numbered arrows in Figure 6.2.10.1-1:

1. Uploaded to the network, by the UE user or some other authorized party.

2. Stored in a data channel application repository in the network.

3. During the DCMTSI call where it should be used, retrieved from the repository.

4. Sent through a bootstrap data channel to the local UE A as a response of its request.

5. Sent through a bootstrap data channel to the remote UE B as a response of its request. This may happen in parallel with and rather independent of step 4.

6. Any additional data channels created and used by the data channel application itself are established (logically) between UE A and UE B. Data transmission on data channels shall not start until there is confirmation that both peers have instantiated the data channel, using the same procedures as described for WebRTC in section 6.5 of [172]. The traffic may effectively go through the Data Channel Server, e.g., when the bootstrap and end-to-end data channels have the same anchoring point. This traffic may pass across an inter-operator border if UE A and UE B belong to different operators’ networks.

The bootstrap data channel is not intended for use directly between DCMTSI clients in terminal. DCMTSI clients in terminal that receive HTTP requests on a bootstrap data channel shall ignore such request and shall update the session by removing the SDP "a=dcmap" line with the stream ID where such HTTP request was received, and closing that stream ID.

The data channel application including its resources retrieved via a bootstrap data channel may be updated at any time, automatically or interactively, using normal HTTP procedures over the bootstrap data channel.

A bootstrap data channel shall be configured as ordered, reliable, with normal SCTP multiplexing priority. The sub-protocol for a bootstrap data channel shall be HTTP (not encapsulating HTTP in TCP), represented by the following, example SDP "a=dcmap" line, which therefore shall be present in each data channel media description in an SDP offer from a DCMTSI client in terminal:

a=dcmap:0 subprotocol="http"

Any other data channels used by the data channel application JavaScript(s) sent in the bootstrap data channel shall be represented in an updated SDP as additional "a=dcmap" lines with stream ID values starting from 1000, using stream ID numbers from the JavaScript(s).

There are multiple, possible providers of data channel applications. In Figure 6.2.10.1-1, assume that UE A is local to the operator hosting the data channel server. Further assume that UE B belongs to a different operator (remote). The user of UE A can create and use data channel applications (steps 1-4), which can also be sent to UE B (step 5). Similarly, some other authorized part associated with UE A’s operator can create data channel applications for use by UE A (steps 1-4), which can also be sent to UE B (step 5). For simplicity, there’s no data channel server and data channel application repository depicted for UE B in Figure 6.2.10.1-1, but those could be present in a more general case. Seen from the perspective of a single UE, there are then at least four possible data channel application providers:

1. The local UE user.

2. Other authorized parties associated with the local network (e.g. the local operator).

3. The remote UE user.

4. Other authorized parties associated with the remote network (e.g. the remote operator).

The HTML web content making up a data channel application in each bootstrap data channel represents a different context of user interaction and should open in a separate tab, or some corresponding user interface construct, but the details are out of scope for this specification and left open for individual implementations. It shall be possible to use and navigate between different data channel applications from different bootstrap data channels with different stream IDs that are open simultaneously.

Table 6.2.10.1-2 describes a mandatory mapping between stream ID and bootstrap channel data channel application content sources, as seen from a single (local) DCMTSI client in terminal, each of which shall be listed as separate "a=dcmap" lines with "http" subprotocol in SDP when the DCMTSI client in terminal supports receiving data channel application content from that source.

Table 6.2.10.1-2: Bootstrap Data Channel Content Sources

|  |  |
| --- | --- |
| **Stream ID** | **Content Source** |
| 0 | Local network provider |
| 10 | Local user |
| 100 | Remote network provider |
| 110 | Remote user |

NOTE 5: When the local user has defined and stored multiple, different data channel applications in the local data channel application repository, the local network provider may provide functionality in the stream ID 0 data channel application that enables a dynamic choice of which user-defined data channel application to use with stream ID 10 in the DCMTSI call.

NOTE 6: To help the SDP answerer's network to distinguish the two media descriptions (m= lines) containing bootstrap data channels with the same stream ID values transferred between two networks, the SDP offerer's network adds an "a=3gpp-bdc-used-by:sender" attribute in the media description of the bootstrap data channel(s) established between the originating UE and the terminating network, and optionally adds "a=3gpp-bdc-used-by:receiver" attribute in the media description of the bootstrap data channel(s) established between the originating network and the terminating UE, before it sends the SDP offer to the remote network.

Figure 6.2.10.1-3, referring to Figure 6.2.10.1-1 and Table 6.2.10.1-2, is depicting the stream IDs used for distribution of a data channel application owned by UE A from its local data channel repository to both UE A (stream ID 10) and its remote UE B (stream ID 110).



Figure 6.2.10.1-3: Distribution of local data channel application to both UE

When the user in UE A in a call with UE B selects data channel application(s) for retrieval and use, and after the new application(s) are launched, the application(s) may make use of additional data channel(s) (see step 6 of 6.2.10.1-1). In this case, UE A initiates a call upgrade to add new data channel(s) to the call for the new application(s). The SDP offer the UE A generates shall include an "a=3gpp-req-app" attribute with a "req-app-id" parameter, as defined by clause 6.2.13, to identify the requesting application as part of the media description creating application data channels for that application. The application should be configured with that identification and the network deployment should ensure that identification to be sufficiently unique to avoid ambiguity. The "a=3gpp-req-app" attribute may also include an "app-dc-info" parameter to allow the application to identify a different end point when creating multiple application data channels used for communication to a network server or to the remote UE.

The combination of "req-app-id" and "app-dc-info" parameters allows the communicating UEs to bind the SDP offers and answers for each data channel and stream IDs being negotiated for the respective applications using these data channel stream IDs.

\* \* \* End of Changes \* \* \* \*