Technical Specification Group Services and System Aspects ***TSGS#19(03)0092***
Meeting #19, Birmingham, UK, 17 - 20 March 2003

Source: **TSG-SA WG4**

Title: **CRs to TS 26.236 - Corrections (Release 5)**

Document for: **Approval**

Agenda Item: **7.4.3**

The following CRs, agreed at the TSG-SA WG4 meeting #25bis, are presented to TSG SA #19 for approval.

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.236	003	2	Rel-5	SDP bandwidth modifier for RTCP bandwidth	F	5.1.0	S4	TSG-SA WG4#25bis	S4-030259
26.236	004		Rel-5	Correction on QoS profile parameters for conversational multimedia applications	F	5.1.0	S4	TSG-SA WG4#25bis	S4-030186

3GPP TSG-SA4 Meeting #25bis
Berlin, Germany, 24-28 February 2003

Tdoc S4-030259

CR-Form-v5

CHANGE REQUEST

⌘	TS 26.236	CR	3	⌘ rev	2	⌘	Current version:	5.1.0	⌘

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Proposed change affects: ⌘	(U)SIM	ME/UE	Radio Access Network	Core Network
	[ ]	[X]	[ ]	[ ]

Title:	⌘	SDP bandwidth modifier for RTCP bandwidth		
Source:	⌘	TSG SA WG4		
Work item code:	⌘	IMS-CODEC	***Date:*** ⌘	18 March 2003
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)*

Reason for change:	⌘	The bandwidth allocated for RTCP traffic needs to be accurately estimated for rightsizing the bearer setup and for authorization purposes in IMS.
Summary of change:	⌘	The bandwidth allocated for RTCP is specified by appropriate "RS" and "RR" SDP modifiers.
Consequences if not approved:	⌘	Media PDP contexts are allocated using wrong assumptions about RTCP bandwidth (2.5% of session bandwidth), even if a different bandwidth for RTCP is used, or even if RTCP is not used at all. This also would lead to wrong authorization of network resources.

Clauses affected:	⌘	2, 7.1			
Other specs affected:	⌘	X	Other core specifications	⌘	24.228, 24.229, 29.208
			Test specifications		
			O&M Specifications		
Other comments:	⌘				

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

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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.

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] IETF RFC 2543: "SIP: Session Initiation Protocol".

[2] IETF RFC 2327: "SDP: Session Description Protocol".

[3] IETF RFC 1889: "RTP: A Transport Protocol for Real-Time Applications".

[4] IETF RFC 1890: "RTP Profile for Audio and Video Conferences with Minimal Control".

[5] 3GPP TS 26.235: "Packet switched conversational multimedia applications; Default codecs".

[6] 3GPP TS 24.228: "Signalling flows for the IP multimedia call control based on SIP and SDP; stage 3".

[7] 3GPP TS 24.229: "IP multimedia call control protocol based on SIP and SDP".

[8] 3GPP TS 23.228: "IP Multimedia Ssubsystem (IMS); Stage 2".

[9] 3GPP TS 23.107: "Quality of Service (QoS) concept and architecture".

[10] 3GPP TS 23.207: "End to end quality of service concept and architecture".

[11] 3GPP TS 23.060: "General Packet Radio Service (GPRS); Service description; Stage 2".

[12] 3GPP TS 26.071: "Mandatory Speech Codec speech processing functions; AMR Speech Codec; General description".

[13] 3GPP TS 26.090: "AMR speech Codec; Transcoding Functions".

[14] 3GPP TS 26.073: "AMR speech Codec; C-source code".

[15] 3GPP TS 26.104: "ANSI-C code for the floating-point Adaptive Multi-Rate AMR speech codec".

[16] 3GPP TS 26.171 (Release 5): "AMR speech codec, wideband; General description".

[17] 3GPP TS 26.190 (Release 5): "Mandatory Speech Codec speech processing functions AMR Wideband speech codec; Transcoding functions".

[18] 3GPP TS 26.201 (Release 5): "AMR speech codec, wideband; Frame structure".

[19] 3GPP TS 26.235: "Packet switched conversational multimedia applications; Default codecs ". Annex B: "RTP payload format and storage format for AMR and AMR-WB audio".

[20] ITU-T Recommendation H.263: "Video coding for low bit rate communication".

[21] IETF RFC 2429: "RTP Payload Format for the 1998 Version of ITU-T Rec. H.263 Video (H.263+)".

[22] ISO/IEC 14496-2 (1999): "Information technology - Coding of audio-visual objects - Part 2: Visual".

[23] IETF RFC 3016: "RTP Payload Format for MPEG-4 Audio/Visual Streams".

[24] ITU-T Recommendation H.263 (annex X): "Annex X: Profiles and levels definition".

[25] 3GPP TS 26.235: "Packet Switched Conversational Multimedia Applications; Default Codecs ". Annex C: "ITU-T H.263 MIME media type registration".

[26] ITU-T Recommendation T.140 (1998): "Protocol for multimedia application text conversation" (with amendment 2000).

[27] IETF RFC 2793: "RTP Payload for Text Conversation".

[28] IETF RFC 3578: "SDP bandwidth modifier for RTCP bandwidth".

END of section 2.

7.1 Bandwidth

The bandwidth information of each media type shall be carried in SDP messages in both session and media type level during codec negotiation, session establishment and resource reallocation.

The bandwidth for RTCP traffic shall be described using the "RS" and "RR" SDP bandwidth modifiers at media level, as specified by [28]. Therefore, a conversational multimedia terminal shall include the "b=RS:" and "b=RR:" fields in SDP, and shall be able to interpret them. There shall be a limit on the allowed RTCP bandwidth for a session signalled by the terminal. This limit is defined as follows:

- 4000 bps for the RS field (at media level);
- 3000 bps for the RR field (at media level).

END of section 7.1.

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CHANGE REQUEST

⌘	**TS 26.236**	**CR**	**004**	⌘ **rev**	**-**	⌘	Current version:	**5.1.0** ⌘

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Proposed change affects: ⌘ (U)SIM [ ] ME/UE [X] Radio Access Network [ ] Core Network [ ]

Title:	⌘ Correction on QoS profile parameters for conversational multimedia applications			
Source:	⌘ TSG SA WG4			
Work item code:	⌘ IMS-CODEC		***Date:*** ⌘	18 March 2003
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)*

Reason for change: ⌘	Annex B contains some errors in the tables related to some QoS profile parameters. Also some clarifications to the text are added.
Summary of change: ⌘	Traffic handling priority and allocation retention priority values are corrected. Clarification on SNDCP fragmentation and transfer delay.
Consequences if not approved: ⌘	Guidance for operators and vendors is wrong.

Clauses affected: ⌘	Annex B	
Other specs affected: ⌘	[ ] Other core specifications [ ] Test specifications [ ] O&M Specifications	⌘
Other comments: ⌘		

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

This clause gives recommendations for mapping of SDP parameters in UMTS QoS parameters for conversational multimedia applications. Different use cases will be considered. Each use case generates an example QoS profile parameters table. The values indicated are derived by applications' QoS requirements, and may not be fulfilled by the network. In the parameters for guaranteed and maximum bit rates a granularity of 1 kbps is assumed for bearers up to 64 kbps, as defined in the TS 24.008. Therefore the "Ceiling" function is used for up-rounding fractional values, wherever needed. In addition, the same specification defines a granularity of 10 bytes for the Maximum SDU sizes values. This is taken into account in the computation of this field in the QoS profile.

Use case 1 – Voice over IP

This use case includes the scenario in which two conversational multimedia terminals establish a bi-directional Voice over IP (VoIP) connection for speech communication, using the AMR or AMR-WB codecs with the same bit rate in both uplink and downlink directions.

For example an AMR VoIP stream encoded at 12.2 kbps, with one speech frame encapsulated into an RTP packet, would yield IP packets of the following size (using the mandated bandwidth efficient mode):

20 (IPv4) + 8 (UDP) + 12 (RTP) + 32 (AMR RTP payload) = 72 bytes, or

40 (IPv6 with no extension headers) + 8 (UDP) + 12 (RTP) + 32 (AMR RTP payload) = 92 bytes.

The gross bit rate including uncompressed RTP/UDP/IPv4 headers would be 28.8 kbps. The value in the b=AS media level parameter would be 29.

To determine the Maximum SDU size parameter we should consider the maximum packet size that can be generated with a speech codec. This is exactly that generated by a AMR-WB stream at 23.85 kbps packetized in bandwidth efficient mode and with 1 speech frame per packet. Considering uncompressed RTP/UDP/IPv6 headers, the maximum packet size is 121 bytes.

The QoS profile would be set then using the following parameters:

Table B.1: QoS profile for AMR VoIP at 12.2 kbps

QoS parameter	Parameter value	Comment
Delivery of erroneous SDUs	No	
Delivery order	No	To minimize delay in the access stratum. The application should take care of eventual packet reordering
Traffic class	Conversational	
Maximum SDU size	130 bytes	10 bytes granularity. The RTCP packet size might change the maximum SDU size limitation [tbc]
Guaranteed bitrate for downlink	SDP media bw in DL + 2.5% * (SDP media bw in DL+ SDP media bw in UL) = Ceil(30.45)=31 kbps	
Maximum bit rate for downlink	Ceil(30.45)=31 kbps	
Guaranteed bitrate for uplink	SDP media bw in UL + 2.5% * (SDP media bw in UL+ SDP media bw in DL) = Ceil(30.45)=31 kbps	
Maximum bit rate for uplink	Ceil(30.45)=31 kbps	
Residual BER	10^{-5}	16 bit CRC
SDU error ratio	$7*10^{-3}$	
Traffic handling priority	Not used in Conversational traffic class~~Subscribed traffic handling priority~~	~~Not relevant~~
Transfer delay	100 ms	
SDU format information	Not used	
Allocation/retention priority	Subscribed allocation/retention~~traffic handling~~ priority	Not relevant for the application
Source statistics descriptor	"Speech"	

In some cases, multiple AMR or AMR-WB rates are available, and rate control techniques allow to switch between different modes based on the received speech quality. For example, if the available AMR mode set is {4.75, 10.2, 12.2} kbps, the set of gross bit rates are:

AMR 4.75 kbps: 21.6 kbps (including RTP/UDP/IPv4 headers). [SDP b=AS parameter would be 22].

AMR 10.2 kbps: 26.8 kbps (including RTP/UDP/IPv4 headers). [SDP b=AS parameter would be 27].

AMR 12.2 kbps: 28.8 kbps (including RTP/UDP/IPv4 headers). [SDP b=AS parameter would be 29].

The maximum bit rate is set to the highest mode of the codec. However, the procedure on how to choose the guaranteed bit rate when several codec rates are available is to be defined. Here we provide an example QoS profile in which the guaranteed speech quality is at least that of 10.2 kbps AMR for both uplink and downlink directions, while the non-guaranteed maximum quality is that of 12.2 kbps for both uplink and downlink directions.

Table B.2: QoS profile for AMR VoIP at 3 bit rates with rate control

QoS parameter	Parameter value	Comment
Delivery of erroneous SDUs	No	
Delivery order	No	To minimize delay in the access stratum. The application should take care of eventual packet reordering
Traffic class	Conversational	
Maximum SDU size	130 bytes	10 bytes granularity. The RTCP packet size might change the maximum SDU size limitation [tbc]
Guaranteed bitrate for downlink	SDP media bw in DL + 2.5% * (SDP media bw in DL+ SDP media bw in UL) = Ceil(28.35)=29 kbps	Guaranteed quality 10.2 kbps (media bw = 27 kbps)
Maximum bit rate for downlink	SDP media bw in DL + 2.5% * (SDP media bw in DL+ SDP media bw in UL) = Ceil(30.35)=31 kbps	Non-guaranteed quality 12.2 kbps (media bw = 29 kbps)
Guaranteed bitrate for uplink	SDP media bw in UL+ 2.5% * (SDP media bw in UL+ SDP media bw in DL) = Ceil(28.35)=29 kbps	Guaranteed quality 10.2 kbps (media bw = 27 kbps)
Maximum bit rate for uplink	SDP media bw in UL + 2.5% * (SDP media bw in UL+ SDP media bw in DL) = Ceil(30.35)=31 kbps	Non-guaranteed quality 12.2 kbps (media bw = 29 kbps)
Residual BER	10^{-5}	16 bit CRC
SDU error ratio	$7*10^{-3}$	
Traffic handling priority	Not used in Conversational traffic class~~Subscribed traffic handling priority~~	~~Not relevant~~
Transfer delay	100 ms	
SDU format information	Not used	
Allocation/retention priority	Subscribed allocation/retention~~traffic handling~~ priority	Not relevant for the application
Source statistics descriptor	"Speech"	

Use case 2 – Unidirectional video

This use case includes the scenario in which two conversational multimedia terminals establish a uni-directional video connection, using the H.263 or MPEG-4 codecs.

The video codec in this example has a bitrate of 36 kbps, with RTP payload packets of 75 bytes (excluding payload header which is, for example, 2 bytes). The sending terminal would produce IP packets of the following size:

20 (IPv4) + 8 (UDP) + 12 (RTP) + 77 (video RTP payload+payload header) = 117 bytes, or

40 (IPv6 with no extension headers) + 8 (UDP) + 12 (RTP) + 77 (video RTP payload+payload header) = 137 bytes.

The gross bit rate including uncompressed RTP/UDP/IPv4 headers would be 56.2 kbps. The value in the b=AS media level parameter would be 57.

The maximum video packet size is limited to 512 bytes in section 5.2. This value is fine if transmission occurs over the UMTS Iu interface. However, in order to avoid SNDCP fragmentation of ~~IP~~ packets over the GERAN Gb interface (where the default size for LLC data field (=SNDCP frame) is 500 bytes) the maximum IP packet size is 500 – 4 (unacknowledged mode SNDCP header) = 496 bytes. Therefore, the maximum size of a video packet is 496 – 60 (RTP/UDP/IPv6 uncompressed headers) = 436 bytes (including RTP payload header). 400 bytes is a safer value.

The QoS profile of the receiving terminal would be set then using the following parameters:

Table B.3: QoS profile for unidirectional video at 36 kbps

QoS parameter	Parameter value	Comment
Delivery of erroneous SDUs	No	
Delivery order	No	To minimize delay in the access stratum. The application should take care of eventual packet reordering
Traffic class	Conversational	
Maximum SDU size	500 bytes	10 bytes granularity
Guaranteed bitrate for downlink	SDP media bw in DL + 2.5% * (SDP media bw in DL) = Ceil(58.43)=59 kbps	
Maximum bit rate for downlink	Equal or higher than guaranteed bit rate	
Guaranteed bitrate for uplink	2.5% * (SDP media bw in DL) = Ceil(1.43)=2 kbps	For RTCP
Maximum bit rate for uplink	Equal or higher than guaranteed bit rate	
Residual BER	10^{-5}	16 bit CRC
SDU error ratio	10^{-3}	
Traffic handling priority	Not used in Conversational traffic class~~Subscribed traffic handling priority~~	~~Not relevant~~
Transfer delay	250 ms	
SDU format information	Not used	
Allocation/retention priority	Subscribed allocation/retention~~traffic handling~~ priority	Not relevant for the application
Source statistics descriptor	"Unknown"	

Use case 3 – Video telephony

This use case includes the scenario in which two conversational multimedia terminals establish a bi-directional speech/video connection, using the AMR/AMR-WB and H.263/MPEG-4 codecs at the same bit rates in uplink and downlink directions.

The video codec in this case has a bitrate of 28 kbps, with RTP payload packets of 250 bytes (excluding payload header which is, for example, 2 bytes). The total video bit rate is 32.7 kbps (including RTP/UDP/IPv4 headers). The value in the b=AS media level parameter would be 33. In the same bearer there is an AMR stream at 10.2 kbps with 1 frame encapsulated per RTP packet using the bandwidth efficient mode. The total voice bit rate is 26.8 kbps (including RTP/UDP/IPv4 headers). The value in the b=AS media level parameter would be 27. The total media bit rate is 28+10.2=38.2 kbps. The total session bit rate is 33+27=60 kbps.

The terminal would produce IP packets of the following size:

AMR: 20 (IPv4) + 8 (UDP) + 12 (RTP) + 27 (AMR RTP payload) = 67 bytes (or 87 bytes for IPv6 with no extension headers).

Video: 20 (IPv4) + 8 (UDP) + 12 (RTP) + 252 (video RTP payload+payload header) = 292 bytes (or 312 bytes for IPv6 with no extension headers).

The same considerations done in Use Case 2 about the maximum packet sizes apply also for this use case.

The QoS profile of the videotelephony terminal would be set then using the following parameters:

Table B.4: QoS profile for videotelephony at 38.2 kbps

QoS parameter	Parameter value	Comment
Delivery of erroneous SDUs	No	
Delivery order	No	To minimize delay in the access stratum. The application should take care of eventual packet reordering
Traffic class	Conversational	
Maximum SDU size	500 bytes	10 bytes granularity
Guaranteed bitrate for downlink	SDP media bw in DL for AMR + 2.5% * (SDP media bw in DL for AMR+ SDP media bw in UL for AMR) + SDP media bw in DL for video + 2.5% * (SDP media bw in DL for video+ SDP media bw in UL for video) = 63 kbps	
Maximum bit rate for downlink	Equal or higher than guaranteed bit rate	
Guaranteed bitrate for uplink	SDP media bw in UL for AMR + 2.5% * (SDP media bw in UL for AMR+ SDP media bw in DL for AMR) + SDP media bw in UL for video + 2.5% * (SDP media bw in UL for video+ SDP media bw in DL for video) = 63 kbps	
Maximum bit rate for uplink	Equal or higher than guaranteed bit rate	
Residual BER	10^{-5}	16 bit CRC
SDU error ratio	10^{-3}	
Traffic handling priority	Not used in Conversational traffic class~~Subscribed traffic handling priority~~	~~Not relevant~~
Transfer delay	100 ms	
SDU format information	Not used	
Allocation/retention priority	Subscribed allocation/retention~~traffic handling~~ priority	Not relevant for the application
Source statistics descriptor	"Unknown"	

In case of usage of separate PDP contexts for the speech and video streams, the speech stream QoS profile parameters are set similarly to use case 1, while the video stream QoS profile parameters are set similarly to use case 2 (but considering that the video flow is bi-directional and considering possibly the same UMTS bearer transfer delay constraints for both media).