**3GPP TSG-SA4 Meeting #121 *S4-221558***

**Toulouse, France, Nov 14 – 18, 2022**

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| *CR-Form-v12.2* | | | | | | | | |
| **CHANGE REQUEST** | | | | | | | | |
|  | | | | | | | | |
|  | **26.114** | **CR** | **0535** | **rev** | **1** | **Current version:** | **16.11.0** |  |
|  | | | | | | | | |
| *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)*.* | | | | | | | | |
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| ***Proposed change affects:*** | UICC apps |  | ME | **X** | Radio Access Network |  | Core Network | **X** |

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|  | | | | | | | | | | |
| ***Title:*** | Use of ICE with IMS Data Channel | | | | | | | | | |
|  |  | | | | | | | | | |
| ***Source to WG:*** | Ericsson LM | | | | | | | | | |
| ***Source to TSG:*** | S4 | | | | | | | | | |
|  |  | | | | | | | | | |
| ***Work item code:*** | 5G\_MEDIA\_MTSI\_ext | | | | |  | ***Date:*** | | | 2022-11-16 |
|  |  | | | |  | |  | | |  |
| ***Category:*** | **F** |  | | | | | ***Release:*** | | | Rel-16 |
|  | *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-16 (Release 16) Rel-17 (Release 17) Rel-18 (Release 18) Rel-19 (Release 19)* | |
|  |  | | | | | | | | | |
| ***Reason for change:*** | | Existing IMS Data Channel specification text is silent on the use of ICE. While ICE should strictly not be needed in an IMS network, one of the main, intended benefits with IMS Data Channel is the possibility to reuse WebRTC data channel technology, and most, if not all, WebRTC implementations rely heavily on the use of ICE and cannot completely disable it. The inability to disable ICE in WebRTC implementation has been confirmed by W3C. The last SDP offer example in Annex A.17 incorrectly doesn’t use the actpass DTLS role, which is required by RFC 8842. The last SDP offer example in Annex A.17 doesn’t use a separate m= line for application (end-to-end) data channel media, which is inconsistent with clause 6.2.10.1. | | | | | | | | |
|  | |  | | | | | | | | |
| ***Summary of change:*** | | Adding text on use of ICE and ICE Lite with IMS Data Channel. Use of ICE Lite has minor network and UE impact and all WebRTC implementations are required to at least interoperate with ICE Lite. Support of ICE Lite is therefore mandated, while support of full ICE is optional but allowed. Correcting the a=setup line in last SDP offer example of Annex A.17 to use actpass value. Changing the application (end-to-end) data channel into using a separate m= line. | | | | | | | | |
|  | |  | | | | | | | | |
| ***Consequences if not approved:*** | | IMS Data Channel user plane will be incompatible with WebRTC data channel user plane, eliminating the intended technology synergy and losing all benefits coming from reuse of existing WebRTC technology. Difficult user plane implementation, not being able to reuse existing WebRTC libraries. An incorrect example in Annex A.17 may cause confusion or incorrect implementation. | | | | | | | | |
|  | |  | | | | | | | | |
| ***Clauses affected:*** | | 2, 6.2.1, 6.2.10.1, 6.2.10.2, 6.2.10.3, Annex A.17 | | | | | | | | |
|  | |  | | | | | | | | |
|  | | **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:*** | | S4-221428 Initial version  S4-221558 Clarifying session-level in SDP examples | | | | | | | | |

**\*\*\*\* 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*.

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[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".

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[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".

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[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.

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[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".

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[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.

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[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".

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[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".

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

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**\*\*\*\* 2nd change \*\*\*\***

6.2 Session setup procedures

6.2.1 General

The session setup for RTP transported media shall determine for each media: IP address(es), RTP profile, UDP port number(s); codec(s); RTP Payload Type number(s), RTP Payload Format(s). The session setup may also determine: ECN usage and any additional session parameters.

The session setup for UDP transported media without RTP shall determine: IP address(es), UDP port number(s) and additional session parameters.

The session setup for data channel (SCTP over DTLS over UDP transported) media shall determine for each media: IP address(es), UDP port number(s), SCTP port number(s), DTLS server/client role(s), DTLS ID(s), DTLS certificate fingerprint(s), and ICE-related information for data channel media as described in clause 6.2.10. The session setup may also determine use of ICE Lite for data channel media and may determine additional session parameters.

The session setup for RTP and data channel transported media shall, when the port number is not set to zero, determine the maximum bandwidth that is allowed in the session, see also clause 6.2.5. The maximum bandwidth for the receiving direction is specified with the "b=AS" bandwidth modifier. Additional requirements and/or recommendations on the bandwidth negotiation are found in clause 6.2.2.1 for speech, in clause 6.2.3.2 for video, and in clause 6.2.10 for data channel.

An MTSI client shall offer at least one RTP profile for each RTP media stream. Multiple RTP profiles may be offered using SDPCapNeg as described in Clause 6.2.1a. For voice and real-time text, the first SDP offer shall include at least the AVP profile. For video, the first SDP offer for a media type shall include at least the AVPF profile. Subsequent SDP offers may include only other RTP profiles if it is known from a preceding offer that this RTP profile is supported by the answerer. The MTSI client shall be capable of receiving an SDP offer containing both AVP and AVPF offers in order to support interworking.

The configuration of ECN for media transported with RTP is described in clause 6.2.2 for speech and in clause 6.2.3.2 for video. The negotiation of ECN at session setup is described in [84]. The adaptation response to congestion events is described in clause 10.

**\*\*\*\* 3rd change \*\*\*\***

6.2.10 Data channel

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 RFC 5688 [177], with a value of "webrtc-datachannel" (the application media format used by [172]), regardless of data channel media being part of the SDP or not.

One or more data channel SDP media descriptions formatted according to [172] may be added to the SDP, alongside other SDP media descriptions such as e.g. speech, video, and text. A data channel SDP media description must 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 must 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 bandwith 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 must 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 must 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 must 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 must support ICE Lite and may support full ICE [x1], for data channel media. DCMTSI clients supporting full ICE must only use host candidate addresses. SDP "a=candidate" line host address information must 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.

Data channel stream IDs below 1000 must be reserved for using the HTTP [73] protocol, henceforth denoted as "bootstrap data channels", to retrieve an HTML web page including JavaScript(s), and optionally image(s) and style sheet(s), henceforth denoted as a "data channel application". 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.

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**

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.

5. Sent through a bootstrap data channel to the remote UE B. 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 sent in a bootstrap data channel may be updated at any time, automatically or interactively, using normal HTTP procedures.

A bootstrap data channel must be configured as ordered, reliable, with normal SCTP multiplexing priority, and using HTTP as subprotocol (not encapsulating HTTP in TCP), represented by the following, example SDP "a=dcmap" line, which therefore must 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 must 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 must 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 3: 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.

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**

6.2.10.2 Generating SDP offer

A DCMTSI client in terminal may include a data channel media description for the "bootstrap" data channels in the initial SDP offer, as described above and according to [172] [x1]. A DCMTSI client in terminal may add or disable (by setting port 0, as for RTP media) additional data channel media descriptions as needed in subsequent SDP offers.

A DCMTSI client in terminal that desires to use data channels with stream IDs from a data channel application retrieved from its local "bootstrap" data channel stream ID 0 or 10, shall initiate a subsequent SDP offer after the initial SDP offer, opening those data channels by adding corresponding "a=dcmap" and (optionally) "a=dcsa" lines. A DCMTSI client in terminal that retrieves a data channel application from a stream ID different than 0 or 10 (e.g. a data channel application from the peer), shall not initiate any subsequent offer to open data channels used by that data channel application.

A data channel media description with specific loss or latency requirements should use "a=3gpp-qos-hint" in the SDP offer, as detailed in section 6.2.7.4. If subsequent SDP offers or answers adds data channels with more strict loss or latency requirements that cannot be met by keeping current "a=3gpp-qos-hint" and providing suitable SCTP "a=dcmap" parameters, the existing "a=3gpp-qos-hint" should be modified accordingly. Similarly, if subsequent SDP offers or answers closes (removes) data channels that are known to be the limiting factor for choosing the existing "a=3gpp-qos-hint", a more relaxed "a=3gpp-qos-hint" should be chosen to better fit the remaining data channels.

6.2.10.3 Generating SDP answer

An answering DCMTSI client in terminal may accept an SDP offer with data channel as described by [172] [x1].

An answering DCMTSI client in terminal that desires to reject the entire SCTP association for all offered data channels shall set the port to 0 (zero) on the corresponding "m=application" line in SDP, as described in [172]. An SCTP association that initially, or as a result of session modification, has no open data channels ("a=dcmap" lines) should be rejected or closed by modifying the session, setting port number to 0 (zero).

An answering DCMTSI client in terminal that desires to accept some offered data channels and reject others shall indicate this by removing the non-desired data channel "a=dcmap" and "a=dcsa" lines from the SDP answer, as described in [172]. The DCMTSI client in terminal accepting a data channel must also accept the corresponding, supported "bootstrap" data channels with stream ID <1000 (e.g. a=dcmap:0 …).

6.2.10.4 Receiving SDP answer

An offering DCMTSI client in terminal receiving an SDP answer where the data channel SCTP association is accepted (port is not 0) may use any offered stream ID that has a corresponding "a=dcmap" line in the SDP answer, as described by section 6.5 in [172]. Data channels with "a=dcmap" lines in the SDP offer that are not included in the SDP answer must be considered as rejected and shall not be used, as described by section 6.5 in [172].

**\*\*\*\* 4th change \*\*\*\***

A.17 SDP offers and answers with data channel capability signaling

The ellipsis ("...") in the examples in this clause is not part of the SDP but indicates possible presence of other media descriptions in addition to the ones shown in the examples.

Table A.17.1 demonstrates an example SDP offer with data channel capability signalling for the "bootstrap" data channel defined in clause 6.2.10. The offering part is an ICE Lite agent, indicated by "a=ice-lite" on SDP session level (i.e., before first m= line), and thus only offers host candidates, in this example a single host candidate aligned with address information on the corresponding m= and c= lines.

**Table A.17.1: Example SDP offer with data channel capability signalling**

|  |
| --- |
| **SDP offer** |
| a=ice-options:ice2  a=ice-lite ...  m=application 52718 UDP/DTLS/SCTP webrtc-datachannel c=IN IP4 192.0.2.156b=AS:500  a=candidate:1 1 UDP 2130706431 192.0.2.156 52718 typ host a=ice-ufrag:8hhY  a=ice-pwd:asd88fgpdd777uzjYhagZga=max-message-size:1024  a=sctp-port:5000  a=setup:actpass  a=fingerprint:SHA-1 4A:AD:B9:B1:3F:82:18:3B:54:02:12:DF:3E:5D:49:6B:19:E5:7C:AB  a=tls-id: abc3de65cddef001be82  a=dcmap:0 subprotocol="http" |

An example SDP answer is shown in Table A.17.2, where the data channel capability signalling from Table A.17.1 is also supported and accepted by the answerer, as indicated by the non-zero port on the m= line. The answering part is an ICE Lite agent, indicated by "a=ice-lite" on SDP session level, and only supports ICE according to the predecessor ICE specification to [x1] as indicated by no "a=ice-options:ice2" being included on SDP session level..

**Table A.17.2: Example SDP answer with data channel capability**

|  |
| --- |
| **SDP answer** |
| a=ice-lite  ...  m=application 52718 UDP/DTLS/SCTP webrtc-datachannel c=IN IP4 192.0.2.1  b=AS:500 a=candidate:1 1 UDP 2130706431 192.0.2.1 52718 typ host a=ice-ufrag:9uB6  a=ice-pwd:YH75Fviy6338Vbrhrlp8Yh a=max-message-size:1024  a=sctp-port:5002  a=setup:passive  a=fingerprint:SHA-1 5B:AD:67:B1:3E:82:AC:3B:90:02:B1:DF:12:5D:CA:6B:3F:E5:54:FA  a=tls-id: dcb3ae65cddef0532d42  a=dcmap:0 subprotocol="http" |

Table A.17.3 demonstrates an example SDP offer with multiple possible data channel application sources for the "bootstrap" data channel defined in Table 6.2.10.1-2. In this example, the offering part supports full ICE, indicated by no "a=ice-lite" on SDP session level.

**Table A.17.3: Example SDP offer with multiple data channel application sources**

|  |
| --- |
| **SDP offer** |
| a=ice-options:ice2 ...  m=application 52718 UDP/DTLS/SCTP webrtc-datachannel  c=IN IP6 fe80::6676:baff:fe9c:ee4a b=AS:500  a=candidate:1 1 UDP 2130706431 fe80::6676:baff:fe9c:ee4a 52718 typ host  a=ice-ufrag:8hhY  a=ice-pwd:asd88fgpdd777uzjYhagZg a=max-message-size:1024  a=sctp-port:5000  a=setup:actpass  a=fingerprint:SHA-1 4A:AD:B9:B1:3F:82:18:3B:54:02:12:DF:3E:5D:49:6B:19:E5:7C:AB  a=tls-id: abc3de65cddef001be82  a=dcmap:0 subprotocol="http"  a=dcmap:10 subprotocol="http"  a=dcmap:100 subprotocol="http"  a=dcmap:110 subprotocol="http" |

An example SDP answer is shown in Table A.17.4, where only one of the the data channel application sources from the offer in Table A.17.3 is accepted by the answerer, removing the other a=dcmap lines.

Figure 6.2.10.1-3 in clause 6.2.10.1 may be used as illustration to this example, in which case UE A in that Figure would send the offer in Table A.17.3, and UE B would send the answer in Table A.17.4.

In this SDP answer, the answerer (UE B) only accepts stream ID 110 to receive the data channel application from the offerer (UE A), but UE B has rejected to use any other data channel application provider.

**Table A.17.4: Example UE SDP answer choosing a single data channel application source**

|  |
| --- |
| **SDP answer** |
| a=ice-options:ice2 a=ice-lite  ...  m=application 52718 UDP/DTLS/SCTP webrtc-datachannel  c=IN IP4 192.0.2.1  b=AS:500  a=candidate:1 1 UDP 2130706431 192.0.2.1 52718 typ host a=ice-ufrag:9uB6  a=ice-pwd:YH75Fviy6338Vbrhrlp8Yh a=max-message-size:1024  a=sctp-port:5002  a=setup:passive  a=fingerprint:SHA-1 5B:AD:67:B1:3E:82:AC:3B:90:02:B1:DF:12:5D:CA:6B:3F:E5:54:FA  a=tls-id: dcb3ae65cddef0532d42  a=dcmap:110 subprotocol="http" |

Figure 6.2.10.1-3 in clause 6.2.10.1 may be used as illustration also to the example in Table A.17.5, in which case UE A in Figure 6.2.10.1-3 would send the offer in Table A.17.3, and the SDP answer sent back to UE A from the network would be the one in Table A.17.5.

In the SDP answer in Table A.17.5 sent from UE A’s (local) network, it is accepting stream ID 10 that would be used by UE A to receive its own, chosen data channel application, corresponding to the data channel application sent to UE B in stream ID 110 based on the SDP answer in Table A.17.4 such that both UEs can use the same application. That application is however received through different stream IDs for UE A and UE B, as shown in Figure 6.2.10.1-3.

**Table A.17.5: Example network SDP answer choosing a single data channel application source**

|  |
| --- |
| **SDP answer** |
| a=ice-options:ice2 a=ice-lite  ...  m=application 52718 UDP/DTLS/SCTP webrtc-datachannel  c=IN IP4 192.0.2.1  b=AS:500  a=candidate:1 1 UDP 2130706431 192.0.2.1 52718 typ host a=ice-ufrag:9uB6  a=ice-pwd:YH75Fviy6338Vbrhrlp8Yh a=max-message-size:1024  a=sctp-port:5010  a=setup:active  a=fingerprint:SHA-1 BC:8A:99:A0:E3:28:CA:B3:09:20:1B:FD:21:D5:AC:B6:F3:5E:45:AF  a=tls-id: cd3bea56dced0f35d224  a=dcmap:10 subprotocol="http" |

Table A.17.6 demonstrates an example SDP (re-)offer that adds two non-bootstrap data channel streams used by the data channel application in the bootstrap data channel in Table A.17.5. The data channel application streams (two in this example) desire specific loss and latency characteristics indicated by the "a=3gpp-qos-hint" line (see also Annex A.16) and are offered as a separate m= line due to having different QoS requirements and different destination (e.g. a peer UE) than the bootstrap data channel. The stream with ID 38754 has a strict latency requirement and data older than 150 ms will not be transmitted or re-transmitted. The stream with ID 7216 requires lower loss but can accept somewhat higher latency than stream ID 38754 and therefore allows at most 5 SCTP-level retransmissions.

**Table A.17.6: Example SDP offer with data channel application streams**

|  |
| --- |
| **SDP offer** |
| c=IN IP4 192.0.2.156 a=ice-options:ice2  a=ice-lite  ...  m=application 52718 UDP/DTLS/SCTP webrtc-datachannel  b=AS:500  a=candidate:1 1 UDP 2130706431 192.0.2.156 52718 typ host a=ice-ufrag:8hhY  a=ice-pwd:asd88fgpdd777uzjYhagZg a=max-message-size:1024  a=sctp-port:5000  a=setup:actpass  a=fingerprint:SHA-1 4A:AD:B9:B1:3F:82:18:3B:54:02:12:DF:3E:5D:49:6B:19:E5:7C:AB  a=tls-id: abc3de65cddef001be82  a=dcmap:10 subprotocol="http"  m=application 52720 UDP/DTLS/SCTP webrtc-datachannel  b=AS:1000  a=candidate:1 1 UDP 2130706431 192.0.2.156 52720 typ host a=ice-ufrag:9uB6  a=ice-pwd: YH75Fviy6338Vbrhrlp8Yh a=max-message-size:1024  a=sctp-port:5000  a=setup: actpass  a=fingerprint:SHA-1 BC:8A:99:A0:E3:28:CA:B3:09:20:1B:FD:21:D5:AC:B6:F3:5E:45:AF  a=tls-id: cd3bea56dced0f35d224  a=dcmap:38754 max-time=150;label="low latency"  a=dcmap:7216 max-retr=5;label="low loss"  a=3gpp-qos-hint:loss=0.01;latency=100 |

**\*\*\*\* End of changes \*\*\*\***