Technical Specification Group Services and System Aspects ***TSGS#17(02)0439***
Meeting #17, Biarritz, France, 9-12 September 2002

Source: **TSG-SA WG4**

Title: **CRs to TS 26.234 - Corrections (Release 5)**

Document for: **Approval**

Agenda Item: **7.4.3**

The following CRs, agreed at the TSG-SA WG4 meeting #22, are presented to TSG SA #17 for approval.

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.234	030	2	Rel-5	Correction regarding support for Timed Text	F	5.1.0	S4	TSG-SA WG4#22	S4-020494
26.234	032	3	Rel-5	Required RTSP header support	F	5.1.0	S4	TSG-SA WG4#22	S4-020471
26.234	034	1	Rel-5	Including bitrate information for H.263	F	5.1.0	S4	TSG-SA WG4#22	S4-020490
26.234	035	1	Rel-5	RTCP Reports and Link Aliveness in Ready State	F	5.1.0	S4	TSG-SA WG4#22	S4-020489
26.234	036	2	Rel-5	Correction of media and session-level bandwidth fields in SDP	F	5.1.0	S4	TSG-SA WG4#22	S4-020491
26.234	037	2	Rel-5	Correction of usage of MIME parameters for AMR	F	5.1.0	S4	TSG-SA WG4#22	S4-020492
26.234	038	1	Rel-5	Correction of mapping of SDP parameters to UMTS QoS parameters (Annex J)	F	5.1.0	S4	TSG-SA WG4#22	S4-020470

3GPP TSG-SA4 Meeting #22
Tampere, Finland, 22-26 July 2002

Tdoc ⌘S4-020494

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CHANGE REQUEST

⌘	**26.234**	**CR**	**030**	⌘ **rev**	**2**	⌘ Current version:	**5.1.0** ⌘

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Proposed change affects:	UICC apps⌘ [ ]	ME [X]	Radio Access Network [ ]	Core Network [ ]

Title:	⌘	Correction regarding support for Timed Text		
Source:	⌘	TSG-SA WG4		
Work item code:	⌘	PSS-E	***Date:*** ⌘	12/Sept/2002
Category:	⌘	**F**	***Release:*** ⌘	Rel-5
		Use one of the following categories: ***F*** *(correction)* ***A*** *(corresponds to a correction 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.		*Use one of the following releases:* *2 (GSM Phase 2)* *R96 (Release 1996)* *R97 (Release 1997)* *R98 (Release 1998)* *R99 (Release 1999)* *Rel-4 (Release 4)* *Rel-5 (Release 5)* *Rel-6 (Release 6)*

Reason for change:	⌘	Misprint suggests that Timed Text is mandatory and optional in PSS clients.
Summary of change:	⌘	Clarified that support for Timed Text is optional. If supported, the definition given in Annex D shall be supported.
Consequences if not approved:	⌘	Contradiction in specification.

Clauses affected:	⌘	7.9				
		Y	**N**			
Other specs affected:	⌘		**N**	Other core specifications	⌘	
			N	Test specifications		
			N	O&M Specifications		
Other comments:	⌘					

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3) With "track changes" disabled, paste the entire CR form (use CTRL-A to select it) into the specification just in front of the clause containing the first piece of changed text. Delete those parts of the specification which are not relevant to the change request.

7.9 Timed text

If timed text is supported, PSS clients shall support ~~timed text as defined in~~ Annex D, clause D.8a, of this specification. There is no support for RTP transport of timed text in this release; 3GPP (MP4) files containing timed text may only be downloaded.

NOTE: When a PSS client supports timed text it needs to be able to receive and parse 3GPP (MP4) files containing the text streams. This does not imply a requirement on PSS clients to be able to render other continuous media types contained in 3GPP (MP4) files, e.g. AMR and H.263,if such media types are included in a presentation together with timed text. Audio and video are instead streamed to the client using RTSP/RTP (see clause 6.2).

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Tampere, Finland, 22-26 July 2002

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CHANGE REQUEST

⌘	**26.234**	**CR**	**032**	⌘ **rev**	**3**	⌘	Current version:	**5.1.0**	⌘

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Proposed change affects:	UICC apps⌘		ME	X	Radio Access Network		Core Network	

Title:	⌘	Required RTSP header support			
Source:	⌘	TSG-SA WG4			
Work item code:	⌘	PSS-E	***Date:***	⌘	12/Sept/2002
Category:	⌘	**F**	***Release:***	⌘	Rel-5
		Use one of the following categories: ***F*** *(correction)* ***A*** *(corresponds to a correction 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.			*Use one of the following releases:* *2 (GSM Phase 2)* *R96 (Release 1996)* *R97 (Release 1997)* *R98 (Release 1998)* *R99 (Release 1999)* *Rel-4 (Release 4)* *Rel-5 (Release 5)* *Rel-6 (Release 6)*

Reason for change:	⌘	Currently some headers of a minimal implementation are missing.
Summary of change:	⌘	Inclusion of RTSP headers Timestamp and User-Agent. Clarification of notes describing the usage of RTSP methods and headers.
Consequences if not approved:	⌘	Specification not compliant with RTSP (except for the User-Agent header which is "highly recommended").

Clauses affected:	⌘	A.2.1, G.2				
		Y	**N**			
Other specs affected:	⌘		**N**	Other core specifications	⌘	
			N	Test specifications		
			N	O&M Specifications		
Other comments:	⌘					

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A.2 RTSP

A.2.1 General

Clause 5.3.2 of the present document defines the required RTSP support in PSS clients and servers by making references to Appendix D of [5]. The current clause gives an overview of the methods (see Table A.2) and headers (see Table A.3) that are specified in the referenced Appendix D. An example of an RTSP session is also given.

Table A.2: Overview of the required RTSP method support

Method	Requirement for a minimal on-demand playback client according to [5].	Requirement for a PSS client according to the present document.	Requirement for a minimal on-demand playback server according to [5].	Requirement for a PSS server according to the present document.
OPTIONS	O	O	Respond	Respond
REDIRECT	Respond	Respond	O	O
DESCRIBE	O	Generate	O	Respond
SETUP	Generate	Generate	Respond	Respond
PLAY	Generate	Generate	Respond	Respond
PAUSE	Generate	Generate	Respond	Respond
TEARDOWN	Generate	Generate	Respond	Respond

NOTE 1: O = Support is optional
NOTE 2: 'Generate' means that the client/server is required to ~~be able to~~ generate the request where applicable.
NOTE 3: 'Respond' means that the client/server is required ~~understand and be able~~ to properly respond to the request.

Table A.3: Overview of the required RTSP header support

Header	Requirement for a minimal on-demand playback client according to [5].	Requirement for a PSS client according to the present document.	Requirement for a minimal on-demand playback server according to [5].	Requirement for a PSS server according to the present document.
Connection	include/understand	include/understand	include/understand	include/understand
Content-Encoding	understand	understand	include	include
Content-Language	understand	understand	include	include
Content-Length	understand	understand	include	include
Content-Type	understand	understand	include	include
CSeq	include/understand	include/understand	include/understand	include/understand
Location	understand	understand	O	O
Public	O	O	include	include
Range	O	include/understand	understand	include/understand
Require	O	O	understand	understand
RTP-Info	understand	understand	include	include
Session	include	include	understand	understand
Timestamp	O	O	include/understand	include/understand
Transport	include/understand	include/understand	include/understand	include/understand
User-Agent[4]	O	O	O	O

NOTE 1: O = Support is optional
NOTE 2: 'include' means that the client/server is required to ~~be able to~~ include the header in a request or response where applicable.
NOTE 3: 'understand' means that the client/server is required to ~~be able to understand the header and~~ respond properly if the header is received in a request or response.
NOTE 4: According to [5] the "User-Agent" header is not strictly required for a minimal RTSP client implementation, although it is highly recommended that it is included with requests. The same applies to a PSS client according to the present document.

The example below is intended to give some more understanding of how RTSP and SDP are used within the 3GPP PSS. The example assumes that the streaming client has the RTSP URL to a presentation consisting of an H.263 video sequence and AMR speech. RTSP messages sent from the client to the server are in **bold** and messages from the server to the client in *italic*. In the example the server provides aggregate control of the two streams.

EXAMPLE:

DESCRIBE rtsp://mediaserver.com/movie.test RTSP/1.0
CSeq: 1
User-Agent: TheStreamClient/1.1b2

RTSP/1.0 200 OK
CSeq: 1
Content-Type: application/sdp
Content-Length: 435

v=0
o=- 950814089 950814089 IN IP4 144.132.134.67
s=Example of aggregate control of AMR speech and H.263 video
e=foo@bar.com

c=IN IP4 0.0.0.0
b=AS:77
t=0 0
a=range:npt=0-59.3478
a=control:*
m=audio 0 RTP/AVP 97
b=AS:13
a=rtpmap:97 AMR/8000
a=fmtp:97
a=maxptime:200
a=control:streamID=0
m=video 0 RTP/AVP 98
b=AS:64
a=rtpmap:98 H263-2000/90000
a=fmtp:98 profile=3;level=10
a=control: streamID=1

SETUP rtsp://mediaserver.com/movie.test/streamID=0 RTSP/1.0
CSeq: 2
Transport: RTP/AVP/UDP;unicast;client_port=3456-3457
User-Agent: TheStreamClient/1.1b2

RTSP/1.0 200 OK
CSeq: 2
Transport: RTP/AVP/UDP;unicast;client_port=3456-3457; server_port=5678-5679
Session: dfhyrio90llk

SETUP rtsp://mediaserver.com/movie.test/streamID=1 RTSP/1.0
CSeq: 3
Transport: RTP/AVP/UDP;unicast;client_port=3458-3459
Session: dfhyrio90llk
User-Agent: TheStreamClient/1.1b2

RTSP/1.0 200 OK
CSeq: 3
Transport: RTP/AVP/UDP;unicast;client_port=3458-3459; server_port=5680-5681
Session: dfhyrio90llk

PLAY rtsp://mediaserver.com/movie.test RTSP/1.0
CSeq: 4
Session: dfhyrio90llk
User-Agent: TheStreamClient/1.1b2

RTSP/1.0 200 OK
CSeq: 4
Session: dfhyrio90llk
Range: npt=0-
RTP-Info: url= rtsp://mediaserver.com/movie.test/streamID=0; seq=9900;rtptime=4470048,
url= rtsp://mediaserver.com/movie.test/streamID=1; seq=1004;rtptime=1070549

NOTE: Headers can be folded onto multiple lines if the continuation line begins with a space or horizontal tab. For more information, see RFC2616 [17].

The user watches the movie for 20 seconds and then decides to fast forward to 10 seconds before the end…

PAUSE rtsp://mediaserver.com/movie.test RTSP/1.0
CSeq: 5
Session: dfhyrio90llk
User-Agent: TheStreamClient/1.1b2

PLAY rtsp://mediaserver.com/movie.test RTSP/1.0
CSeq: 6
Range: npt=50-59.3478
Session: dfhyrio90llk
User-Agent: TheStreamClient/1.1b2

RTSP/1.0 200 OK
CSeq: 5
Session: dfhyrio90llk

RTSP/1.0 200 OK
CSeq: 6
Session: dfhyrio90llk
Range: npt=50-59.3478
RTP-Info: url= rtsp://mediaserver.com/movie.test/streamID=0;
seq=39900;rtptime=44470648,
url= rtsp://mediaserver.com/movie.test/streamID=1;
seq=31004;rtptime=41090349

After the movie is over the client issues a TEARDOWN to end the session…

TEARDOWN rtsp://mediaserver.com/movie.test RTSP/1.0
CSeq: 7
Session: dfhyrio90llk
User-Agent: TheStreamClient/1.1b2

RTSP/1.0 200 OK
Cseq: 7
Session: dfhyrio90llk
Connection: close

(…un-altered sections left out here…)

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.
- By default, the peak decoding byte rate is defined according to the video coding profile and level in use. For example, H.263 Level 10 requires support for bit-rates up to 64000 bits per second. Thus, the peak decoding byte rate equals to 8000 bytes per second.
- The default decoding macroblock rate is defined according to the video coding profile and level in use. If MPEG-4 Visual is in use, the default macroblock rate equals to VCV decoder rate. If H.263 is in use, the default macroblock rate equals to (1 / minimum picture interval) multiplied by number of macroblocks in maximum picture format. For example, H.263 Level 10 requires support for picture formats up to QCIF and minimum picture interval down to 2002 / 30000 sec. Thus, the default macroblock rate would be 30000 x 99 / 2002 ≈ 1484 macroblocks per second.

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 following header fields are defined for RTSP:

- x-predecbufsize:<size of the hypothetical pre-decoder buffer>
  This gives the suggested size of the Annex G hypothetical pre-decoder buffer in bytes.
- x-initpredecbufperiod:<initial pre-decoder buffering period>
  This gives the required initial pre-decoder buffering period specified according to Annex G. Values are

interpreted as clock ticks of a 90-kHz clock. That is, the value is incremented by one for each 1/90 000 seconds. For example, value 180 000 corresponds to a two second initial pre-decoder buffering.

- x-initpostdecbufperiod:<initial post-decoder buffering period>
  This gives the required initial post-decoder buffering period specified according to Annex G. Values are interpreted as clock ticks of a 90-kHz clock.

These header fields are defined for the response of an RTSP PLAY request only. Their use is optional.

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 post-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
```

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CHANGE REQUEST

⌘	**26.234**	**CR**	**034**	⌘ **rev**	**1**	⌘	Current version:	**5.1.0**	⌘

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Proposed change affects:	UICC apps⌘ [ ]	ME [X]	Radio Access Network [ ]	Core Network [ ]

Title:	⌘	Including bitrate information for H.263		
Source:	⌘	TSG-SA WG4		
Work item code:	⌘	PSS-E	***Date:*** ⌘	12/Sept/2002
Category:	⌘	**F**	***Release:*** ⌘	Rel-5
		Use one of the following categories: ***F*** *(correction)* ***A*** *(corresponds to a correction 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.	*Use one of the following releases:* *2 (GSM Phase 2)* *R96 (Release 1996)* *R97 (Release 1997)* *R98 (Release 1998)* *R99 (Release 1999)* *Rel-4 (Release 4)* *Rel-5 (Release 5)* *Rel-6 (Release 6)*	

Reason for change:	⌘	There is no indication for H.263 bitrate information in the .3gp file format. Two additional fields are needed to describe the bitrate of H.263 streams: Max Bitrate: the maximum bitrate in bits per second of this elementary stream in any time window of one second duration. Avg Bitrate: the average bitrate in bits per second of this elementary stream. For streams with variable bitrate this value shall be set to zero.
Summary of change:	⌘	Adding an atom with two additional fields to describe bitrate information of H.263 streams.
Consequences if not approved:	⌘	Client has no advance indication of H.263 bitrate information in the .3gp file format.

Clauses affected:	⌘	D.8			
		Y	**N**		
Other specs affected:	⌘		**X**	Other core specifications	⌘
			X	Test specifications	
			X	O&M Specifications	
Other comments:	⌘				

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D.8 H263SpecificAtom field for H263SampleEntry atom

The H263SpecificAtom fields for H. 263 shall be as defined in table D.7. The H263SpecificAtom for the H263SampleEntry Atom shall always be included if the MP4 file contains H.263 media.

The H263SpecificAtom for H263 is composed of the following fields.

Table D.7: The H263SpecificAtom fields H263SampleEntry

Field	Type	Details	Value
AtomHeader.Size	Unsigned int(32)		
AtomHeader.Type	Unsigned int(32)		'd263'
DecSpecificInfo	H263DecSpecStruc	Structure which holds the H.263 Specific information	
BitrateAtom		Specific bitrate information (optional)	

AtomHeader Size and Type: indicate the size and type of the H.263 decoder-specific atom. The type must be 'd263'.

DecSpecificInfo: This is the structure where the H263 stream specific information resides.

H263DecSpecStruc is defined as follows:

struct **H263DecSpecStruc**{

Unsigned int (32) **vendor**

Unsigned int (8) **decoder_version**

Unsigned int (8) **H263_Level**

Unsigned int (8) **H263_Profile**

}

The definitions of H263DecSpecStruc members are as follows:

vendor: four character code of the manufacturer of the codec, e.g. 'VXYZ'. The vendor field gives information about the vendor whose codec is used to create the encoded data. It is an informative field which may be used by the decoding end. If a manufacturer already has a four character code, it is recommended that it uses the same code in this field. Else, it is recommended that the manufacturer creates a four character code which best addresses the manufacturer's name. It can be safely ignored.

decoder_version: version of the vendor's decoder which can decode the encoded stream in the best (i.e. optimal) way. This field is closely tied to the vendor field. It may give advantage to the vendor which has optimal encoder-decoder version pairs. . The value is set to 0 if decoder version has no importance for the vendor. It can be safely ignored.

H263_Level and H263_Profile: These two parameters define which H263 profile and level is used. These parameters are based on the MIME media type video/H263-2000. The profile and level specifications can be found in [23].

EXAMPLE 1: H.263 Baseline = {H263_Level = 10, H263_Profile = 0}

EXAMPLE 2: H.263 Profile 3 @ Level 10 = {H263_Level = 10 , H263_Profile = 3}

NOTE: The "hinter", for the creation of the hint tracks, can use the information given by the H263DecSpecStruc members.

The BitrateAtom field shall be as defined in table D.7.1. The BitrateAtom may be included if the MP4 file contains H.263 media.

The BitrateAtom is composed of the following fields.

Table D.7.1: The BitrateAtom fields

Field	Type	Details	Value
AtomHeader.Size	Unsigned int(32)		
AtomHeader.Type	Unsigned int(32)		'bitr'
DecBitrateInfo	DecBitrStruc	Structure which holds the Bitrate information	

AtomHeader Size and Type: indicate the size and type of the bitrate atom. The type must be 'bitr'.

DecBitrateInfo: This is the structure where the stream bitrate information resides.

DecBitrStruc is defined as follows:

struct **DecBitrStruc**{

Unsigned int (32) **Avg_Bitrate**

Unsigned int (32) **Max_Bitrate**

}

The definitions of DecBitrStruc members are as follows:

Avg_Bitrate: the average bitrate in bits per second of this elementary stream. For streams with variable bitrate this value shall be set to zero.

Max_Bitrate: the maximum bitrate in bits per second of this elementary stream in any time window of one second duration.

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CHANGE REQUEST

⌘	26.234	CR	035	⌘ rev	1	⌘	Current version:	5.1.0	⌘

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Proposed change affects: UICC apps⌘ [ ] ME [X] Radio Access Network [ ] Core Network [ ]

Title:	⌘	RTCP Reports and Link Aliveness in Ready State		
Source:	⌘	TSG-SA WG4		
Work item code:	⌘	PSS-E	***Date:*** ⌘	12/Sept/2002
Category:	⌘	**F**	***Release:*** ⌘	Rel-5
		Use one of the following categories: ***F*** *(correction)* ***A*** *(corresponds to a correction 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.		*Use one of the following releases:* *2 (GSM Phase 2)* *R96 (Release 1996)* *R97 (Release 1997)* *R98 (Release 1998)* *R99 (Release 1999)* *Rel-4 (Release 4)* *Rel-5 (Release 5)* *Rel-6 (Release 6)*

Reason for change: ⌘

3GPP TS 26.234 section A.2.2.2 deals with implementation guidelines for detecting link aliveness. It specifies that 'In order for the server to be able to detect the client's aliveness, the PSS client should send "wellness" information to the PSS server for a defined interval as described in the RFC2326'. The use of RTCP reports is suggested as first option.

The appropriate section of RFC2326 (section A.2) specifies:

> In general, the server changes state on receiving requests. If the server is in state Playing or Recording and in unicast mode, it MAY revert to Init and tear down the RTSP session if it has not received "wellness" information, such as RTCP reports or RTSP commands, from the client for a defined interval, with a default of one minute. The server can declare another timeout value in the Session response header (Section 12.37). If the server is in state Ready, it MAY revert to Init if it does not receive an RTSP request for an interval of more than one minute. Note that some requests (such as PAUSE) may be effective at a future time or position, and server state changes at the appropriate time. The server reverts from state Playing or Recording to state Ready at the end of the range requested by the client.

Analysis of this text tells us:
· In 'Playing' or 'Recording' states we may use RTCP reports for link aliveness mechanism (not mandatory).
· In 'Ready' state (e.g., following a 'SETUP' or 'PAUSE' request) there is a need for link aliveness mechanism (not mandatory), but RTCP reports are not mentioned.

This is a small difference that may cause interoperability problems, when a client

	that does not send wellness information following a 'SETUP' or 'PAUSE' request will be disconnected. Therefore a somewhat more strict specification may be needed in this specific case.
Summary of change: ⌘	Recommending that the client should send the same wellness information in 'Ready' state as in 'Playing' and 'Recording' states, and the server should detect the same client's wellness information in 'Ready' state as in 'Playing' and 'Recording' states.
Consequences if not approved: ⌘	Possible interoperability problems, when a client that does not send wellness information following a 'SETUP' or 'PAUSE' request could be disconnected.

Clauses affected: ⌘	A.2.2.2				
	Y	**N**			
Other specs affected: ⌘		**X**	Other core specifications	⌘	
		X	Test specifications		
		X	O&M Specifications		
Other comments: ⌘					

How to create CRs using this form:

Comprehensive information and tips about how to create CRs can be found at http://www.3gpp.org/specs/CR.htm. Below is a brief summary:

1) Fill out the above form. The symbols above marked ⌘ contain pop-up help information about the field that they are closest to.

2) Obtain the latest version for the release of the specification to which the change is proposed. Use the MS Word "revision marks" feature (also known as "track changes") when making the changes. All 3GPP specifications can be downloaded from the 3GPP server under ftp://ftp.3gpp.org/specs/ For the latest version, look for the directory name with the latest date e.g. 2001-03 contains the specifications resulting from the March 2001 TSG meetings.

3) With "track changes" disabled, paste the entire CR form (use CTRL-A to select it) into the specification just in front of the clause containing the first piece of changed text. Delete those parts of the specification which are not relevant to the change request.

A.2.2.2 Detecting link aliveness

In the wireless environment, connection may be lost due to fading, shadowing, loss of battery power, or turning off the terminal even though the PSS session is active. In order for the server to be able to detect the client's aliveness, the PSS client should send "wellness" information to the PSS server for a defined interval as described in the RFC2326. There are several ways for detecting link aliveness described in the RFC2326, however, the client should be careful about issuing "PLAY method without Range header field" too close to the end of the streams, because it may conflict with pipelined PLAY requests. Below is the list of recommended "wellness" information for the PSS clients and servers in a prioritised order.

1. RTCP
2. OPTIONS method with Session header field

NOTE: Both servers and clients can initiate this OPTIONS method.

<u>The client should send the same wellness information in 'Ready' state as in 'Playing' and 'Recording' states, and the server should detect the same client's wellness information in 'Ready' state as in 'Playing' and 'Recording' states. In particular, the same link aliveness mechanism should be managed following a 'PAUSE' request and response.</u>

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CHANGE REQUEST

⌘	**26.234**	**CR**	**036**	⌘ **rev**	**2**	⌘	Current version:	**5.1.0**	⌘

*For **HELP** on using this form, see bottom of this page or look at the pop-up text over the ⌘ symbols.*

Proposed change affects:	UICC apps⌘ [ ]	ME [X]	Radio Access Network [ ]	Core Network [ ]

Title:	⌘	Correction of media and session-level bandwidth fields in SDP.			
Source:	⌘	TSG-SA WG4			
Work item code:	⌘	PSS-E	***Date:***	⌘	12/Sept/2002
Category:	⌘	**F**	***Release:***	⌘	Rel-5

Use one of the following categories:
F *(correction)*
A *(corresponds to a correction 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.

Use one of the following releases:
2 (GSM Phase 2)
R96 (Release 1996)
R97 (Release 1997)
R98 (Release 1998)
R99 (Release 1999)
Rel-4 (Release 4)
Rel-5 (Release 5)
Rel-6 (Release 6)

Reason for change:	⌘	Release-5 PSS clients need SDP bandwidth information in order to properly set up QoS parameters. Inclusion of this information is currently optional from the server point of view. Also, the "session level" bandwidth field is redundant, and how it is computed and used is subject to interpretation.
Summary of change:	⌘	State that PSS servers shall provide the media-level "b=AS:" field in SDP, and that clients should compute bandwidth information using this media-level "b=AS:" information rather than the session level value.
Consequences if not approved:	⌘	Clients do not have enough information about presentation bandwidth to properly set up QoS in PSS systems.

Clauses affected:	⌘	5.3.3, A.1			
		Y	**N**		
Other specs affected:	⌘		**X**	Other core specifications	⌘
			X	Test specifications	
			X	O&M Specifications	
Other comments:	⌘				

. . .

5.3.3 SDP

5.3.3.1 General

RTSP requires a presentation description. SDP shall be used as the format of the presentation description for both PSS clients and servers. PSS servers shall provide and clients interpret the SDP syntax according to the SDP specification [6] and appendix C of [5]. The SDP delivered to the PSS client shall declare the media types to be used in the session using a codec specific MIME media type for each media. MIME media types to be used in the SDP file are described in clause 5.4 of the present document.

The SDP [6] specification requires certain fields to always be included in an SDP file. Apart from this a PSS server shall always include the following fields in the SDP:

- "a=control:" according to clauses C.1.1, C.2 and C.3 in [5];
- "a=range:" according to clause C.1.5 in [5];
- "a=rtpmap:" according to clause 6 in [6];
- "a=fmtp:" according to clause 6 in [6].

~~The bandwidth field in SDP should be used to indicate to the PSS client the amount of bandwidth that is required for the session and the individual media in the presentation. Therefore, a PSS server should include the "b=AS:" field in the SDP (both on the session and media level) and a PSS client shall be able to interpret this field. For RTP based applications, AS gives the RTP "session bandwidth" (including UDP/IP overhead) as defined in section 6.2 of [9].~~

The bandwidth field in SDP is needed by the client in order to properly set up QoS parameters. Therefore, a PSS server shall include the “b=AS:” field at the media level for each media stream in SDP, and a PSS client shall interpret this field. When a client receives SDP, it should ignore the session level “b=AS:” parameter (if present), and instead calculate session bandwidth from the media level bandwidth values of the relevant streams. Note that for RTP based applications , ‘b=AS:’ gives the RTP "session bandwidth" (including UDP/IP overhead) as defined in section 6.2 of [9].

NOTE: The SDP parsers and/or interpreters shall be able to accept NULL values in the 'c=' field (e.g. 0.0.0.0 in IPv4 case). This may happen when the media content does not have a fixed destination address. For more details, see Section C.1.7 of [5] and Section 6 of [6].

5.3.3.2 Additional SDP fields

The following Annex G-related media level SDP fields are defined for PSS:

. . .

Annex A (informative): Protocols

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 2327 [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	~~R~~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
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
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

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 2327, 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.

The example below shows an SDP file that could be sent to a PSS client to initiate unicast streaming of a H.263 video sequence.

EXAMPLE:
```
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
b=AS:128
t=0 0
a=range:npt=0-45.678
m=video 1024 RTP/AVP 96
b=AS:128
a=rtpmap:96 H263-2000/90000
a=fmtp:96 profile=3;level=10
a=control:rtsp://mediaserver.com/movie
a=recvonly
```

A.2 RTSP

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Tdoc ⌘S4-020492

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CHANGE REQUEST

⌘ **26.234**	**CR 037**	⌘ **rev**	**2** ⌘	Current version:	**5.1.0** ⌘

For ***HELP*** *on using this form, see bottom of this page or look at the pop-up text over the ⌘ symbols.*

Proposed change affects:	UICC apps⌘ [ ]	ME [X]	Radio Access Network [ ]	Core Network [ ]

Title: ⌘	Correction of usage of MIME parameters for AMR
Source: ⌘	TSG-SA WG4
Work item code: ⌘	PSS-E
Date: ⌘	12/Sept/2002
Category: ⌘	**F**
Release: ⌘	Rel-5

Use *one* *of the following categories:*
F *(correction)*
A *(corresponds to a correction 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.

Use *one* *of the following releases:*
2 (GSM Phase 2)
R96 (Release 1996)
R97 (Release 1997)
R98 (Release 1998)
R99 (Release 1999)
Rel-4 (Release 4)
Rel-5 (Release 5)
Rel-6 (Release 6)

Reason for change: ⌘	Certain legacy AMR parameters (mode_set, mode_change_period, mode_change_neighbor) are not relevant to PSS, and current PSS usage of these parameters conflicts with the description of the parameters in RFC3267.
Summary of change: ⌘	Recommend that PSS servers should not send these parameters, and that PSS clients shall ignore these parameters if received.
Consequences if not approved: ⌘	PSS systems would continue to use these parameters in a way that conflicts with RFC3267 usage and delivers no relevant information to the client.

Clauses affected: ⌘	D.7

Other specs affected: ⌘	Y	N		
		X	Other core specifications	⌘
		X	Test specifications	
		X	O&M Specifications	

Other comments: ⌘	

…

D.7 AMRSpecificAtom field for AMRSampleEntry atom

The AMRSpecificAtom fields for AMR and AMR-WB shall be as defined in table D.6. The AMRSpecificAtom for the AMRSampleEntry Atom shall always be included if the MP4 file contains AMR or AMR-WB media.

Table D.6: The AMRSpecificAtom fields for AMRSampleEntry

Field	Type	Details	Value
AtomHeader.Size	Unsigned int(32)		
AtomHeader.Type	Unsigned int(32)		'damr'
DecSpecificInfo	AMRDecSpecStruc	Structure which holds the AMR and AMR-WB Specific information	

AtomHeader Size and Type: indicate the size and type of the AMR decoder-specific atom. The type must be 'damr'.

DecSpecificInfo: the structure where the AMR and AMR-WB stream specific information resides.

The AMRDecSpecStruc is defined as follows:

struct **AMRDecSpecStruc**{

Unsigned int (32) **vendor**

Unsigned int (8) **decoder_version**

Unsigned int (16) **mode_set**

Unsigned int (8) **mode_change_period**

Unsigned int (8) **frames_per_sample**

}

The definitions of AMRDecSpecStruc members are as follows:

vendor: four character code of the manufacturer of the codec, e.g. 'VXYZ'. The vendor field gives information about the vendor whose codec is used to create the encoded data. It is an informative field which may be used by the decoding end. If a manufacturer already has a four character code, it is recommended that it uses the same code in this field. Else, it is recommended that the manufacturer creates a four character code which best addresses the manufacturer's name. It can be safely ignored.

decoder_version: version of the vendor's decoder which can decode the encoded stream in the best (i.e. optimal) way. This field is closely tied to the vendor field. It may give advantage to the vendor which has optimal encoder-decoder version pairs. The value is set to 0 if decoder version has no importance for the vendor. It can be safely ignored.

mode_set: the active codec modes. Each bit of the mode_set parameter corresponds to one mode. The bit index of the mode is calculated according to the 4 bit FT field of the AMR or AMR-WB frame structure. The mode_set bit structure is as follows: (B15xxxxxxB8B7xxxxxxB0) where B0 (Least Significant Bit) corresponds to Mode 0, and B8 corresponds to Mode 8.

The mapping of existing AMR modes to FT is given in table 1.a in [19]. A value of 0x81FF means all modes and comfort noise frames are possibly present in an AMR stream.

The mapping of existing AMR-WB modes to FT is given in Table 1.a in TS 26.201 [37]. A value of 0x83FF means all modes and comfort noise frames are possibly present in an AMR-WB stream.

As an example, if mode_set = 0000000110010101b, only Modes 0, 2, 4, 7 and 8 are present in the stream.

mode_change_period: defines a number N, which restricts the mode changes only at a multiple of N frames. If no restriction is applied, this value should be set to 0. If mode_change_period is not 0, the following restrictions apply to it according to the frames_per_sample field:

if (mode_change_period < frames_per_sample)

frames_per_sample = k x (mode_change_period)

else if (mode_change_period > frames_per_sample)

mode_change_period = k x (frames_per_sample)

where k : integer [2, …]

If mode_change_period is equal to frames_per_sample, then the mode is the same for all frames inside one sample.

frames_per_sample: defines the number of frames to be considered as 'one sample' inside the MP4 file. This number shall be greater than 0 and less than 16. A value of 1 means each frame is treated as one sample. A value of 10 means that 10 frames (of duration 20 msec each) are put together and treated as one sample. It must be noted that, in this case, one sample duration is 20 (msec/frame) x 10 (frame) = 200 msec. For the last sample of the stream, the number of frames can be smaller than frames_per_sample, if the number of remaining frames is smaller than frames_per_sample.

NOTE: The "hinter", for the creation of the hint tracks, can use the information given by the AMRDecSpecStruc members.

NOTE2: The following AMR MIME parameters are not relevant to PSS: {mode_set, mode_change_period, mode_change_neighbor}. PSS servers should not send these parameters in SDP, and PSS clients shall ignore these parameters if received.

D.8 H263SpecificAtom field for H263SampleEntry atom

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CHANGE REQUEST

⌘	**26.234**	**CR**	**038**	⌘ **rev**	**1**	⌘	Current version:	**5.1.0**	⌘

For ***HELP*** *on using this form, see bottom of this page or look at the pop-up text over the ⌘ symbols.*

Proposed change affects: UICC apps⌘ [ ] ME [X] Radio Access Network [ ] Core Network [ ]

Title:	⌘	Correction of Mapping of SDP parameters to UMTS QoS parameters (Annex J)
Source:	⌘	TSG-SA WG4
Work item code:	⌘	PSS-E
Date:	⌘	12/Sept/2002
Category:	⌘	**F**
Release:	⌘	Rel-5

Use one of the following categories:
F *(correction)*
A *(corresponds to a correction 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.

Use one of the following releases:
2 (GSM Phase 2)
R96 (Release 1996)
R97 (Release 1997)
R98 (Release 1998)
R99 (Release 1999)
Rel-4 (Release 4)
Rel-5 (Release 5)
Rel-6 (Release 6)

Reason for change:	⌘	Several parameters of Table J.1 are still marked with TBC (to be committed) or are wrong. This CR corrects these values.
Summary of change:	⌘	The parameters, “Maximum SDU-size”, “Delivery of erroneous SDUs”, “Delivery Order”, “Guaranteed bit rate for down-link”, “Guaranteed bit rate for uplink”, “Residual BER” and “Transfer Delay” are corrected or committed.
Consequences if not approved:	⌘	If not approved, the wrong values may result in failure of the PSS service.

Clauses affected:	⌘	Annex J

		Y	N		
Other specs affected:	⌘		**N**	Other core specifications	⌘
			N	Test specifications	
			N	O&M Specifications	

Other comments:	⌘	

Annex J (informative): Mapping of SDP parameters to UMTS QoS parameters

This Annex gives recommendation for the mapping rules needed by the PSS applications to request the appropriate QoS from the UMTS network (see Table J.1).

Table J.1: Mapping of SDP parameters to UMTS QoS parameters for PSS

QoS parameter	Parameter value	comment
Delivery of erroneous SDUs	"~~no~~No"~~[TBC]~~	
Delivery order	~~Yes~~"No"	
Traffic class	"Streaming class"	
Maximum SDU size	~~1520~~ 1400 bytes	According to RFC 2460 the SDU size must not exceed 1500 octets. A packet size of 1400 guarantees efficient transportation.
Guaranteed bit rate for downlink	1.025 * ~~SDP~~ session bandwidth ~~[TBC]~~	This session bandwidth is calculated from the SDP media level bandwidth values.
Maximum bit rate for downlink	Equal or higher to guaranteed bit rate in downlink	~~Specifying a minimum overhead bit rate per media might be useful and is FFS~~
Guaranteed bit rate for uplink	0.025 * ~~SDP~~ session bandwidth ~~[TBC]~~	
Maximum bit rate for uplink	Equal or higher to guaranteed bit rate in uplink	
Residual BER	1*10-5 ~~[TBC]~~	16 bit CRC should be enough
SDU error ratio	1*10-4 or better	~~1*10-3 could be acceptable. RLC AM mode should easily enable 10-4.~~
Traffic handling priority	Subscribed traffic handling priority	Ignored
Transfer delay	~~[1s to 1.5s]~~2 sec.	