Technical Specification Group Services and System Aspects
Meeting #20, Hämeenlinna, Finland
09-12 June 2003

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Source: **TSG-SA WG4**

Title: CR to TS 26.236 - Examples of QoS profiles for conversational multimedia applications (Release 5)

Document for: **Approval**

Agenda Item: **7.4.3**

The following CR, agreed at the TSG-SA WG4 meeting #26, is presented to TSG SA #20 for approval.

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.236	005		Rel-5	Examples of QoS profiles for conversational multimedia applications	F	5.2.0	S4	TSG-SA WG4#26	S4-030382

CR-Form-v7

CHANGE REQUEST

⌘	**26.236**	**CR**	**005**	⌘ **rev**	**-**	⌘ Current version:	**5.2.0** ⌘

*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:	⌘ Examples of QoS profiles for conversational multimedia applications		
Source:	⌘ TSG SA WG4		
Work item code:	⌘ IMS-CODEC	***Date:*** ⌘	10/06/2003
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)* *Rel-6 (Release 6)*

Reason for change: ⌘	The examples of QoS profiles are only considering IPv4 headers. IMS is using IPv6.
Summary of change: ⌘	The examples are updated to include also IPv6 cases.
Consequences if not approved: ⌘	IMS specifications regarding conversational multimedia applications are incomplete (lack of IPv6 use cases)

Clauses affected: ⌘	Annex B			
	Y	**N**		
Other specs affected: ⌘		X	Other core specifications	⌘ TS 24.228
			Test specifications	
			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.

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 (with values for IPv4 and IPv6 addressing). 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. The gross bit rate including uncompressed RTP/UDP/IPv6 headers would be 36.8 kbps. The value in the b=AS media level parameter would be 37.

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 (for the IPv4 case) Ceil(38.85)=39 kbps (for the IPv6 case)	
Maximum bit rate for downlink	Ceil(30.45)=31 kbps (for the IPv4 case) Ceil(38.85)=39 kbps (for the IPv6 case)	
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 (for the IPv4 case) Ceil(38.85)=39 kbps (for the IPv6 case)	
Maximum bit rate for uplink	Ceil(30.45)=31 kbps (for the IPv4 case) Ceil(38.85)=39 kbps (for the IPv6 case)	
Residual BER	10^{-5}	16 bit CRC
SDU error ratio	$7*10^{-3}$	
Traffic handling priority	Not used in Conversational traffic class	
Transfer delay	100 ms	
SDU format information	Not used	
Allocation/retention priority	Subscribed allocation/retention 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].

In case of IPv6 addressing, the gross bit rates are:

AMR 4.75 kbps: 29.6 kbps (including RTP/UDP/IPv6 headers). [SDP b=AS parameter would be 30].

AMR 10.2 kbps: 34.8 kbps (including RTP/UDP/IPv6 headers). [SDP b=AS parameter would be 35].

AMR 12.2 kbps: 36.8 kbps (including RTP/UDP/IPv6 headers). [SDP b=AS parameter would be 37].

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 (for the IPv4 case) Ceil(36.75)=37 kbps (for the IPv6 case)	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 (for the IPv4 case) Ceil(38.85)=39 kbps (for the IPv6 case)	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 (for the IPv4 case) Ceil(36.75)=37 kbps (for the IPv6 case)~~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 (for the IPv4 case) Ceil(38.85)=39 kbps (for the IPv6 case)~~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	
Transfer delay	100 ms	
SDU format information	Not used	
Allocation/retention priority	Subscribed allocation/retention 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 gross bit rate including uncompressed RTP/UDP/IPv6 headers would be 65.8 kbps. The value in the b=AS media level parameter would be 66.

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 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 (for the IPv4 case) Ceil(67.65)=68 kbps (for the IPv6 case)	
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 the IPv4 case) Ceil(1.65)=2 kbps (for the IPv6 case)	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	
Transfer delay	250 ms	
SDU format information	Not used	
Allocation/retention priority	Subscribed allocation/retention 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. For IPv6 addressing, the total video bit rate is 34.9 kbps (including RTP/UDP/IPv6 headers). The value in the b=AS media level parameter would be 35.

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. For IPv6 addressing, the total voice bit rate is 34.8 kbps (including RTP/UDP/IPv6 headers). The value in the b=AS media level parameter would be 35.

The total media bit rate is 28+10.2=38.2 kbps. The total session bit rate is 33+27=60 kbps for IPv4 addressing, and 35+35=70 kbps for IPv6 addressing.

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) = Ceil(63.0)=63 kbps (for the IPv4 case) = Ceil(73.3)=74 kbps (for the IPv6 case)	
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) = Ceil(63.0)=63 kbps (for the IPv4 case) = Ceil(73.3)=74 kbps (for the IPv6 case)~~= 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	
Transfer delay	100 ms	
SDU format information	Not used	

Allocation/retention priority	Subscribed allocation/retention 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).