**3GPP TSG SA WG4#109-e meeting *S4-200971***

**20th May – 3rd June 2020**

|  |  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- | --- |
| *CR-Form-v12.0* | | | | | | | | |
| **Draft CHANGE REQUEST** | | | | | | | | |
|  | | | | | | | | |
|  | **26.234** | **CR** | **0230** | **rev** | **-** | **Current version:** | **15.1.0** |  |
|  | | | | | | | | |
| *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 | **X** |

|  |  |  |  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- |
|  | | | | | | | | | | |
| ***Title:*** | Removing H.263 and MPEG-4 Visual from PSS | | | | | | | | | |
|  |  | | | | | | | | | |
| ***Source to WG:*** | Qualcomm Incorporated | | | | | | | | | |
| ***Source to TSG:*** | SA4 | | | | | | | | | |
|  |  | | | | | | | | | |
| ***Work item code:*** | RM\_H263\_MP4V | | | | |  | ***Date:*** | | | 2020-06-11 |
|  |  | | | |  | |  | | |  |
| ***Category:*** | **C** |  | | | | | ***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-12 (Release 12)* *Rel-13 (Release 13) Rel-14 (Release 14) Rel-15 (Release 15) Rel-16 (Release 16)* | |
|  |  | | | | | | | | | |
| ***Reason for change:*** | | H.263 was a state-of-the art codec in the last millennium and made mobile video possible and an actual reality. Many 3GPP specs adopted H.263 and H.263 was the format of choice for the first mobile video deployments.  However, more than 20 years later, this format has done its duty and 3GPP should feel good about sending this codec to retirement as part of their Rel-16 specs. Actually, several specifications already removed any status around H.263 from their specifications, but have some leftover H.263 related statements.  Why is it relevant to retire older codecs? Supporting codecs on hardware is a significant amount effort and cost, including area size, design and testing. Even if the codec is supported in SW only (which may well be ok for H.263), it still requires a significant amount of unnecessary and costly testing efforts. Supporting such codecs on newly shipping 5G device will just reduce space for new codecs and technologies to be potentially added. One important reason is, that despite on Android there is SW codec for these formats, there are more and more devices such as watches which which do not use Android and hence would require custom H.263 integration.  The same applies for MPEG-4 Visual. | | | | | | | | |
|  | |  | | | | | | | | |
| ***Summary of change:*** | | Remove recommendation and any leftovers for H.263 and MPEG-4 Visual | | | | | | | | |
|  | |  | | | | | | | | |
| ***Consequences if not approved:*** | | Unnecessary costs for testing and implementation | | | | | | | | |
|  | |  | | | | | | | | |
| ***Clauses affected:*** | | 2, 5.2.3.2.2, 5.2.3.2.3, 5.3.2.3.3.2, 5.3.2.4, 5.3.3.2, 5.4, 6.2.3.2, 6.2.4, 7.4.1, 11.2.9.1, A.1, A.3.2.1, A.4.7, Annex F, G.2, L.2.2.1 | | | | | | | | |
|  | |  | | | | | | | | |
|  | | **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 is not yet sent for approval. The reason is that 5GMS3 may deprecate TS 26.234 entirely. In this case we do not want that this work item creates a Rel-16 specification. If 5GMS3 does not retire TS26.234, we would send the formal CR to the next SA plenary (this is indicated in the exception sheet). | | | | | | | | |
|  | |  | | | | | | | | |
| ***This CR's revision history:*** | |  | | | | | | | | |

**===== 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 TS 22.233: "Transparent End-to-End Packet-switched Streaming Service; Stage 1".

[2] 3GPP TS 26.233: "Transparent end-to-end packet switched streaming service (PSS); General description".

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

[4] (void)

[5] IETF RFC 2326: "Real Time Streaming Protocol (RTSP)", Schulzrinne H., Rao A. and Lanphier R., April 1998.

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

[7] IETF STD 0006: "User Datagram Protocol", Postel J., August 1980.

[8] IETF STD 0007: "Transmission Control Protocol", Postel J., September 1981.

[9] IETF RFC 3550: "RTP: A Transport Protocol for Real-Time Applications", Schulzrinne H. et al., July 2003.

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

[11] IETF RFC 4867: "RTP Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs", Sjoberg J. et al., April 2007.

[12] (void)

[13] (void).

[14] (void).

[15] IETF RFC 2046: "Multipurpose Internet Mail Extensions (MIME) Part Two: Media Types", Freed N. and Borenstein N., November 1996.

[16] IETF RFC 3236: "The 'application/xhtml+xml' Media Type", Baker M. and Stark P., January 2002.

[17] IETF RFC 2616: "Hypertext Transfer Protocol – HTTP/1.1", Fielding R. et al., June 1999.

[18] 3GPP TS 26.071: "Mandatory Speech CODEC speech processing functions; AMR Speech CODEC; General description".

[19] (void)

[20] 3GPP TS 26.171: "AMR Wideband Speech Codec; General Description".

[21] ISO/IEC 14496-3:2005: "Information technology – Coding of audio-visual objects – Part 3: Audio".

[22] (void).

[23] (void)

[24] ISO/IEC 14496-2:2004: "Information technology – Coding of audio-visual objects – Part 2: Visual".

[25] (void)

[26] ITU-T Recommendation T.81 (1992) | ISO/IEC 10918-1:1993: "Information technology – Digital compression and coding of continuous-tone still images – Requirements and guidelines".

[27] C-Cube Microsystems: "JPEG File Interchange Format", Version 1.02, September 1, 1992.

[28] (void)

[29] ISO/IEC 10646-1:2000: "Information technology – Universal Multiple-Octet Coded Character Set (UCS) – Part 1: Architecture and Basic Multilingual Plane".

[30] IETF RFC 3629: "UTF-8, a transformation format of ISO 10646", F. Yergeau, November 2003.

[31] (void)

[32] CompuServe Incorporated: "GIF Graphics Interchange Format: A Standard defining a mechanism for the storage and transmission of raster-based graphics information", Columbus, OH, USA, 1987.

[33] CompuServe Incorporated: "Graphics Interchange Format: Version 89a", Columbus, OH, USA, 1990.

[34] (void)

[35] (void)

[36] (void)

[37] (void)

[38] IETF RFC 2083: "PNG (Portable Networks Graphics) Specification Version 1.0", Boutell T., et al., March 1997.

[39] W3C Recommendation: "Composite Capability/Preference Profiles (CC/PP): Structure and Vocabularies 1.0", <http://www.w3.org/TR/2004/REC-CCPP-struct-vocab-20040115/>, January 2004.

[40] Open Mobile Alliance: "User Agent Profile Version 2.0", February 2006.

[41] W3C Recommendation: "RDF Vocabulary Description Language 1.0: RDF Schema", <http://www.w3.org/TR/2004/REC-rdf-schema-20040210/>, February 2004.

[42] W3C Last Call Working Draft: "Scalable Vector Graphics (SVG) 1.2", <http://www.w3.org/TR/2004/WD-SVG12-20041027/>, October 2004.

[43] W3C Last Call Working Draft: "Mobile SVG Profile: SVG Tiny, Version 1.2", <http://www.w3.org/TR/2004/WD-SVGMobile12-20040813/>, August 2004.

[44] Scalable Polyphony MIDI Specification Version 1.0, RP-34, MIDI Manufacturers Association, Los Angeles, CA, February 2002.

[45] Scalable Polyphony MIDI Device 5-to-24 Note Profile for 3GPP Version 1.0, RP-35, MIDI Manufacturers Association, Los Angeles, CA, February 2002.

[46] "Standard MIDI Files 1.0", RP-001, in "The Complete MIDI 1.0 Detailed Specification, Document Version 96.1", The MIDI Manufacturers Association, Los Angeles, CA, USA, February 1996.

[47] WAP Forum Specification: "XHTML Mobile Profile", <http://www1.wapforum.org/tech/terms.asp?doc=WAP-277-XHTMLMP-20011029-a.pdf>, October 2001.

[48] (void)

[49] (void)

[50] 3GPP TS 26.244: "Transparent end-to-end packet switched streaming service (PSS); 3GPP file format (3GP)".

[51] 3GPP TS 26.245: "Transparent end-to-end packet switched streaming service (PSS); Timed text format".

[52] 3GPP TS 26.246: "Transparent end-to-end packet switched streaming service (PSS); 3GPP SMIL Language Profile".

[53] IETF RFC 4234: "Augmented BNF for Syntax Specifications: ABNF", Crocker D. and Overell P., October 2005.

[54] IETF RFC 3066: "Tags for Identification of Languages", Alvestrand H., January 2001.

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

[56] 3GPP TS 23.107: "Quality of Service (QoS) concept and architecture".

[57] IETF RFC 4585: "Extended RTP Profile for Real-time Transport Control Protocol (RTCP)-Based Feedback (RTP/AVPF)", Ott J. et al., July 2006.

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

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

[60] IETF RFC 3986: "Uniform Resource Identifiers (URI): Generic Syntax", Berners-Lee T., Fielding R. and Masinter L., January 2005.

[61] (void)

[62] (void)

[63] 3GPP TS 26.090: "Mandatory Speech Codec speech processing functions; Adaptive Multi-Rate (AMR) speech codec; Transcoding functions".

[64] 3GPP TS 26.073: "ANSI-C code for the Adaptive Multi Rate (AMR) speech codec".

[65] 3GPP TS 26.104: "ANSI-C code for the floating-point Adaptive Multi Rate (AMR) speech codec".

[66] 3GPP TS 26.190: "Speech Codec speech processing functions; AMR Wideband speech codec; Transcoding functions".

[67] 3GPP TS 26.173: "ANCI-C code for the Adaptive Multi Rate - Wideband (AMR-WB) speech codec".

[68] 3GPP TS 26.204: "ANSI-C code for the Floating-point Adaptive Multi-Rate Wideband (AMR-WB) speech codec".

[69] IETF RFC 4648: "The Base16, Base32, and Base64 Data Encodings", Josefsson S., October 2006.

[70] Mobile DLS, MMA specification v1.0. RP-41 Los Angeles, CA, USA. 2004.

[71] Mobile XMF Content Format Specification, MMA specification v1.0., RP-42, Los Angeles, CA, USA. 2004.

[72] IETF RFC 3711: "The Secure Real-time Transport Protocol (SRTP)", Baugher M. et al, March 2004.

[73] Bellovin, S., "Problem Areas for the IP Security Protocols" in Proceedings of the Sixth Usenix Unix Security Symposium, pp. 1-16, San Jose, CA, July 1996

[74] Open Mobile Alliance: "DRM Specification 2.0".

[75] Open Mobile Alliance: "DRM Content Format V 2.0".

[76] IETF RFC 3675: "IPv6 Jumbograms", Borman D., Deering S. and Hinden R., August 1999.

[77] NIST, "Advanced Encryption Standard (AES)", FIPS PUB 197, <http://www.nist.gov/aes/>.

[78] IETF RFC 3394: "Advanced Encryption Standard (AES) Key Wrap Algorithm", Schaad J. and Housley R., September 2002.

[79] IETF RFC 3839: "MIME Type Registrations for 3rd Generation Partnership Project (3GPP) Multimedia files", Castagno R. and Singer D., July 2004.

[80] IETF RFC 4396: "RTP Payload Format for 3rd Generation Partnership Project (3GPP) Timed Text", Rey J. and Matsui Y., February 2006.

[81] IETF RFC 4588: "RTP Retransmission Payload Format", Rey J. et al., July 2006.

[82] 3GPP TS 26.290: "Extended AMR Wideband codec; Transcoding functions".

[83] 3GPP TS 26.304: "ANSI-C code for the Floating-point; Extended AMR Wideband codec".

[84] 3GPP TS 26.273: "ANSI-C code for the Fixed-point; Extended AMR Wideband codec".

[85] IETF RFC 4352: "RTP Payload Format for the Extended Adaptive Multi-Rate Wideband (AMR-WB+) Audio Codec", Sjoberg J. et al., January 2006.

[86] 3GPP TS 26.401: "General audio codec audio processing functions; Enhanced aacPlus general audio codec; General description".

[87] 3GPP TS 26.410: "General audio codec audio processing functions; Enhanced aacPlus general audio codec; Floating-point ANSI-C code".

[88] 3GPP TS 26.411: "General audio codec audio processing functions; Enhanced aacPlus general audio codec; Fixed-point ANSI-C code".

[89] (void)

[90] ITU-T Recommendation H.264 (04/2013): "Advanced video coding for generic audiovisual services".

[91] (void)

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

[93] IETF RFC 3890: "A Transport Independent Bandwidth Modifier for the Session Description Protocol (SDP)", Westerlund M., September 2004.

[94] Standard ECMA-327: "ECMAScript 3rd Edition Compact Profile", June 2001.

[95] 3GPP TR [26.936](http://www.3gpp.org/ftp/Specs/html-info/26936.htm): "Performance characterization of 3GPP audio codecs".

[96] IETF RFC 4288: "Media Type Specifications and Registration Procedures", Freed N. and Klensin J., December 2005.

[97] IETF RFC 4613: "Media Type Registrations for Downloadable Sounds for Musical Instrument Digital Interface (MIDI)", Frojdh P., Lindgren U. and Westerlund M., September 2006.

[98] (void)

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

[100] IETF RFC 1123: "Requirements for Internet Hosts -- Application and Support", Braden R., October 1989.

[101] Internet Streaming Media Alliance (ISMA), ISMA Encryption and Authentication, Version 1.1 release version, September 2006.

[102] Internet Streaming Media Alliance (ISMA), ISMA Encryption and Authentication, Version 2.0 release version, November 2007.

[103] Open Mobile Alliance, Service and Content Protection for Mobile Broadcast Services, Approved Version 1.0, February 2009.

[104] (void)

[105] (void)

[106] (void)

[107] (void)

[108] (void)

[109] 3GPP TS 26.430: "Timed Graphics".

[110] 3GPP TS 23.003 "Numbering, addressing and identification".

[111] 3GPP TS 33.310 "Network Domain Security (NDS); Authentication Framework (AF)".

[112] 3GPP TS 26.247: "Transparent end-to-end Packet-switched Streaming Service (PSS); Progressive Download and Dynamic Adaptive Streaming over HTTP (3GP-DASH)".

[113] 3GPP TR 26.905, "Mobile Stereoscopic 3D Video".

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

[115] 3GPP TS 26.114: "IP Multimedia Subsystem (IMS); Multimedia Telephony; Media handling and interaction".

[116] ISO/IEC 23001-8:2013 Information technology - MPEG systems technologies - Part 8: Coding-independent code points.

[117] ITU-T Recommendation H.265 (04/2013): "High efficiency video coding".

[118] 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.

[119] 3GPP TS 26.307, "Presentation Layer for 3GPP Services".

[120] 3GPP TS 26.116, "Television (TV) over 3GPP Services; Video Profiles".

[121] 3GPP TS 26.118, "3GPP Virtual reality profiles for streaming applications".

**===== CHANGE =====**

##### 5.2.3.2.2 Streaming component

Attribute name: **StreamingMethod**

Attribute definition: List of streaming methods supported by the PSS application. The client may support RTP streaming, HTTP streaming, Progressive Download, or all.

Component: Streaming

Type: Literal (Bag)

Legal values: "RTP", "HTTP", "Progressive"

Resolution rule: Append

EXAMPLE: <StreamingMethod>  
 <rdf:Bag>  
 <rdf:li>RTP</rdf:li>  
 <rdf:li>HTTP</rdf:li>  
 </rdf:Bag>  
 </StreamingMethod>

Attribute name: **StreamingAccept**

Attribute definition: List of content types (MIME types) relevant for streaming over RTP or HTTP supported by the PSS application. Content types listed shall be possible to stream over RTP. For each content type a set of MIME parameters can be specified to signal receiver capabilities. A content type that supports multiple parameter sets may occur several times in the list.

Component: Streaming

Type: Literal (Bag)

Legal values: List of MIME types with related parameters.

Resolution rule: Append

EXAMPLE: <StreamingAccept>  
 <rdf:Bag>  
 <rdf:li>audio/AMR-WB+</rdf:li>  
 <rdf:li>video/H264; profile-level-id=42e00a</rdf:li>  
 <rdf:li>video/richmedia+xml; Version-profile=10</rdf:li>  
 </rdf:Bag>  
</StreamingAccept>

Attribute name: **StreamingAccept-Subset**

Attribute definition: List of content types for which the PSS application supports a subset. MIME types can in most cases effectively be used to express variations in support for different media types. Many MIME types, e.g. AMR-WB have several parameters that can be used for this purpose. There may exist content types for which the PSS application only supports a subset and this subset cannot be expressed with MIME-type parameters. In these cases the attribute StreamingAccept-Subset is used to describe support for a subset of a specific content type. If a subset of a specific content type is declared in StreamingAccept-Subset, this means that StreamingAccept-Subset has precedence over StreamingAccept. StreamingAccept shall always include the corresponding content types for which StreamingAccept-Subset specifies subsets of.  
  
Subset identifiers and corresponding semantics shall only be defined by the TSG responsible for the present document.

Component: Streaming

Type: Literal (Bag)

Legal values: No subsets defined.

Resolution rule: Append

Attribute name: **StreamingFramePackingFormatsRTP**

Attribute definition: List of supported frame packing formats relevant for streaming of stereoscopic 3D video over RTP supported by the PSS application. The frame packing formats within scope for stereoscopic 3D video are defined in Table D-8 of [90].

Component: Streaming

Type: Literal (Bag)

Legal values: List of integer values corresponding to the supported frame packing formats. The integer values shall correspond to the ‘Value’ column as specified in Table D-8 of [90] and be interpreted according to the ‘Interpretation’ column in the same table.

Resolution rule: Append

EXAMPLE: <StreamingFramePackingFormatsRTP>  
 <rdf:Bag>  
 <rdf:li>3</rdf:li>  
 <rdf:li>4</rdf:li>  
 </rdf:Bag>  
 </StreamingFramePackingFormatsRTP>

Attribute name: **StreamingFramePackingFormatsHTTP**

Attribute definition: List of supported frame packing formats relevant for streaming of stereoscopic 3D video over HTTP supported by the PSS application. The frame packing formats within scope for stereoscopic 3D video are defined in Table D-8 of [90].

Component: Streaming

Type: Literal (Bag)

Legal values: List of integer values corresponding to the supported frame packing formats. The integer values shall correspond to the ‘Value’ column as specified in Table D-8 of [90] and be interpreted according to the ‘Interpretation’ column in the same table.

Resolution rule: Append

EXAMPLE: <StreamingFramePackingFormatsHTTP>  
 <rdf:Bag>  
 <rdf:li>3</rdf:li>  
 <rdf:li>4</rdf:li>  
 </rdf:Bag>  
 </StreamingFramePackingFormatsHTTP>

Attribute name: **StreamingCVOCapable**

Attribute definition: Indicates whether the client is a CVO capable receiver of RTP streams, i.e. provided that the video orientation information for the delivered content is communicated to the client in an RTP extension header as specified in clause 6.2.5 (corresponding to urn:3gpp:video-orientation), the client can interpret the video orientation and align the video correctly for rendering/display purposes. If this attribute is reported and the StreamingHighGranularityCVOCapable attribute is reported as a "Yes", then the value of this attribute shall be a "Yes".

Component: Streaming

Type: Literal

Legal values: "Yes", "No"

Resolution rule: Locked

EXAMPLE: <StreamingCVOCapable>Yes</StreamingCVOCapable>

Attribute name: **StreamingHighGranularityCVOCapable**

Attribute definition: Indicates whether the client is a Higher Granularity CVO capable receiver of RTP streams, i.e. provided that the video orientation information of the delivered content is communicated to the client in an RTP extension header as specified in clause 6.2.5 (corresponding to **urn:3GPP:video-orientation:6)**, the client can interpret the video orientation and align the video correctly for rendering/display purposes.

Component: Streaming

Type: Literal

Legal values: "Yes", "No"

Resolution rule: Locked

EXAMPLE: <StreamingHighGranularityCVOCapable>Yes</StreamingHighGranularityCVOCapable>

Attribute name: **ThreeGPPLinkChar**

Attribute definition: Indicates whether the device supports the 3GPP-Link-Char header according to clause 10.2.1.1.

Component: Streaming

Type: Literal

Legal values: "Yes", "No"

Resolution rule: Override

EXAMPLE: <ThreeGPPLinkChar>Yes</ThreeGPPLinkChar>

Attribute name: **AdaptationSupport**

Attribute definition: Indicates whether the device supports client buffer feedback signaling according to clause 10.2.3.

Component: Streaming

Type: Literal

Legal values: "Yes", "No"

Resolution rule: Locked

EXAMPLE: <AdaptationSupport>Yes</AdaptationSupport>

Attribute name: **QoESupport**

Attribute definition: Indicates whether the device supports QoE signaling according to clauses 5.3.2.3, 5.3.3.6, and 11 in case of RTSP/RTP-based streaming or according to clause 10 of 3GPP TS 26.247 [112] in case of HTTP-based streaming and progressive download.

Component: Streaming

Type: Literal

Legal values: "Yes", "No"

Resolution rule: Locked

EXAMPLE: <QoESupport>Yes</QoESupport>

Attribute name: **ExtendedRtcpReports**

Attribute definition: Indicates whether the device supports extended RTCP reports according to clause 6.2.3.1.

Component: Streaming

Type: Literal

Legal values: "Yes", "No"

Resolution rule: Locked

EXAMPLE: <ExtendedRtcpReports>Yes</ExtendedRtcpReports>

Attribute name: **RtpRetransmission**

Attribute definition: Indicates whether the device supports RTP retransmission according to clause 6.2.3.3.

Component: Streaming

Type: Literal

Legal values: "Yes", "No"

Resolution rule: Locked

EXAMPLE: <RtpRetransmission>Yes</RtpRetransmission>

Attribute name: **MediaAlternatives**

Attribute definition: Indicates whether the device interprets the SDP attributes "alt", "alt-default-id", and "alt-group", defined in clauses 5.3.3.3 and 5.3.3.4.

Component: Streaming

Type: Literal

Legal values: "Yes", "No"

Resolution rule: Override

EXAMPLE: <MediaAlternatives>Yes</MediaAlternatives>

Attribute name: **RtpProfiles**

Attribute definition: List of supported RTP profiles.

Component: Streaming

Type: Literal (Bag)

Legal values: Profile names registered through the Internet Assigned Numbers Authority (IANA), [www.iana.org](http://www.iana.org).

Resolution rule: Append

EXAMPLE: <RtpProfiles>  
 <rdf:Bag>  
 <rdf:li>RTP/AVP</rdf:li>  
 <rdf:li>RTP/AVPF</rdf:li>  
 </rdf:Bag>  
</RtpProfiles>

Attribute name: **ProtectedStreaming**

Attribute definition: Indicates whether the device supports streamed protected PSS as defined by Annex R.

Component: Streaming

Type: Literal

Legal values: "Yes", "No"

Resolution rule: Locked

EXAMPLE: <ProtectedStreaming>Yes</ProtectedStreaming>

Attribute name: **ThreeGPPPipelined**

Attribute definition: Indicates whether the device supports fast content start-up with pipelining according to clause 5.5.3.

Component: Streaming

Type: Literal

Legal values: "Yes", "No"

Resolution rule: Override

EXAMPLE: <ThreeGPPPipelined>Yes</ThreeGPPPipelined>

Attribute name: **ThreeGPPSwitch**

Attribute definition: Indicates whether the device supports fast content switching with known SDP according to clause 5.5.4.3.

Component: Streaming

Type: Literal

Legal values: "Yes", "No"

Resolution rule: Override

EXAMPLE: <ThreeGPPSwitch>Yes</ThreeGPPSwitch>

Attribute name: **ThreeGPPSwitchReqSDP**

Attribute definition: Indicates whether the device supports fast content switching without SDP according to clause 5.5.4.4.

Component: Streaming

Type: Literal

Legal values: "Yes", "No"

Resolution rule: Override

EXAMPLE: <ThreeGPPSwitchReqSDP>Yes</ThreeGPPSwitchReqSDP>

Attribute name: **ThreeGPPSwitchStream**

Attribute definition: Indicates whether the device supports the fast switching of media streams according to clause 5.5.4.5.

Component: Streaming

Type: Literal

Legal values: "Yes", "No"

Resolution rule: Override

EXAMPLE: <ThreeGPPSwitchStream>Yes</ThreeGPPSwitchStream>

Attribute name: **AcceptRanges**

Attribute definition: List of range indications that are accepted by the client. The client may support UTC or NPT or both.

Component: Streaming

Type: Literal (Bag)

Legal values: "NPT", "UTC"

Resolution rule: Append

EXAMPLE: <AcceptRanges>  
 <rdf:Bag>  
 <rdf:li>NPT</rdf:li>  
 <rdf:li>UTC</rdf:li>  
 </rdf:Bag>  
 </AcceptRanges>

Attribute name: **ISMACryp**

Attribute definition: Indicates whether the device supports streamed content in ISMACryp format as defined in ISMACryp and Annex R.

Component: Streaming

Type: Literal (Bag)

Legal values: ISMACryp Version numbers supported as a floating number. 0.0 indicates no support.

Resolution rule: Locked

EXAMPLE: <ISMACryp>  
 <rdf:Bag>  
 <rdf:li>2.0</rdf:li>  
 </rdf:Bag>  
</ISMACryp>

Attribute name: **VideoDecodingByteRate**

Attribute definition: If Annex G is not supported, the attribute has no meaning. If Annex G is supported, this attribute defines the peak decoding byte rate the PSS client is able to support. In other words, the PSS client fulfils the requirements given in Annex G with the signalled peak decoding byte rate. The values are given in bytes per second and shall be greater than or equal to 16000.

Component: Streaming

Type: Number

Legal values: Integer value greater than or equal to 16000.

Resolution rule: Locked

EXAMPLE: <VideoDecodingByteRate>16000</VideoDecodingByteRate>

Attribute name: **VideoInitialPostDecoderBufferingPeriod**

Attribute definition: If Annex G is not supported, the attribute has no meaning. If Annex G is supported, this attribute defines the maximum initial post-decoder buffering period of video. Values are interpreted as clock ticks of a 90-kHz clock. In other words, the value is incremented by one for each 1/90 000 seconds. For example, the value 9000 corresponds to 1/10 of a second initial post-decoder buffering.

Component: Streaming

Type: Number

Legal values: Integer value equal to or greater than zero.

Resolution rule: Locked

EXAMPLE: <VideoInitialPostDecoderBufferingPeriod>9000  
</VideoInitialPostDecoderBufferingPeriod>

Attribute name: **VideoPreDecoderBufferSize**

Attribute definition: This attribute signals if the optional video buffering requirements defined in Annex G are supported. It also defines the size of the hypothetical pre-decoder buffer defined in Annex G. A value equal to zero means that Annex G is not supported. A value equal to one means that Annex G is supported. In this case the size of the buffer is the default size defined in Annex G. A value equal to or greater than the default buffer size defined in Annex G means that Annex G is supported and sets the buffer size to the given number of octets.

Component: Streaming

Type: Number

Legal values: Integer value equal to or greater than zero. Values greater than one but less than the default buffer size defined in Annex G are not allowed.

Resolution rule: Locked

EXAMPLE: <VideoPreDecoderBufferSize>30720</VideoPreDecoderBufferSize>

**===== CHANGE =====**

##### 5.2.3.2.3 ThreeGPFileFormat component

Attribute name: **Brands**

Attribute definition: List of supported 3GP profiles identified by brand.

Component: ThreeGPFileFormat

Type: Literal (Bag)

Legal values: Brand identifiers according to 5.3.4 and 5.4 in [50].

Resolution rule: Append

EXAMPLE: <Brands>  
 <rdf:Bag>  
 <rdf:li>3gp4</rdf:li>  
 <rdf:li>3gp5</rdf:li>  
 <rdf:li>3gp6</rdf:li>  
 <rdf:li>3gr6</rdf:li>  
 <rfd:li>3gp7</rdf:li>  
 <rfd:li>3gr7</rdf:li>  
 <rfd:li>3ge7</rdf:li>  
 </rdf:Bag>  
</Brands>

Attribute name: **ThreeGPAccept**

Attribute definition: List of content types (MIME types) that can be included in a 3GP file and handled by the PSS application. The content types included in this attribute can be rendered in a 3GP file or a presentation contained therein. If the identifier "Streaming-Media" is included, streaming media can be included within a contained presentation. Details on the streaming support can then be found in the Streaming component. For each content type a set of supported parameters can be given. A content type that supports multiple parameter sets may occur several times in the list.

Component: ThreeGPFileFormat

Type: Literal (Bag)

Legal values: List of MIME types with related parameters and the "Streaming-Media" identifier.

Resolution rule: Append

EXAMPLE 1: <ThreeGPAccept>  
 <rdf:Bag>  
 <rdf:li>audio/AMR</rdf:li>  
 <rdf:li>audio/AMR-WB+</rdf:li>  
  
 <rdf:li>video/H264; profile-level-id=42e00a</rdf:li>  
 <rdf:li>image/jpeg</rdf:li>  
 <rdf:li>video/richmedia+xml; Version-profile=10</rdf:li>  
 <rdf:li>Streaming-Media</rdf:li>  
 </rdf:Bag>  
</ThreeGPAccept>

Attribute name: **ThreeGPAccept-Subset**

Attribute definition: List of content types for which the PSS application supports a subset. MIME types can in most cases effectively be used to express variations in support for different media types. Many MIME types have several parameters that can be used for this purpose. There may exist content types for which the PSS application only supports a subset and this subset cannot be expressed with MIME-type parameters. In these cases the attribute ThreeGPAccept-Subset is used to describe support for a subset of a specific content type. If a subset of a specific content type is declared in ThreeGPAccept-Subset, this means that ThreeGPAccept-Subset has precedence over ThreeGPAccept. ThreeGPAccept shall always include the corresponding content types for which ThreeGPAccept-Subset specifies subsets of.  
  
Subset identifiers and corresponding semantics shall only be defined by the TSG responsible for the present document.

Component: ThreeGPFileFormat

Type: Literal (Bag)

Legal values: No subsets defined.

Resolution rule: Append

Attribute name: **ThreeGPFramePackingFormats**

Attribute definition: List of supported frame packing formats relevant for stereoscopic 3D video that can be included in a 3GP file and handled by the PSS application.

Component: ThreeGPFileFormat

Type: Literal (Bag)

Legal values: List of integer values corresponding to the supported frame packing formats. Integer values shall be either 3 or 4 corresponding to the Side-by-Side and Top-and-Bottom frame packing formats respectively, as specified in the ‘Value’ column of Table D-8 of [90] and interpreted according to the ‘Interpretation’ column in the same table.

Resolution rule: Append

EXAMPLE: <ThreeGPFramePackingFormats>  
 <rdf:Bag>  
 <rdf:li>3</rdf:li>  
 <rdf:li>4</rdf:li>  
 </rdf:Bag>  
 </ThreeGPFramePackingFormats>

Attribute name: **ThreeGPCVOCapable**

Attribute definition: Indicates whether the client is a CVO capable receiver of 3GP files, i.e. provided that the video orientation information (corresponding to urn:3gpp:video-orientation) of the delivered content is communicated to the client in a 3GP file, the client can interpret the video orientation and align the video correctly for rendering/display purposes. If this attribute is reported and the ThreeGPHighGranularityCVOCapable attribute is reported as a "Yes", then the value of this attribute shall be a "Yes".

Component: ThreeGPFileFormat

Type: Literal

Legal values: "Yes", "No"

Resolution rule: Locked

EXAMPLE: <ThreeGPCVOCapable>Yes</ThreeGPCVOCapable>

Attribute name: **ThreeGPHighGranularityCVOCapable**

Attribute definition: Indicates whether the client is a Higher Granularity CVO capable receiver of 3GP files, i.e. provided that the video orientation information (corresponding to urn:3gpp:video-orientation:6) of the delivered content is communicated to the client in a 3GP file, the client can interpret the video orientation and align the video correctly for rendering/display purposes.

Component: ThreeGPFileFormat

Type: Literal

Legal values: "Yes", "No"

Resolution rule: Locked

EXAMPLE: <ThreeGPHighGranularityCVOCapable>Yes</ThreeGPHighGranularityCVOCapable>

Attribute name: **ThreeGPOmaDrm**

Attribute definition: List of the OMA DRM versions that is supported to be used for DRM protection of content present in the 3GP file format.

Component: ThreeGPFileFormat

Type: Literal (Bag)

Legal values: OMA DRM version numbers as floating point values. 0.0 indicates no support.

Resolution rule: Locked

EXAMPLE: <3gpOMADRM>  
 <rdf:Bag>  
 <rdf:li>2.0 </rdf:li>  
 </rdf:Bag>  
</3gpOMADRM>

**===== CHANGE =====**

###### 5.3.2.3.3.2 Example XML for the QoE Report

The example shows a QoE report for a streaming session.

<?xml version="1.0" encoding="UTF-8"?>

<receptionReport xmlns="urn:3gpp:metadata:2009:PSS:receptionreport"

xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance"

xsi:schemaLocation="urn:3gpp:metadata:2009:PSS:receptionreport receptionreport.xsd">

<statisticalReport

clientId="79261234567"

<qoeMetrics

numberOfRebufferingEvents="0 1 0"

initialBufferingDuration="3.213"

totalRebufferingDuration="0 1.23 0"

contentAccessTime="2.621"

sessionStartTime="1219322514"

sessionStopTime="1219322541">

bufferDepth="3.571 2.123 2.241"

allContentBuffered="false">

<medialevel\_qoeMetrics

sessionId="10.50.65.30:5050"

framerate="15.1 14.8 15.0"

t="false"

d="a"

numberOfSuccessiveLossEvents="5 0 3"

numberOfCorruptionEvents="6 5 2"

numberOfJitterEvents="0 1 0"

totalCorruptionDuration="152 234 147"

totalNumberofSuccessivePacketLoss="25 0 6"

numberOfReceivedPackets="456 500 478"

codecInfo=" video/H264/90000 = ="

codecProfileLevel="profile-level-id=42e00a = ="

codecImageSize="176x144 = ="

averageCodecBitRate="124.5 128.0 115.1"

totalJitterDuration="0 0.346 0"/>

</qoeMetrics>

</statisticalReport>

</receptionReport>

**===== CHANGE =====**

#### 5.3.2.4 Video buffering headers

The following header fields are specified for the response of an RTSP PLAY request only:

- x-predecbufsize:<size of the pre-decoder buffer>

- x-initpredecbufperiod:<initial pre-decoder buffering period>

- x-initpostdecbufperiod:<initial post-decoder buffering period>

- 3gpp-videopostdecbufsize:<size of the video post-decoder buffer>

The header fields "x-predecbufsize", "x-initpredecbufperiod", "x-initpostdecbufperiod", and "3gpp-postdecbufsize" have the same definitions as the corresponding SDP attributes (see clause 5.3.3.2) "X-predecbufsize", "X-initpredecbufperiod", "X-initpostdecbufperiod", and "3gpp-postdecbufsize", respectively, with the exception that the RTSP video buffering header fields are valid only for the range specified in the RTSP PLAY response.

For H.264 (AVC) or H.265 (HEVC), PSS servers shall include these header fields in an RTSP PLAY response whenever the values are available in the 3GP file used for the streaming session. If the values are not available in the 3GP file, it is optional for the servers to signal the parameter values in RTSP PLAY responses.

**===== CHANGE =====**

#### 5.3.3.2 Additional SDP fields

The following additional media level SDP fields are defined for PSS:

-

If the field is an attribute for an H.264 (AVC) stream, the H.264 (AVC) bitstream is constrained by the value of "CpbSize" equal to X-predecbufsize \* 8 for NAL HRD parameters, as specified in [90]. For the VCL HRD parameters, the value of "CpbSize" is equal to X-predecbufsize \* 40 / 6. The value of "X-predecbufsize" for H.264 (AVC) streams shall be smaller than or equal to 1200 \* MaxCPB, in which the value of "MaxCPB" is derived according to the H.264 (AVC) profile and level of the stream, as specified in [90]. If "X-predecbufsize" is not present for an H.264 (AVC) stream, the value of "CpbSize" is calculated as specified in [90].

If the field is an attribute for an H.265 (HEVC) stream, the H.265 (HEVC) bitstream is constrained such that, for the NAL HRD parameters, the value of CpbSize[ i ] for at least one value of i in the range of 0 to cpb\_cnt\_minus1[ HighestTid ], inclusive, as specified in [117], is less than or equal to X-predecbufsize \* 8, and for the VCL HRD parameters, the value of CpbSize[ i ] for at least one value of i in the range of 0 to cpb\_cnt\_minus1[ HighestTid ], is less than or equal to X-predecbufsize \* 80 / 11. The value of "X-predecbufsize" for H.265 (HEVC) streams shall be smaller than or equal to 1100 \* MaxCPB, in which the value of "MaxCPB" is derived according to the H.265 (HEVC) profile and level of the stream, as specified in [117]. If "X-predecbufsize" is not present for an H.265 (HEVC) stream, the value of "CpbSize" is calculated as specified in [117].

- "a=X-initpredecbufperiod:<initial pre-decoder buffering period>"

If the field is an attribute for an H.264 (AVC) stream, the H.264 (AVC) bitstream is constrained by the value of the nominal removal time of the first access unit from the coded picture buffer (CPB), tr,n( 0 ), equal to "X-initpredecbufperiod" as specified in [90]. If "X-initpredecbufperiod" is not present for an H.264 (AVC) stream, tr,n( 0 ) shall be equal to the earliest time when the first access unit in decoding order has been completely received.

If the field is an attribute for an H.265 (HEVC) stream, the H.265 (HEVC) bitstream is constrained such that the value of the nominal removal time of the first access unit from the coded picture buffer (CPB), AuNominalRemovalTime[ 0 ], as specified in [117], is equal to "X-initpredecbufperiod". If "X-initpredecbufperiod" is not present for an H.265 (HEVC) stream, the value of AuNominalRemovalTime[ 0 ] shall be equal to the earliest time when the first access unit in decoding order has been completely received.

- "a=X-initpostdecbufperiod:<initial post-decoder buffering period>"

If the field is an attribute for an H.264 (AVC) stream, the H.264 (AVC) bitstream is constrained by the value of dpb\_output\_delay for the first decoded picture in output order equal to "X-initpostdecbufperiod" as specified in [90] assuming that the clock tick variable, tc, is equal to 1 / 90 000. If "X-initpostdecbufperiod" is not present for an H.264 (AVC) stream, the value of dpb\_output\_delay for the first decoded picture in output order is inferred to be equal to 0.

If the field is an attribute for an H.265 (HEVC) stream, the H.265 (HEVC) bitstream is constrained such that the value of pic\_dpb\_output\_delay for the first decoded picture in output order, as specifeid in [117], is equal to "X-initpostdecbufperiod", assuming that the clock tick, ClockTick, is equal to 1 / 90 000. If "X-initpostdecbufperiod" is not present for an H.265 (HEVC) stream, the value of pic\_dpb\_output\_delay for the first decoded picture in output order is inferred to be equal to 0.

- "a=X-decbyterate:<peak decoding byte rate>"

This field shall not be present for an H.264 (AVC) or H.265 (HEVC) stream.

- "a=3gpp-videopostdecbufsize:<size of the video post-decoder buffer>"

This attribute may be present for an H.264 (AVC) or H.265 (HEVC) stream and it shall not be present for other types of streams.

If the attribute is present for an H.264 (AVC) stream, the H.264 (AVC) bitstream is constrained by the value of "max\_dec\_frame\_buffering" equal to Min( 16, Floor( 3gpp-videopostdecbufsize / ( PicWidthInMbs \* FrameHeightInMbs \* 256 \* ChromaFormatFactor ) ) ) as specified in [90]. If "3gpp-videopostdecbufsize" is not present for an H.264 (AVC) stream, the value of "max\_dec\_frame\_buffering" is inferred as specified in [90].

If the attribute is present for an H.265 (HEVC) Main profile stream, the H.265 (HEVC) bitstream is constrained such that the value of sps\_max\_dec\_pic\_buffering\_minus1[ HighestTid ] + 1 as specified in [117] is less than or equal to Floor( 3gpp-videopostdecbufsize / ( PicSizeInSamplesY \* 3 / 2 ) ), where PicSizeInSamplesY is as specified in [117]. If "3gpp-videopostdecbufsize" is not present for an H.265 (HEVC) stream, the value of sps\_max\_dec\_pic\_buffering\_minus1[ HighestTid ] + 1 is inferred as specified in [117].

If the interleaved packetization mode of H.264 (AVC) is in use, attributes "a=X-predecbufsize:", "a=X-initpredecbufperiod:", "a=X-initpostdecbufperiod:", and "a=3gpp-videopostdecbufsize:" apply to an H.264 (AVC) bitstream when de-interleaving of the stream from transmission order to decoding order has been done.

For an H.265 (HEVC) stream transmitted over RTP using the RTP payload format as specified in [118], the attributes "a=X-predecbufsize:", "a=X-initpredecbufperiod:", "a=X-initpostdecbufperiod:", and "a=3gpp-videopostdecbufsize:" apply to the video stream that is the output of the de-packetization process.

The following media level SDP field is defined for PSS:

- "a=framesize:<payload type number> <width>-<height>"

The frame size field in SDP is needed by the client in order to properly allocate frame buffer memory. For H.264 (AVC) streams, the frame size shall be extracted from the sprop-parameters-sets parameter in the SDP, when present. For H.265 (HEVC) streams, the frame size shall be extracted from the sprop-sps parameter in the SDP, when present.

If this attribute is present, the frame size parameters shall exactly match the largest frame size defined in the video stream. The width and height values shall be expressed in pixels.

If RTP retransmission is supported, the following SDP attribute shall be supported by the client and server:

- "a=rtcp-fb" according to clause 4.2 in [57].

If CVO information is signalled in the RTP Header Extension as specified in clause 6.2.5, the PSS server shall signal this in the SDP by including the a=extmap attribute [114] indicating the CVO URN under the relevant media line scope. The CVO URN is: urn:3gpp:video-orientation. Here is an example usage of this URN to signal CVO relative to a media line:

a=extmap:7 urn:3gpp:video-orientation

The number 7 in the example may be replaced with any number in the range 1-14.

If Higher Granularity CVO information is signalled in the RTP Header Extension as specified in clause 6.2.5, the PSS server shall signal this in the SDP in a similar fashion with the CVO URN: urn:3gpp:video-orientation:6. Here is an example usage of this URN to signal CVO relative to a media line:

a=extmap:5 urn:3gpp:video-orientation:6

The following media level SDP attribute is defined, in ABNF [53] format, for PSS:

sdp-3GPP-frame-packing-type-line = "a" "=" "3GPP-framepackingtype" ":" frame-packing-type ":" payload-type-number CRLF

frame-packing-type  =  1\*DIGIT

payload-type-number = 1\*DIGIT

The frame-packing-type value specifies the frame packing format of the described frame-packed stereoscopic 3D video. The frame-packing-type value is an integer value that shall be equal to a value in the ‘Value’ column of VideoFramePackingType table specified in [116] and be interpreted according to the ‘Interpretation’ column in the same table. The payload-type-number value indicates to which payload formats the attribute applies to.

If offering frame-packed stereoscopic 3D video as defined in clause 7.4, a PSS server shall include the sdp-3GPP-frame-packing-type-line at the media level. If a PSS client supports frame packed stereoscopic 3D video as defined in clause 7.4, then it shall be able to interpret this SDP attribute when present. The absence of this attribute indicates that the video component is not a frame-packed stereoscopic 3D video.

NOTE: If a PSS client supports frame-packed stereoscopic 3D video, frame packing types as defined in clause 7.4 are supported by the PSS client.

## 5.4 MIME media types

For continuous media the following MIME media types shall be used:

- AMR narrow-band speech codec (see sub-clause 7.2) MIME media type as defined in [11];

- AMR wideband speech codec (see sub-clause 7.2) MIME media type as defined in [11];

- Extended AMR-WB codec (see sub-clause 7.3) MIME media type as defined in [85];

- Enhanced aacPlus and MPEG-4 AAC audio codecs (see clause 7.3) MIME media type as defined in RFC 6416 [13].  
The following applies to servers when this MIME type is used in SDP:

1) Configuration information is exclusively carried out-of-band in the SDP "config" parameter; this shall be signaled by sending "cpresent=0".

2) A PSS server serving implicitly signaled Enhanced aacPlus content shall include "SBR-enabled=1" in the "a=fmtp" line; it shall include "SBR-enabled=0" if it serves plain AAC content.

3) A PSS server serving explicitly signaled content is recommended not to include the "SBR-enabled" parameter in the "a=fmtp" line.

Therefore, the following applies to terminals:

1) The rtpmap rate parameter should not be considered definitive of the sampling rate (though it is, of course, definitive of the timescale of the RTP timestamps).

2) If explicit signaling is in use, the StreamMuxConfig contains both the core AAC sampling rate and the SBR sampling rate. The appropriate output sampling rate may be chosen dependant on Enhanced aacPlus support.

3) If explicit signalling is not in use and no SBR-enabled parameter is present, the StreamMuxConfig contains the AAC sampling rate and the appropriate output sampling rate may be set to this indicated rate.

4) If explicit signalling is not in use and the SBR-enabled parameter is present, terminals supporting Enhanced aacPlus should set the output sampling rate to either the core AAC sampling rate as indicated in the StreamMuxConfig [21] (where "SBR-enabled" is set to "0") or twice the indicated rate (where "SBR-enabled" is set to "1");

- H.264 (AVC) [90] video codec (see sub-clause 7.4) MIME media type as defined in [92];

- H.265 (HEVC) [117] video codec (see sub-clause 7.4) MIME media type as defined in [118];

- 3GPP timed text format [51] MIME media type as defined in sub-clause 7.1 of [80];

- enc-isoff-generic MIME media type as defined in [102] and used in Annex R;

- RTP retransmission payload format MIME media types as defined in clause 8 of [81].

MIME media types for JPEG, GIF, PNG, SP-MIDI, Mobile DLS, Mobile XMF, SVG, timed text and 3GP can be used in the "Content-type" field in HTTP, "content\_type" field in the item information box of 3GP files. The following MIME media types shall be used for these media:

- JPEG (see sub-clause 7.5) MIME media type as defined in [15];

- GIF (see sub-clause 7.6) MIME media type as defined in [15];

- PNG (see sub-clause 7.6) MIME media type as defined in [38];

- SP-MIDI (see sub-clause 7.3A) MIME media type as defined in clause C.2 in Annex C of the present document;

- DLS MIME media type to represent Mobile DLS (see sub-clause 7.3A) as defined in [97];

- Mobile XMF (see sub-clause 7.3A) MIME media type as defined in clause C.3 in Annex C of the present document;

- SVG (see sub-clause 7.7) MIME media type as defined in [42];

- Timed text (see sub-clause 7.9) MIME media type as defined in [79];

- 3GP files (see sub-clause 7.10) MIME media type as defined in [79].

NOTE: The 3GP MIME media type [79] is used for all 3GP files, including 3GP files carrying timed text, images, etc.

#### 6.2.3.2 RTCP App packet for client buffer feedback (NADU APP packet)

A PSS client supporting Signalling for Client Buffer Feedback (see clause 10.2.3) shall report the next application data unit to be decoded for buffer status reporting and rate adaptation by sending the RTCP APP packet. A NADU APP packet shall be sent only after the client has received at least one RTP packet on the media stream and shall be accompanied by a complementary RR packet. The RR and NADU packets shall contain information that represents a single simultaneous 'snapshot' of the media stream. The format of a generic RTCP APP packet is shown in Figure 3 below:

0 1 2 3   
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1   
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+  
|V=2|P| subtype | PT=APP=204 | length |  
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+  
| SSRC/CSRC |  
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+  
| name (ASCII) |  
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+  
| application-dependent data ...  
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+

Figure 3: Generic Format of an RTCP APP packet.

For rate adaptation the name and subtype fields must be set to the following values:

*name*: The NADU APP data format is detected through the name "PSS0", i.e. 0x50535330 and the subtype.

*subtype*: This field shall be set to 0 for the NADU format.

*length*: The number of 32 bit words –1, as defined in RFC 3550 [9]. This means that the field will be 2+3\*N, where N is the number of sources reported on. The length field will typically be 5, i.e. 24 bytes packets.

*application-dependent data*: One or more of the following data format blocks (as described in Figure 4) can be included in the application-dependent data location of the APP packet. The APP packets length field is used to detect how many blocks of data are present. The block shall be sent for the SSRCs for which there are a report block as part of either a Receiver Report or a Sender Report, included in the RTCP compound packet. A NADU APP packet shall not contain any other data format than the one described in figure 4 below.

0 1 2 3  
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1  
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+  
| SSRC |  
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+  
| Playout Delay | NSN |  
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+  
| Reserved | NUN | Free Buffer Space (FBS) |  
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+

Figure 4: Data format block for NADU reporting

*SSRC*: The SSRC of the media stream the buffered packets belong to.

*Playout delay (16 bits)*: The difference in milliseconds between the scheduled playout time of the next ADU to be decoded, (whose sequence number is indicated in the NSN field) and the current time when generating the RTCP packet that contains the NADU APP block, both measured on the media playout clock. The client shall always indicate this value, unless it is not well defined, when it may use the reserved value (0xFFFF). When the buffer is empty (the client has not yet received the packet with sequence number NSN), the playout delay is not well defined and the client should use the reserved value 0xFFFF for this field. When the media clock is not advancing (e.g. while paused or re-buffering), the playout delay corresponds to the difference between the playout time of the next ADU and the media time at which playout will resume.

The point at which the media playout clock is measured should be chosen such that, if the only packet in the buffer is that with sequence number NSN, the playout delay indicates the time remaining until the media playout will 'starve' and this stream might need re-buffering. In the calculations of playout delay above, this point is used to determine the playout point of a media packet even though actual playout may occur later in the decoding chain. The target buffer time (see clause 5.3.2.2) must be measured from the same point.

The playout delay allows the server to have a more precise value of the amount of time before the client will underflow. The playout delay shall be computed until the actual media playout (i.e., audio playback or video display).

*NSN (16 bits)*: The RTP sequence number of the next ADU to be decoded for the SSRC reported on. In the case where the buffer does not contain any packets for this SSRC, the next not yet received sequence number shall be reported, i.e. an NSN value that is one larger than the least significant 16 bits of the RTCP SR or RR report block's "extended highest sequence number received".

*NUN (5 bits)*: The unit number (within the RTP packet) of the next ADU to be decoded. The first unit in a packet has a unit number equal to zero. The unit number is incremented by one for each ADU in an RTP packet. In the case of an audio codec, an ADU is defined as an audio frame. In the case of H.264 (AVC) or H.265 (HEVC), an ADU is defined as a NAL unit.

*FBS (16 bit)*: The amount of free buffer space available in the client at the time of reporting. The reported free buffer space shall be less than or equal to the buffer space that has been reported as available for adaptation by the 3GPP-Adaptation RTSP header, see clause 5.3.2.2. The amount of free buffer space are reported in number of complete 64 byte blocks, thus allowing for up to 4194304 bytes to be reported as free. If more is available, it shall be reported as the maximal amount available, i.e. 4194304 with a field value 0xffff.

*Reserved (11 bits)*: These bits are not used and shall be set to 0 and shall be ignored by the receiver.

### 6.2.4 RTP payload formats

For RTP/UDP/IP transport of continuous media the following RTP payload formats shall be used:

- AMR narrow-band speech codec (see clause 7.2) RTP payload format according to [11]. A PSS client is not required to support multi-channel sessions;

- AMR wideband speech codec (see clause 7.2) RTP payload format according to [11]. A PSS client is not required to support multi-channel sessions;

- Extended AMR-WB codec (see clause 7.3) RTP payload format according to [85];

- Enhanced aacPlus and MPEG-4 AAC codec (see clause 7.3) RTP payload format according to [13]; the size of audioMuxElements shall be limited to the maximum size of one audio frame, which is 6144 bits per AAC channel; moreover multiplexing of multiple audio frames into one audioMuxElement should be avoided if this would lead to fragmentation across RTP packets;

- H.264 (AVC) video codec (see clause 7.4) RTP payload format according to [92]. A PSS client is required to support all three packetization modes: single NAL unit mode, non-interleaved mode and interleaved mode. For the interleaved packetization mode, a PSS client shall support streams for which the value of the "sprop-deint-buf-req" MIME parameter is less than or equal to MaxCPB \* 1000 / 8, inclusive, in which "MaxCPB" is the value for VCL parameters of the H.264 (AVC) profile and level in use, as specified in [90]. Parameter sets shall not be transmitted within the RTP payload, i.e., all parameter sets required for a session must be provided in the SDP;

- H.265 (HEVC) video codec (see clause 7.4) RTP payload format according to [118];

- 3GPP timed text format (see clause 7.9) RTP payload format according to [80];

- encrypted "enc-isoff-generic" (see Annex R) RTP payload format according [102];

- RTP retransmission payload format according to [81];

- RTP Header Extension to signal CVO information as specified in clause 6.2.5.

### 7.4.1 General video decoder requirements

If a PSS client supports video, the following applies:

- H.264 (AVC) Progressive High Profile Level 3.1 decoder [90] shall be supported, wherein the maximum VCL Bit Rate is constrained to be 14Mbps with cpbBrVclFactor and cpbBrNalFactor being fixed to be 1000 and 1200, respectively.

- H.265 (HEVC) Main Profile, Main Tier, Level 3.1 decoder [117] should be supported.

When H.265 (HEVC) Main Profile decoder is supported, the client is only required to process H.265 (HEVC) Main Profile bitstreams that have general\_progressive\_source\_flag equal to 1, general interlaced\_source\_flag equal to 0, general\_non\_packed\_constraint\_flag equal to 1, and general\_frame\_only\_constraint\_flag equal to 1.

NOTE 1: An H.264 (AVC) High Profile decoder is able to decode an H.264 (AVC) Main Profile stream that is progressively encoded.

#### 11.2.9.1 Default reporting format

The codec information metrics contain details of the media codec used during the measurement period. The unit of this metric is a string value. No "white space" characters are allowed in the string values, and shall be removed if necessary.

For audio media the codec information contains the audio codec type, represented as in an SDP offer, for instance "AMR-WB/16000/1".

For video media, the codec information contains the video codec type, represented as in an SDP offer. Furthermore, the video profile and level used, as well as the image size used shall be reported. For instance "profile=0;level=45" for the profile and level information and "176x144" for the image size. In some cases the profile and level is reported together, for instance "profile-level-id=42e00a". Note that the image size reported for each measurement resolution period shall be the one actually used, not the maximum size allowed by the SDP negotiation.

For timed text media, the codec information contains the text encoding, represented as in an SDP offer, for instance "3gpp-tt/1000".

The syntax for the metric "CodecInfo", "CodecProfileLevel" and "CodecImageSize" are defined in sub-clause 5.3.2.3.2.

There is the possibility that the codec information is changed during the measurement period. In that case the metrics can occur more than once indicating the codecs used.

The optional "Timestamp" field indicates the time when codec changes have occurred, relative to the beginning of the measurement period.

# A.1 SDP

This clause gives some background information on SDP for PSS clients.

Table A.1 provides an overview of the different SDP fields that can be identified in a SDP file. The order of SDP fields is mandated as specified in RFC 4566 [6].

Table A.1: Overview of fields in SDP for PSS clients

|  |  |  |  |  |
| --- | --- | --- | --- | --- |
| Type | Description | | Requirement according to [6] | Requirement according to the present document |
| Session Description | | | | |
| V | Protocol version | | R | R |
| O | Owner/creator and session identifier | | R | R |
| S | Session Name | | R | R |
| I | Session information | | O | O |
| U | URI of description | | O | O |
| E | Email address | | O | O |
| P | Phone number | | O | O |
| C | Connection Information | | R | R |
| B | Bandwidth information | AS | O | O |
| RS | ND | O |
| RR | ND | O |
| TIAS | ND | O |
| One or more Time Descriptions (See below) | | | | |
| Z | Time zone adjustments | | O | O |
| K | Encryption key | | O | O |
| A | Session attributes | control | O | R |
| range | O | R |
| alt-group | ND | O |
| 3GPP-QoE-Metrics | ND | O |
| 3GPP-Asset-Information | ND | O |
| maxprate | ND | O |
| One or more Media Descriptions (See below) | | | | |
|  | | | | |
| Time Description | | | | |
| T | Time the session is active | | R | R |
| R | Repeat times | | O | O |
|  | | | | |
| Media Description | | | | |
| M | Media name and transport address | | R | R |
| I | Media title | | O | O |
| C | Connection information | | R | R |
| B | Bandwidth information | AS | O | R |
| RS | ND | R |
| RR | ND | R |
| TIAS | ND | R |
| K | Encryption Key | | O | O |
| A | Attribute Lines | control | O | R |
| range | O | R |
| fmtp | O | R |
| rtpmap | O | R |
| X-predecbufsize | ND | O |
| X-initpredecbufperiod | ND | O |
| X-initpostdecbufperiod | ND | O |
| X-decbyterate | ND | O |
| 3GPP-framepackingtype | ND | R (see note 7) |
| framesize | ND | R (see note 5) |
| alt | ND | O |
| alt-default-id | ND | O |
| 3GPP-Adaptation-Support | ND | O |
| 3GPP-QoE-Metrics | ND | O |
| 3GPP-Asset-Information | ND | O |
| rtcp-fb | O | O |
| maxprate | ND | R |
| Note 1: R = Required, O = Optional, ND = Not Defined  Note 2: The "c" type is only required on the session level if not present on the media level.  Note 3: The "c" type is only required on the media level if not present on the session level.  Note 4: According to RFC 4566, either an 'e' or 'p' field must be present in the SDP description. On the other hand, both fields will be made optional in the future release of SDP. So, for the sake of robustness and maximum interoperability, either an 'e' or 'p' field shall be present during the server's SDP file creation, but the client should also be ready to receive SDP content containing neither 'e' nor 'p' fields.  Note 5: The "framesize" attribute is not required for any codec.  Note 6: The "range" attribute is required on either session or media level: it is a session-level attribute unless the presentation contains media streams of different durations. If a client receives "range" on both levels, however, media level shall override session level.  Note 7: The "3GPP-framepackingtype" attribute is only required for frame-packed stereoscopic 3D video as described in Section 7.4. | | | | |

The example below shows an SDP file that could be sent to a PSS client to initiate unicast streaming of a H.264 video sequence.

EXAMPLE 1: v=0  
o=ghost 2890844526 2890842807 IN IP4 192.168.10.10  
s=3GPP Unicast SDP Example  
i=Example of Unicast SDP file  
u=http://www.infoserver.com/ae600  
e=ghost@mailserver.com  
c=IN IP4 0.0.0.0  
t=0 0

a=range:npt=0-45.678  
m=video 1024 RTP/AVP 96

b=AS:1030

b=TIAS:1000000

a=maxprate:90  
a=rtpmap:96 H264/90000

a=fmtp:96 packetization-mode=1; profile-level-id=64001e; \

sprop-parameter-sets= Z2QAHpWQC0PaAfyQ,aOuOoA==

a=control:rtsp://mediaserver.com/movie.3gp/trackID=1

The following examples show some usage of the "alt" and the "alt-default-id" attributes (only the affected part of the SDP is shown):

EXAMPLE 2: m=audio 0 RTP/AVP 97

b=AS:12

b=TIAS:8500

a=maxprate:10

a=rtpmap:97 AMR/8000

a=control:trackID=1

a=fmtp:97 octet-align=1

a=range:npt=0-150.2

a=alt-default-id:1

a=alt:2:b=AS:16

a=alt:2:b=TIAS:12680

a=alt:2:a=control:trackID=2

The equivalent SDP for alternative 1 (default) is:

EXAMPLE 3: m=audio 0 RTP/AVP 97

b=AS:12

b=TIAS:8500

a=maxprate:10

a=rtpmap:97 AMR/8000

a=control:trackID=1

a=fmtp:97 octet-align=1

a=range:npt=0-150.2

Alternative 2 is based on the default alternative but replaces two lines, "b=AS" and "a=control". Hence, the equivalent SDP for alternative 2 is:

EXAMPLE 4: m=audio 0 RTP/AVP 97

b=AS:16

b=TIAS:12680

a=maxprate:10

a=rtpmap:97 AMR/8000

a=control:trackID=2

a=fmtp:97 octet-align=1

a=range:npt=0-150.2

Below is an example on the usage of the "alt-group" attribute with the subtype "BW":

EXAMPLE 5: a=alt-group:BW:AS:32=1,4;56=2,4;64=3,5

The above line gives three groupings based on application-specific bitrate values. The first grouping will result in 32 kbps using media alternatives 1 and 4. The second grouping has a total bitrate of 56 kbps using media alternatives 2 and 4. The last grouping needs 64 kbps when combing media alternatives 3 and 5.

Here follows an example on the usage of the "alt-group" attribute with the subtype "LANG":

EXAMPLE 6: a=alt-group:LANG:RFC3066:en-US=1,2,4,5;se=3,4,5

The above line claims that the media alternatives 1, 2, 4, and 5 support US English and that the media alternatives 3, 4, and 5 support Swedish.

A more complex example where a combination of "alt", "alt-default-id" and "alt-group" are used is seen below. The example allows a client to select a bandwidth that is suitable for the current context in an RTSP SETUP message.

The client sends an RTSP DESCRIBE to the server and the server responds with the following SDP. A client, who supports the "alt", "alt-default-id" and "alt-group" attributes, can now select the most suitable alternative by using the control URLs corresponding to the selected alternatives in the RTSP SETUP message. The server sets up the selected alternatives and the client starts playing them. If the client is unaware of the attributes, they will be ignored. The result will be that the client uses the default "a=control" URLs at setup and receives the default alternatives.

EXAMPLE 7: v=0

o=ericsson\_user 1 1 IN IP4 130.240.188.69

s=A basic audio and video presentation

c=IN IP4 0.0.0.0

b=AS:325

b=TIAS: 308500

a=maxprate:50

a=control:\*

a=range:npt=0-150.2

a=alt-group:BW:AS:222=1,3;325=1,4;329=2,4;413=2,5

a=alt-group:BW:TIAS:208500\_40=1,3;308500\_50=1,4;312680\_50=2,4;396680\_50=2,5

t=0 0

m=audio 0 RTP/AVP 97

b=AS:12

b=TIAS:8500

a=maxprate:10

a=rtpmap:97 AMR/8000

a=control:trackID=1

a=fmtp:97 octet-align=1

a=range:npt=0-150.2

a=alt-default-id:1

a=alt:2:b=AS:16

a=alt:2:b=TIAS:12680

a=alt:2:a=control:trackID=2

m=video 0 RTP/AVP 98

b=AS:313

b=TIAS:3000000

a=maxprate:40

a=rtpmap:98 H264/90000

a=control:trackID=4

a=fmtp:98 profile-level-id=42c00c; sprop-parameter-sets= Z0KADJWgUH6Af1A=,aM46gA==

a=range:npt=0-150.2

a=X-initpredecbufperiod:98000

a=alt-default-id:4

a=alt:3:b=AS:210

a=alt:3:b=TIAS:200000

a=alt:3:a=maxprate:30

a=alt:3:a=control:trackID=3

a=alt:3:a=X-initpredecbufperiod:48000

a=alt:5:b=AS:397

a=alt:5:b=TIAS:384000

a=alt:5:a=maxprate:40

a=alt:5:a=control:trackID=5

a=alt:5:a=X-initpredecbufperiod:150000

The above example has 5 alternatives, 2 for audio and 3 for video. That would allow for a total of six combinations between audio and video. However, the grouping attribute in this example recommends that only 4 of these combinations be used. The equivalent SDP for the default alternatives (alternatives 1 and 4) with a total session bitrate of 325 kbps follows:

EXAMPLE 8: v=0

o=ericsson\_user 1 1 IN IP4 130.240.188.69

s=Ericsson commercial

c=IN IP4 0.0.0.0

b=AS:325

b=TIAS: 308500

a=maxprate:50

a=control:\*

a=range:npt=0-150.2

t=0 0

m=audio 0 RTP/AVP 97

b=AS:12

b=TIAS:8500

a=maxprate:10

a=rtpmap:97 AMR/8000

a=control:trackID=1

a=fmtp:97 octet-align=1

a=range:npt=0-150.2

m=video 0 RTP/AVP 98

b=AS:313

b=TIAS:300000

a=maxprate:40

a=rtpmap:98 H264/90000

a=control:trackID=4

a=fmtp:98 profile-level-id=42c00c; sprop-parameter-sets = Z0KADJWgUH6Af1A=,aM46gA==

a=range:npt=0-150.2

a=X-initpredecbufperiod:98000

The equivalent SDP for the 222 kbps total session bitrate (alternatives 1 and 3) is:

EXAMPLE 9: v=0

o=ericsson\_user 1 1 IN IP4 130.240.188.69

s=A basic audio and video presentation

c=IN IP4 0.0.0.0

b=AS:222

b=TIAS:208500

a=maxprate:40

a=control:\*

a=range:npt=0-150.2

t=0 0

m=audio 0 RTP/AVP 97

b=AS:12

b=TIAS:8500

a=maxprate:10

a=rtpmap:97 AMR/8000

a=control:trackID=1

a=fmtp:97 octet-align=1

a=range:npt=0-150.2

m=video 0 RTP/AVP 98

b=AS:210

b=TIAS:200000

a=maxprate:30

a=rtpmap:98 H264/90000

a=control:trackID=3

a=fmtp:98 profile-level-id=42c00c; sprop-parameter-sets = Z0KADJWgUH6Af1A=,aM46gA==

a=range:npt=0-150.2

a=X-initpredecbufperiod:48000

The equivalent SDP for the grouping with a 413 kbps total session bandwidth (alternatives 2 and 5):

EXAMPLE 10: v=0

o=ericsson\_user 1 1 IN IP4 130.240.188.69

s=A basic audio and video presentation

c=IN IP4 0.0.0.0

b=AS:413

b=TIAS: 396680

a=maxprate:50

a=control:\*

a=range:npt=0-150.2

t=0 0

m=audio 0 RTP/AVP 97

b=AS:16

b=TIAS:12680

a=maxprate:10

a=rtpmap:97 AMR/8000

a=control:trackID=2

a=fmtp:97 octet-align=1

a=range:npt=0-150.2

m=video 0 RTP/AVP 98

b=AS:397

b=TIAS:384000

a=maxprate:40

a=rtpmap:98 H264/90000

a=control:trackID=5

a=fmtp:98 profile-level-id=42c00c; sprop-parameter-sets = Z0KADJWgUH6Af1A=,aM46gA==

a=range:npt=0-150.2

a=X-initpredecbufperiod:150000

If the client only has 250 kbps it selects the media alternatives 1 and 3, which use 222 kbps. The client sets this up by sending two normal RTSP requests using the control URLs from the chosen alternatives.

The audio SETUP request for the default (i.e. 325 kbps in the example above) looks like this:

EXAMPLE 11: SETUP rtsp://media.example.com/examples/3G\_systems.3gp/trackID=1 RTSP/1.0

CSeq: 2

Transport: RTP/AVP/UDP;unicast;client\_port=3456-3457

The response from the server would be:

EXAMPLE 12: RTSP/1.0 200 OK

CSeq: 2

Session: jEs.EdXCSKpB

Transport: RTP/AVP/UDP;unicast;client\_port=3456-3457;server\_port=4002-4003;ssrc=5199dcb1

Also the video is added to the RTSP session under aggregated control:

EXAMPLE 13: SETUP rtsp://media.example.com/examples/3G\_systems.3gp/trackID=3 RTSP/1.0

CSeq: 3

Transport: RTP/AVP/UDP;unicast;client\_port=3458-3459

Session: jEs.EdXCSKpB

And the response would be:

EXAMPLE 14: RTSP/1.0 200 OK

CSeq: 3

Session: jEs.EdXCSKpB

Transport: RTP/AVP/UDP;unicast;client\_port=3458-3459;server\_port=4004-4005;ssrc=ae75904f

Had the client had more available bandwidth it could have set up another pair of alternatives in order to get better quality. The only change had been the RTSP URLs that had pointed at other media streams. For example the 413 kbps version would have been received if the audio SETUP request had used:

EXAMPLE 15: rtsp://media.example.com/examples/3G\_systems.3gp/trackID=2

and the video request

EXAMPLE 16: rtsp://media.example.com/examples/3G\_systems.3gp/trackID=5

The following example shows an SDP file that contains asset information, defined in Clause 5.3.3.7.

EXAMPLE 17: v=0  
o=ghost 2890844526 2890842807 IN IP4 192.168.10.10  
s=3GPP Unicast SDP Example  
i=Example of Unicast SDP file  
u=http://www.infoserver.com/ae600  
e=ghost@mailserver.com  
c=IN IP4 0.0.0.0  
t=0 0

a=range:npt=0-45.678

a=3GPP-Asset-Information: {url="http://www.movie-database.com/title/thismovieinfo.xhtml"}

a=3GPP-Asset-Information: {Title=MjhDRTA2NzI},{Copyright=Mjc0MkUwMUVGNDE2}  
m=video 1024 RTP/AVP 96

b=AS:1030

b=TIAS:1000000

a=maxprate:90  
a=rtpmap:96 H264/90000  
a=fmtp:96 profile-level-id=64001e; sprop-parameter-sets=Z2QAHpWQC0PaAfyQ,aOuOoA==  
a=control:rtsp://mediaserver.com/movie.3gp/trackID=1  
a=framesize:96 176-144

The following example shows the SDP media lines for AMR-WB+ Audio according to [85]

EXAMPLE 18:

m=audio 49120 RTP/AVP 99

a=rtpmap:99 AMR-WB+/72000/2

a=fmtp:99 interleaving=30; int-delay=86400

a=maxptime:100

The following example shows the SDP media lines for HE-AAC 48kHz, stereo (64kbps) using RFC6416 [13]

EXAMPLE 19:

m=audio 49230 RTP/AVP 96

a=rtpmap:96 MP4A-LATM/48000/2

a=fmtp:96 profile-level-id=44; bitrate=64000; cpresent=0; config=40005623101fe0; \  
SBR-enabled=1

### A.3.2.1 Maximum RTP packet size

The RFC 3550 (RTP) [9] does not impose a maximum size on RTP packets. However, when RTP packets are sent over the radio link of a 3GPP PSS system there is an advantage in limiting the maximum size of RTP packets.

Two types of bearers can be envisioned for streaming using either acknowledged mode (AM) or unacknowledged mode (UM) RLC. The AM uses retransmissions over the radio link whereas the UM does not. In UM mode large RTP packets are more susceptible to losses over the radio link compared to small RTP packets since the loss of a segment may result in the loss of the whole packet. On the other hand in AM mode large RTP packets will result in larger delay jitter compared to small packets as there is a larger chance that more segments have to be retransmitted.

For these reasons it is recommended that the maximum size of RTP packets should be limited in size taking into account the wireless link. This will decrease the RTP packet loss rate particularly for RLC in UM. For RLC in AM the delay jitter will be reduced permitting the client to use a smaller receiving buffer. It should also be noted that too small RTP packets could result in too much overhead if IP/UDP/RTP header compression is not applied or unnecessary load at the streaming server.

In the case of transporting video in the payload of RTP packets it may be that a video frame is split into more than one RTP packet in order not to produce too large RTP packets. Then, to be able to decode packets following a lost packet in the same video frame, it is recommended that synchronisation information be inserted at the start of such RTP packets.

## A.4.7 Example of a PSS device capability description

The following is an example of a device capability profile as it could be available from a device profile server. The XML document includes the description of the imaginary "Phone007" phone.

Instead of a single XML document the description could also be spread over several files. The PSS server would need to retrieve these profiles separately in this case and would need to merge them. For instance, this would be useful when device capabilities of this phone that are related to streaming would differ among different versions of the phone. In this case the part of the profile for streaming would be separated from the rest into its own profile document. This separation allows describing the difference in streaming capabilities by providing multiple versions of the profile document for the streaming capabilities.

<?xml version="1.0"?>

<rdf:RDF xmlns:rdf="http://www.w3.org/1999/02/22-rdf-syntax-ns#"

xmlns:ccpp="http://www.w3.org/2002/11/08-ccpp-ns#"

xmlns:prf="http://www.wapforum.org/profiles/UAPROF/ccppschema-20070511#"

xmlns:pss6="http://www.3gpp.org/profiles/PSS/ccppschema-PSS6#">

<rdf:Description rdf:about="http://www.bar.com/Phones/Phone007">

<ccpp:component>

<rdf:Description rdf:ID="HardwarePlatform">

<rdf:type rdf:resource="http://www.wapforum.org/profiles/UAPROF/ccppschema-20070511#HardwarePlatform" />

<prf:BitsPerPixel>4</prf:BitsPerPixel>

<prf:ColorCapable>Yes</prf:ColorCapable>

<prf:PixelAspectRatio>1x2</prf:PixelAspectRatio>

<prf:PointingResolution>Pixel</prf:PointingResolution>

<prf:Model>Phone007</prf:Model>

<prf:Vendor>Ericsson</prf:Vendor>

</rdf:Description>

</ccpp:component>

<ccpp:component>

<rdf:Description rdf:ID="SoftwarePlatform">

<rdf:type rdf:resource="http://www.wapforum.org/profiles/UAPROF/ccppschema-20070511#SoftwarePlatform" />

<prf:CcppAccept-Charset>

<rdf:Bag>

<rdf:li>UTF-8</rdf:li>

<rdf:li>ISO-10646-UCS-2</rdf:li>

</rdf:Bag>

</prf:CcppAccept-Charset>

<prf:CcppAccept-Encoding>

<rdf:Bag>

<rdf:li>base64</rdf:li>

<rdf:li>quoted-printable</rdf:li>

</rdf:Bag>

</prf:CcppAccept-Encoding>

<prf:CcppAccept-Language>

<rdf:Seq>

<rdf:li>en</rdf:li>

<rdf:li>se</rdf:li>

</rdf:Seq>

</prf:CcppAccept-Language>

<prf:DMCapable>Yes</prf:DMCapable>

<prf:DMVersion>1.2</prf:DMVersion>

</rdf:Description>

</ccpp:component>

<ccpp:component>

<rdf:Description rdf:ID="PssCommon">

<rdf:type rdf:resource="http://www.3gpp.org/profiles/PSS/ccppschema-PSS6#PssCommon" />

<pss6:AudioChannels>Stereo</pss6:AudioChannels>

<pss6:MaxPolyphony>24</pss6:MaxPolyphony>

<pss6:PssVersion>3GPP-R6</pss6:PssVersion>

<pss6:RenderingScreenSize>160x120</pss6:RenderingScreenSize>

</rdf:Description>

</ccpp:component>

<ccpp:component>

<rdf:Description rdf:ID="Streaming">

<rdf:type rdf:resource="http://www.3gpp.org/profiles/PSS/ccppschema-PSS6#Streaming" />

<pss6:ThreeGPPLinkChar>Yes</pss6:ThreeGPPLinkChar>

<pss6:AdaptationSupport>Yes</pss6:AdaptationSupport>

<pss6:ExtendedRtcpReports>Yes</pss6:ExtendedRtcpReports>

<pss6:MediaAlternatives>Yes</pss6:MediaAlternatives>

<pss6:RtpProfiles>

<rdf:Bag>

<rdf:li>RTP/AVP</rdf:li>

<rdf:li>RTP/AVPF</rdf:li>

</rdf:Bag>

</pss6:RtpProfiles>

<pss6:VideoPreDecoderBufferSize>30720</pss6:VideoPreDecoderBufferSize>

<pss6:VideoInitialPostDecoderBufferingPeriod>0</pss6:VideoInitialPostDecoderBufferingPeriod>

<pss6:VideoDecodingByteRate>16000</pss6:VideoDecodingByteRate>

<pss6:StreamingAccept>

<rdf:Bag>

<rdf:li>audio/AMR</rdf:li>

<rdf:li>video/H264; profile-level-id=42e00a</rdf:li>

</rdf:Bag>

</pss6:StreamingAccept>

</rdf:Description>

</ccpp:component>

<ccpp:component>

<rdf:Description rdf:ID="ThreeGPFileFormat">

<rdf:type rdf:resource="http://www.3gpp.org/profiles/PSS/ccppschema-PSS6#ThreeGPFileFormat" />

<pss6:Brands>

<rdf:Bag>

<rdf:li>3gp4</rdf:li>

<rdf:li>3gp5</rdf:li>

<rdf:li>3gp6</rdf:li>

<rdf:li>3gr6</rdf:li>

</rdf:Bag>

</pss6:Brands>

<pss6:ThreeGPAccept>

<rdf:Bag>

<rdf:li>audio/AMR</rdf:li>

<rdf:li>audio/AMR-WB;octet-alignment=1</rdf:li>

<rdf:li>video/H264; profile-level-id=42e00a</rdf:li>

<rdf:li>video/Timed-Text</rdf:li>

</rdf:Bag>

</pss6:ThreeGPAccept>

</rdf:Description>

</ccpp:component>

Annex F (normative):  
RDF schema for the PSS base vocabulary

<?xml version="1.0"?>

<!--

This document is the RDF Schema for Packet-switched Streaming

Service (PSS)-specific vocabulary as defined in 3GPP TS 26.234

Release 12 (in the following "the specification").

The URI for unique identification of this RDF Schema is

http://www.3gpp.org/profiles/PSS/ccppschema-PSS12#

This RDF Schema includes the same information as the respective

chapter of the specification. Greatest care has been taken to keep

the two documents consistence. However, in case of any divergence

the specification takes presidence.

All reference in this RDF Schmea are to be interpreted relative to

the specification. This means all references using the form

[ref] are defined in chapter 2 "References" of the specification.

All other references refer to parts within that document.

Note: This [Schemas has been aligned in structure and base](https://qualcomm-my.sharepoint.com/personal/tsto_qti_qualcomm_com/Documents/Standards/3GPP/SA4/TSGS4_109-e/Own Contributions/Video/Schemas has been aligned in structure and base%0d)

vocabulary to the RDF Schema used by UAProf [40].

-->

<rdf:RDF xmlns:rdf="http://www.w3.org/1999/02/22-rdf-syntax-ns#"

xmlns:rdfs="http://www.w3.org/2000/01/rdf-schema#" >

<!-- \*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\* -->

<!-- \*\*\*\*\* Properties shared among the components\*\*\*\*\* -->

<rdf:Description rdf:ID="defaults">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#PssCommon"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:domain rdf:resource="#ThreeGPFileFormat"/>

<rdfs:domain rdf:resource="#PssSmil"/>

<rdfs:comment>

An attribute used to identify the default capabilities.

</rdfs:comment>

</rdf:Description>

<!-- \*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\* -->

<!-- \*\*\*\*\* Component Definitions \*\*\*\*\* -->

<rdf:Description rdf:ID="PssCommon">

<rdf:type rdf:resource="http://www.w3.org/2000/01/rdf-schema#Class"/>

<rdfs:subClassOf rdf:resource="http://www.wapforum.org/profiles/UAPROF/ccppschema-20070511#Component"/>

<rdfs:label>Component: PssCommon</rdfs:label>

<rdfs:comment>

The PssCommon component specifies the base vocabulary common for all

PSS applications, in contrast to application-specific parts of the PSS

base vocabulary which are described by the Streaming, ThreeGPFileFormat and

PssSmil components defined below.

PSS servers supporting capability exchange should understand the attributes

in this component as explained in detail in 3GPP TS 26.234 Release 7.

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="Streaming">

<rdf:type rdf:resource="http://www.w3.org/2000/01/rdf-schema#Class"/>

<rdfs:subClassOf rdf:resource="http://www.wapforum.org/profiles/UAPROF/ccppschema-20070511#Component"/>

<rdfs:label>Component: Streaming</rdfs:label>

<rdfs:comment>

The Streaming component specifies the base vocabulary for pure RTSP/RTP-

based streaming in PSS.

PSS servers supporting capability exchange should understand the attributes

in this component as explained in detail in 3GPP TS 26.234 Release 7.

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="ThreeGPFileFormat">

<rdf:type rdf:resource="http://www.w3.org/2000/01/rdf-schema#Class"/>

<rdfs:subClassOf rdf:resource="http://www.wapforum.org/profiles/UAPROF/ccppschema-20070511#Component"/>

<rdfs:label>Component: ThreeGPFileFormat</rdfs:label>

<rdfs:comment>

The ThreeGPFileFormat component specifies the base vocabulary for 3GP file

download or progressive download in PSS.

PSS servers supporting capability exchange should understand the attributes

in this component as explained in detail in 3GPP TS 26.234 Release 7.

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="PssSmil">

<rdf:type rdf:resource="http://www.w3.org/2000/01/rdf-schema#Class"/>

<rdfs:subClassOf rdf:resource="http://www.wapforum.org/profiles/UAPROF/ccppschema-20070511#Component"/>

<rdfs:label>Component: PssSmil</rdfs:label>

<rdfs:comment>

The PssSmil component specifies the base vocabulary for SMIL presentations

in PSS. Note that capabibilites regarding streaming and 3GP files that are

part of a SMIL presentation are expressed by the vocabularies specified by

the Streaming and ThreeGPFileFormat components, respectively.

PSS servers supporting capability exchange should understand the attributes

in this component as explained in detail in 3GPP TS 26.234 Release 7.

</rdfs:comment>

</rdf:Description>

<!-- \*\*

\*\* In the following property definitions, the defined types

\*\* are as follows:

\*\*

\*\* Number: A positive integer

\*\* [0-9]+

\*\* Boolean: A yes or no value

\*\* Yes|No

\*\* Literal: An alphanumeric string

\*\* [A-Za-z0-9/.\-\_]+

\*\* Dimension: A pair of numbers

\*\* [0-9]+x[0-9]+

\*\*

-->

<!-- \*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\* -->

<!-- \*\*\*\*\* Component: PssCommon \*\*\*\*\* -->

<rdf:Description rdf:ID="AudioChannels">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#PssCommon"/>

<rdfs:comment>

Description: This attribute describes the stereophonic capability of the

natural audio device. The only legal values are "Mono" and "Stereo".

Type: Literal

Resolution: Locked

Examples: "Mono", "Stereo"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="MaxPolyphony">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#PssCommon"/>

<rdfs:comment>

Description: The MaxPolyphony attribute refers to the maximal polyphony

that the synthetic audio device supports as defined in [44]. Legal values

are integer between 5 to 24.

NOTE: MaxPolyphony attribute can be used to signal the maximum polyphony

capabilities supported by the PSS client. This is a complementary

mechanism for the delivery of compatible SP-MIDI content and thus

the PSS client is required to support Scalable Polyphony MIDI i.e.

Channel Masking defined in [44].

Type: Number

Resolution: Locked

Examples: 8

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="NumOfGM1Voices">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#PssCommon"/>

<rdfs:comment>

Description: The NumOfGM1Voices attribute refers to the maximum number

of simultaneous GM1 voices that the synthetic audio engine supports.

Legal values are integers greater or equal than 5.

Type: Number

Resolution: Locked

Examples: 24

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="NumOfMobileDLSVoicesWithoutOptionalBlocks">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#PssCommon"/>

<rdfs:comment>

Description: The NumOfMobileDLSVoicesWithoutOptionalBlocks attribute

refers to the maximum number of simultaneous voices without optional

group of processing blocks that the synthetic audio engine supports.

Legal values are integers greater or equal than 5.

Type: Number

Resolution: Locked

Examples: 24

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="NumOfMobileDLSVoicesWithOptionalBlocks">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#PssCommon"/>

<rdfs:comment>

Description: The NumOfMobileDLSVoicesWithOptionalBlocks attribute refers

to the maximum number of simultaneous voices with optional group of

processing blocks that the synthetic audio engine supports. This attribute

is set to zero for devices that do not support the optional group of

processing blocks. Legal values are integers greater or equal than 0.

Type: Number

Resolution: Locked

Examples: 24

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="PssVersion">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#PssCommon"/>

<rdfs:comment>

Description: Latest PSS version supported by the client. Legal

values are "3GPP-R4", "3GPP-R5", "3GPP-R6", "3GPP-R7" and so forth.

Type: Literal

Resolution: Locked

Examples: "3GPP-R5", "3GPP-R6"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="RenderingScreenSize">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#PssCommon"/>

<rdfs:comment>

Description: The rendering size of the device's screen in unit of

pixels available for PSS media presentation. The horizontal size is

given followed by the vertical size. Legal values are pairs of integer

values equal or greater than zero. A value equal "0x0"means that there

exists no display or just textual output is supported.

Type: Dimension

Resolution: Locked

Examples: "160x120"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID=" **RenderingScreenSizeMm**">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#PssCommon"/>

<rdfs:comment>

Description: The rendering size of the device's screen in unit of

millimiters available for PSS media presentation. The horizontal size is

given followed by the vertical size. Legal values are pairs of floating

values equal or greater than zero. A value equal "0.0x0.0"means that there

exists no possibility to render visual PSS media presentation.

Type: Dimension

Resolution: Locked

Examples: "110.5x56.0"

</rdfs:comment>

</rdf:Description>

<!-- \*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\* -->

<!-- \*\*\*\*\* Component: Streaming \*\*\*\*\* -->

<rdf:Description rdf:ID="StreamingMethod">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: List of streaming methods supported by the PSS application. The client may support RTP streaming, HTTP streaming, or both.

Type: Literal (bag)

Resolution: Append

Examples: "RTP,HTTP"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="StreamingAccept">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: List of content types (MIME types) relevant for streaming

over RTP supported by the PSS application. Content types listed shall be

possible to stream over RTP. For each content type a set of MIME parameters  
 can be specified to signal receiver capabilities. A content type that

supports multiple parameter sets may occur several times in the list.

Legal values are lists of MIME types with related parameters.

Type: Literal (bag)

Resolution: Append

Examples: "audio/AMR-WB;octet-alignment=1,application/smil"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="StreamingAccept-Subset">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: List of content types for which the PSS application supports

a subset. MIME types can in most cases effectively be used to express

variations in support for different media types. Many MIME types, e.g.

AMR-WB has several parameters that can be used for this purpose. There

may exist content types for which the PSS application only supports a

subset and this subset cannot be expressed with MIME-type parameters.

In these cases the attribute StreamingAccept-Subset is used to describe

support for a subset of a specific content type. If a subset of a specific

content type is declared in StreamingAccept-Subset, this means that

StreamingAccept-Subset has precedence over StreamingAccept.

StreamingAccept shall always include the corresponding content types for

which StreamingAccept-Subset specifies subsets of.

No legal values are currently defined.

Type: Literal (bag)

Resolution: Locked

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="StreamingFramePackingFormatsRTP">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: List of supported frame packing formats relevant for streaming of stereoscopic 3D video over RTP supported by the PSS application. The frame packing formats within scope for stereoscopic 3D video are defined in Table D-8 of [90].

Type: Literal (bag)

Resolution: Append

Examples: "3,4"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="StreamingFramePackingFormatsHTTP">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: List of supported frame packing formats relevant for streaming of stereoscopic 3D video over HTTP supported by the PSS application. The frame packing formats within scope for stereoscopic 3D video are defined in Table D-8 of [90].

Type: Literal (bag)

Resolution: Append

Examples: "3,4"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="StreamingCVOCapable">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: Indicates whether the client is a CVO capable receiver of RTP streams, i.e. provided that the video orientation information for the delivered content is communicated to the client in an RTP extension header as specified in clause 6.2.5 (corresponding to urn:3gpp:video-orientation), the client can interpret the video orientation and align the video correctly for rendering/display purposes. If this attribute is reported and the StreamingHighGranularityCVOCapable attribute is reported as a "Yes", then the value of this attribute shall be a "Yes".

Type: Literal

Resolution: Locked

Examples: "Yes"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="StreamingHighGranularityCVOCapable">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: Indicates whether the client is a Higher Granularity CVO capable receiver of RTP streams, i.e. provided that the video orientation information of the delivered content is communicated to the client in an RTP extension header as specified in clause 6.2.5 (corresponding to urn:3GPP:video-orientation:6**)**, the client can interpret the video orientation and align the video correctly for rendering/display purposes.

Type: Literal

Resolution: Locked

Examples: "Yes"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="LinkChar">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: This attribute indicates whether the device supports the

3GPP-Link-Char header according to clause 10.2.1.1 of the specification.

Legal values are "Yes" and "No".

Type: Literal

Resolution: Override

Examples: "Yes"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="AdaptationSupport">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: This attribute indicates whether the device supports

client buffer feedback signaling according to clause 10.2.3 of the

specification. Legal values are "Yes" and "No".

Type: Literal

Resolution: Locked

Examples: "Yes"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="QoESupport">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: This attribute indicates whether the device supports

QoE signaling according to clauses 5.3.2.3, 5.3.3.6, and 11 of the

specification. Legal values are "Yes" and "No".

Type: Literal

Resolution: Locked

Examples: "Yes"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="ExtendedRtcpReports">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: This attribute indicates whether the device supports

extended RTCP reports according to clause 6.2.3.1 of the specification.

Legal values are "Yes" and "No".

Type: Literal

Resolution: Locked

Examples: "Yes"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="RtpRetransmission">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: This attribute indicates whether the device supports RTP

retransmission according to clause 6.2.3.3 of the specification.

Legal values are "Yes" and "No".

Type: Literal

Resolution: Locked

Examples: "Yes"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="MediaAlternatives">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: This attribute indicates whether the device interprets the

SDP attributes "alt", "alt-default-id", and "alt-group", defined in

clauses 5.3.3.3 and 5.3.3.4 of the specification.

Legal values are "Yes" and "No".

Type: Literal

Resolution: Override

Examples: "Yes"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="RtpProfiles">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: This attribute lists the supported RTP profiles. Legal

values are profile names registered through the Internet Assigned Numbers

Authority (IANA), www.iana.org.

Type: Literal (bag)

Resolution: Append

Examples: "RTP/AVP,RTP/AVPF"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="ProtectedStreaming">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: Indicates whether the device protection

for streamed content as defined by Annex R. Legal values are "Yes" and

"No".

Type: Literal

Resolution: Locked

Examples: "Yes"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="3GPPPipelined">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: This attribute indicates whether the device supports fast content start-up with pipelining according to clause 5.5.3.

Legal values are "Yes" and "No".

Type: Literal

Resolution: Locked

Examples: "Yes"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="3GPPSwitch">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: This attribute indicates whether the device supports fast content switching with known SDP according to clause 5.5.4.3..

Legal values are "Yes" and "No".

Type: Literal

Resolution: Locked

Examples: "Yes"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="3GPPSwitchReqSDP">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: This attribute indicates whether the device supports fast content switching without SDP according to clause 5.5.4.4.

Legal values are "Yes" and "No".

Type: Literal

Resolution: Locked

Examples: "Yes"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="3GPPSwitchStream">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: This attribute indicates whether the device supports the fast switching of media streams according to clause 5.5.4.5.

Legal values are "Yes" and "No".

Type: Literal

Resolution: Locked

Examples: "Yes"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="AcceptRanges">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: List of range indications that are accepted by the client. The client may support UTC or NPT or both.

Type: Literal (bag)

Resolution: Append

Examples: "NPT,UTC"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="ISMACryp">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: Indicates whether the device supports streamed protected

content in ISMACryp format, as defined by ISMACryp and Annex R. Legal values are ISMACryp

Version numbers supported as a floating number. 0.0 indicates no support.

Type: Literal (bag)

Resolution: Locked

Examples: "2.0"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="VideoDecodingByteRate">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: If Annex G is not supported, the attribute has no meaning.

If Annex G is supported, this attribute defines the peak decoding byte

rate the PSS client is able to support. In other words, the PSS client

fulfils the requirements given in Annex G with the signalled peak decoding

byte rate. The values are given in bytes per second and shall be greater

than or equal to 16000.

Type: Number

Resolution: Locked

Examples: "16000"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="VideoInitialPostDecoderBufferingPeriod">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: If Annex G is not supported, the attribute has no

meaning. If Annex G is supported, this attribute defines the

maximum initial post-decoder buffering period of video. Values are

interpreted as clock ticks of a 90-kHz clock. In other words, the

value is incremented by one for each 1/90 000 seconds. For

example, the value 9000 corresponds to 1/10 of a second initial

post-decodder buffering. Legal values are all integer values equal

to or greater than zero.

Type: Number

Resolution: Locked

Examples: "9000"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="VideoPreDecoderBufferSize">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#Streaming"/>

<rdfs:comment>

Description: This attribute signals if the optional video

buffering requirements defined in Annex G are supported. It also

defines the size of the hypothetical pre-decoder buffer defined in

Annex G. A value equal to zero means that Annex G is not

supported. A value equal to one means that Annex G is

supported. In this case the size of the buffer is the default size

defined in Annex G. A value equal to or greater than the default

buffer size defined in Annex G means that Annex G is supported and

sets the buffer size to the given number of octets. Legal values are all

integer values equal to or greater than zero. Values greater than

one but less than the default buffer size defined in Annex G are

not allowed.

Type: Number

Resolution: Locked

Examples: "0", "4096"

</rdfs:comment>

</rdf:Description>

<!-- \*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\* -->

<!-- \*\*\*\*\* Component: ThreeGPFileFormat \*\*\*\*\* -->

<rdf:Description rdf:ID="Brands">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#ThreeGPFileFormat"/>

<rdfs:comment>

Description: This attribute lists the supported 3GP profiles identified

by brand. Legal values are brand identifiers according to 5.3.4 and 5.4

in [50].

Type: Literal (bag)

Resolution: Append

Examples: "3gp4,3gp5,3gp6,3gr6,3gp7,3gr7,3ge7"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="ThreeGPAccept">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#ThreeGPFileFormat"/>

<rdfs:comment>

Description: List of content types (MIME types) that can be included

in a 3GP file and handled by the PSS application. If the identifier

"Streaming-Media" is included, streaming media can be included in the

presentation. Details on the streaming support can then be

found in the Streaming component. For each content

type a set of supported parameters can be given. A content type that

supports multiple parameter sets may occur several times in the list.

Type: Literal (bag)

Resolution: Append

Examples: "video/H264; profile-level-id=42e00a,audio/AMR"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="ThreeGPAccept-Subset">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#ThreeGPFileFormat"/>

<rdfs:comment>

Description: List of content types for which the PSS application

supports a subset. MIME types can in most cases effectively be used

to express variations in support for different media types. Many MIME

types have several parameters that can be used for this purpose. There

may exist content types for which the PSS application only supports a

subset and this subset cannot be expressed with MIME type parameters.

In these cases the attribute ThreeGPAccept-Subset is used to describe

support for a subset of a specific content type. If a subset of a

specific content type is declared in ThreeGPAccept-Subset, this means that

ThreeGPAccept-Subset has precedence over ThreeGPAccept. ThreeGPAccept shall always

include the corresponding content types for which ThreeGPAccept-Subset

specifies subsets of. No legal values are currently defined.

Type: Literal (bag)

Resolution: Locked

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="ThreeGPFramePackingFormats">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#ThreeGPFileFormat"/>

<rdfs:comment>

Description: List of supported frame packing formats relevant for stereoscopic 3D video that can be included in a 3GP file and handled by the PSS application.

Type: Literal (bag)

Resolution: Append

Examples: "3,4"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="ThreeGPCVOCapable">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#ThreeGPFileFormat"/>

<rdfs:comment>

Description: Indicates whether the client is a CVO capable receiver of 3GP files, i.e. provided that the video orientation information (corresponding to urn:3gpp:video-orientation) of the delivered content is communicated to the client in a 3GP file, the client can interpret the video orientation and align the video correctly for rendering/display purposes. If this attribute is reported and the ThreeGPHighGranularityCVOCapable attribute is reported as a "Yes", then the value of this attribute shall be a "Yes".

Type: Literal

Resolution: Locked

Examples: "Yes"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="ThreeGPHighGranularityCVOCapable">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#ThreeGPFileFormat"/>

<rdfs:comment>

Description: Indicates whether the client is a Higher Granularity CVO capable receiver of 3GP files, i.e. provided that the video orientation information (corresponding to urn:3gpp:video-orientation:6) of the delivered content is communicated to the client in a 3GP file, the client can interpret the video orientation and align the video correctly for rendering/display purposes.

Type: Literal

Resolution: Locked

Examples: "Yes"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="ThreeGPOmaDrm">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#ThreeGPFileFormat"/>

<rdfs:comment>

Description: List of the OMA DRM versions that is supported to be used

for DRM protection of content present in the 3GP file format. Legal values

are OMA DRM version numbers as floating values. 0.0 indicates no support.

Type: Literal (bag)

Resolution: Locked

Examples: "2.0"

</rdfs:comment>

</rdf:Description>

<!-- \*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\*\* -->

<!-- \*\*\*\*\* Component: PssSmil \*\*\*\*\* -->

<rdf:Description rdf:ID="SmilAccept">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#PssSmil"/>

<rdfs:comment>

Description: List of content types (MIME types) that can be part of a

SMIL presentation. The content types included in this attribute can be

rendered in a SMIL presentation. If video/3gpp (or audio/3gpp) is

included, downloaded 3GP files can be included in a SMIL presentation.

Details on the 3GP file support can then be found in the ThreeGPFileFormat

component. If the identifier "Streaming-Media" is included, streaming

media can be included in the SMIL presentation. Details on the

streaming support can then be found in the Streaming component.

For each content type a set of supported parameters can be given.

A content type that supports multiple parameter sets may occur several

times in the list. Legal values are lists of MIME types with related

parameters and the "Streaming-Media" identifier.

Type: Literal (bag)

Resolution: Append

Examples: "image/gif,image/jpeg,Streaming-Media"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="SmilAccept-Subset">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#PssSmil"/>

<rdfs:comment>

Description: List of content types for which the PSS application

supports a subset. MIME types can in most cases effectively be used to

express variations in support for different media types. Many MIME types

have several parameters that can be used for this purpose. There may

exist content types for which the PSS application only supports a subset

and this subset cannot be expressed with MIME-type parameters. In these

cases the attribute SmilAccept-Subset is used to describe support for a

subset of a specific content type. If a subset of a specific content type

is declared in SmilAccept-Subset, this means that SmilAccept-Subset has

precedence over SmilAccept. SmilAccept shall always include the

corresponding content types for which SmilAccept-Subset specifies subsets

of.

The following values are defined:

- "JPEG-PSS": Only the two JPEG modes described in clause 7.5 of the

specifictaion are supported.

- "SVG-Tiny"

- "SVG-Basic"

Subset identifiers and corresponding semantics shall only be defined by

the TSG responsible for the present document.

Type: Literal (bag)

Resolution: Append

Examples: "JPEG-PSS,SVG-Tiny"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="SmilBaseSet">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:domain rdf:resource="#PssSmil"/>

<rdfs:comment>

Description: Indicates a base set of SMIL 2.0 modules that the client

supports. Leagal values are the following pre-defined identifiers:

"SMIL-3GPP-R4" and "SMIL-3GPP-R5" indicate all SMIL 2.0 modules required

for SMIL scene-description support according to clause 8 of Release 4 and

Release 5, respectively, of TS 26.234. "SMIL-3GPP-R6" and "SMIL-3GPP-R7"

indicate all SMIL 2.0 modules required for SMIL scene description support

according to Release 6 and Release 7, respectively, of clause 8 of the

specification and of TS 26.246 [52].

Type: Literal

Resolution: Locked

Examples: "SMIL-3GPP-R4", "SMIL-3GPP-R5"

</rdfs:comment>

</rdf:Description>

<rdf:Description rdf:ID="SmilModules">

<rdf:type rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Property"/>

<rdfs:range rdf:resource="http://www.w3.org/1999/02/22-rdf-syntax-ns#Bag"/>

<rdfs:domain rdf:resource="#PssSmil"/>

<rdfs:comment>

Description: This attribute defines a list of SMIL 2.0 modules

supported by the client. If the SmilBaseSet is used those modules

do not need to be explicitly listed here. In that case only

additional module support needs to be listed. Legal values are all

SMIL 2.0 module names defined in the SMIL 2.0 recommendation [31],

section 2.3.3, table 2.

Type: Literal (bag)

Resolution: Locked

Examples: "BasicTransitions,MulitArcTiming"

</rdfs:comment>

</rdf:Description>

</rdf:RDF>

# G.2 PSS Buffering Parameters

The behaviour of the PSS buffering model is controlled with the following parameters: the initial pre-decoder buffering period, the initial post-decoder buffering period, the size of the hypothetical pre-decoder buffer, the peak decoding byte rate, and the decoding macroblock rate. The default values of the parameters are defined below.

- The default initial pre-decoder buffering period is 1 second.

- The default initial post-decoder buffering period is zero.

- The default size of the hypothetical pre-decoder buffer is defined according to the maximum video bit-rate according to the table below:

Table G.1: Default size of the hypothetical pre-decoder buffer

|  |  |
| --- | --- |
| Maximum video bit-rate | Default size of the hypothetical pre-decoder buffer |
| 65536 bits per second | 20480 bytes |
| 131072 bits per second | 40960 bytes |
| Undefined | 51200 bytes |

- The maximum video bit-rate can be signalled in the media-level bandwidth attribute of SDP as defined in clause 5.3.3 of this document. If the video-level bandwidth attribute was not present in the presentation description, the maximum video bit-rate is defined according to the video coding profile and level in use.

- The size of the hypothetical post-decoder buffer is an implementation-specific issue. The buffer size can be estimated from the maximum output data rate of the decoders in use and from the initial post-decoder buffering period.

- The default decoding macroblock rate is defined according to the video coding profile and level in use.

PSS clients may signal their capability of providing larger buffers and faster peak decoding byte rates in the capability exchange process described in clause 5.2 of the present document. The average coded video bit-rate should be smaller than or equal to the bit-rate indicated by the video coding profile and level in use, even if a faster peak decoding byte rate were signalled.

Initial parameter values for each stream can be signalled within the SDP description of the stream. Signalled parameter values override the corresponding default parameter values. The values signalled within the SDP description guarantee pauseless playback from the beginning of the stream until the end of the stream (assuming a constant-delay reliable transmission channel).

PSS servers may update parameter values in the response for an RTSP PLAY request. If an updated parameter value is present, it shall replace the value signalled in the SDP description or the default parameter value in the operation of the PSS buffering model. An updated parameter value is valid only in the indicated playback range, and it has no effect after that. Assuming a constant-delay reliable transmission channel, the updated parameter values guarantee pauseless playback of the actual range indicated in the response for the PLAY request. The indicated pre-decoder buffer size and initial post-decoder buffering period shall be smaller than or equal to the corresponding values in the SDP description or the corresponding default values, whichever ones are valid. The header fields for RTSP are specified in clause 5.3.2.4.

The following example plays the whole presentation starting at SMPTE time code 0:10:20 until the end of the clip. The playback is to start at 15:36 on 23 Jan 1997. The suggested initial pre-decoder buffering period is half a second.

C->S: PLAY rtsp://audio.example.com/twister.en RTSP/1.0

CSeq: 833

Session: 12345678

Range: smpte=0:10:20-;time=19970123T153600Z

User-Agent: TheStreamClient/1.1b2

S->C: RTSP/1.0 200 OK

CSeq: 833

Date: 23 Jan 1997 15:35:06 GMT

Range: smpte=0:10:22-;time=19970123T153600Z

x-initpredecbufperiod: 45000

### L.2.2.1 Inclusion of the video element in SVG content

The video element should be included within a "switch" element. The feature string for video could be

1. http://www.w3.org/TR/SVG12/feature#3GPPTransformedVideo

2. the feature string for video is http://www.w3.org/TR/SVG12/feature#3GPPVideo

3. or the alternate representation of a "video" element could be an image.

EXAMPLE:

<g transform="translate(10,0);scale(1.5)">  
 <switch>  
 <video  
 xlink:href="video.3gp"  
 type="video/H264; profile-level-id=42e00a"  
 requiredFeatures="http://www.w3.org/TR/SVG12/feature#TransformedVideo"/>  
 <video  
 xlink:href="video.3gp"  
 type="video/H264; profile-level-id=42e00a"  
 requiredFeatures="http://www.w3.org/TR/SVG12/feature#Video  
 transformBehavior ="pinned"/>  
 <image xlink:ref="image.jpg" width="176" height="144">  
 </switch>  
 </g>

The above example shows a transformed video. If the PSS client supports "TransformedVideo", the video shall be transformed, if not, a video-enabled PSS client shall display the video without scaling and rotation ("pinned"). Finally, an image shall be displayed if neither one of the above cases is possible at the PSS client.