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Contents

Foreword

This Technical Specification has been produced by the 3rd Generation Partnership Project (3GPP).

The present document specifies the codec specific RTP protocol details applying to packet switched conversational multimedia applications within the 3GPP IM Subsystem.

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of the present document, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

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where:

- x the first digit:
  - 1 presented to TSG for information;
  - 2 presented to TSG for approval;
  - 3 or greater indicates TSG approved document under change control.
- y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.
- z the third digit is incremented when editorial only changes have been incorporated in the document.

Introduction

The present document contains a specification for required protocol usage within 3GPP specified Conversational Packet Switched Multimedia Services [5] which is based IP Multimedia Subsystem (IM Subsystem). IM Subsystem as a subsystem includes specifically the conversational IP multimedia services, whose service architecture, call control and media capability control procedures have been defined in 3GPP TS 24.229 [7], and are based on the 3GPP adopted version of IETF Session Initiated Protocol (SIP) [1].

In conversational packet switched multimedia service depends on IM Subsystem. The individual media types are independently encoded and packetized to appropriate separate Real Time Protocol (RTP) packets. These packets are then transported end-to-end inside UDP datagrams over real-time IP connections that have been negotiated and opened between the terminals during the SIP call as specified in 3GPP TS 24.229 [7].

The UEs operating within IM Subsystem need to provide encoding/decoding of the derived codecs, and perform corresponding packetization/depacketization functions. Logical bound between the media streams is handled in the SIP session layer, and inter-media synchronization in the receiver is handled with the use of RTP time stamps.

1 Scope

The present document introduces the required protocols for packet switched conversational multimedia applications within 3GPP IP Multimedia Subsystem. Visual and sound communications are specifically addressed. The intended applications are assumed to require low-delay, real-time functionality.

The present document describes the required protocol related elements for 3G PS multimedia terminal:

- required SDP signalling regarding the media type bit rate, packet size, packet transport frequency;
- usage of RTP payload for media types;
- bandwidth adaptation;
- QoS negotiation.

The present document is applicable, but not limited, to packet switched video telephony. All media handling, including codecs, for Multimedia Telephony Service for IMS (MTSI) service is defined in [35]. This specification does not apply to MTSI.

The applicability of the present document to GERAN is FFS.

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 3261: "SIP: Session Initiation Protocol".

[2] IETF RFC 4566: "SDP: Session Description Protocol".

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

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

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

[6] (void)

[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] IETF RFC 4867: "RTP payload format and file storage format for the Adaptive Multi-Rate (AMR) Adaptive Multi-Rate Wideband (AMR-WB) audio codecs", April 2007.

[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] Void

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

[29] IETF RFC 4060 "RTP Payload Formats for European Telecommunications Standards Institute (ETSI) European Standard ES 202 050, ES 202 211, and ES 202 212 Distributed Speech Recognition Encoding".

[30] Open Mobile Alliance: “PoC User Plane Version 1, Draft Version 1.0.10 Nov 2004”, OMA-UP-PoC-V1_0_10-20041103-D.

[31] 3GPP TS 26.103: "Speech codec list for GSM and UMTS".

[32] IETF RFC 4103 "RTP Payload for Text Conversation".

[33] ITU-T Recommendation H.264 (2003): "Advanced video coding for generic audiovisual services" | ISO/IEC 14496-10:2003: "Information technology – Coding of audio-visual objects – Part 10: Advanced Video Coding".

[34] IETF RFC 3984 (2005): "RTP Payload Format for H.264 Video", S. Wenger, M.M. Hannuksela, T. Stockhammer, M. Westerlund and D. Singer.

[35] 3GPP TS 26.114, “IP Multimedia Subsystem (IMS); Multimedia telephony; Media handling and interaction”.

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following term and definition applies:

3G PS multimedia terminal: terminal based on IETF SIP/SDP internet standards modified by 3GPP for purposes of 3GPP IM Subsystem services

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

AMR	Adaptive MultiRate codec
DSR	Distributed Speech Recognition
IETF	Internet Engineering Task Force
IM Subsystem	Internet protocol Multimedia Subsystem
ITU-T	International Telecommunications Union-Telecommunications
MTSI	Multimedia Telephony Service for IMS
NAT	Network Address Translation
RFC	IETF Request For Comments
RTPCP	RTP Control Protocol
RTP	Real-time Transport Protocol
SDP	Session Description Protocol
SES	Speech Enabled Service
SIP	Session Initiation Protocol

4 General

3G PS multimedia terminals provide real-time video, audio, SES or data, in any combination, including none, over 3GPP IM Subsystem. Terminals are based on IETF defined multimedia protocols SIP, SDP, RTP and RTCP. Communication may be either 1-way or 2-way. Such terminals may be part of a portable device or integrated into an automobile or other non-fixed location device. They may also be fixed, stand-alone devices; for example, a video telephone or kiosk. Multimedia terminals may also be integrated into PCs and workstations.

In the case of SES then uplink communication is from the terminal to a server containing speech recognition.

The transmission and reception of audio in Push-to-Talk over Cellular (PoC) communication is controlled by a RTCP APP conveyed Talk Burst Control Protocol defined in OMA PoC User Plane Version 1 [30].

In addition, interoperation with other types of multimedia telephone terminals, such as 3G-324M may be possible, however in such case a media gateway functionality supporting 3G-324M - IM Subsystem interworking will be required within or outside the IM subsystem.

Figure 1 presents the user plane protocol stack of a 3G PS conversational multimedia terminal explaining the transport of different media types and QoS reports.

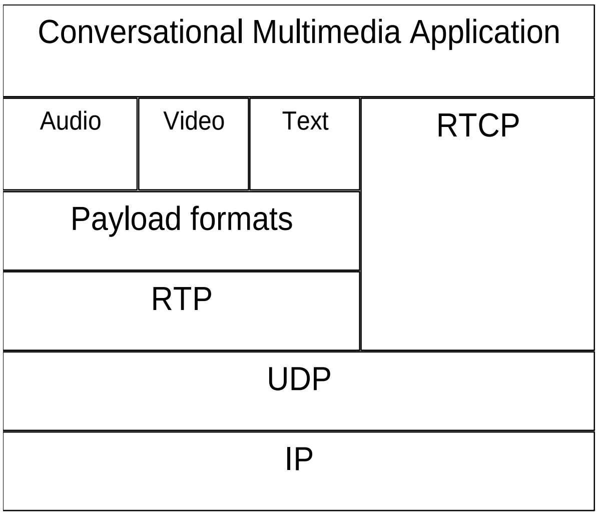

Figure 1 – User plane protocol stack for 3G PS conversational multimedia terminal

5 Media type requirements

Media type RTP payload usage is specified in this clause. The media types and corresponding codecs are specified in 3GPP TS 26.235 [5]. The continuous media type RTP payloads are mapped to RTP packets according to IETF RTP Profile for Audio and Video Conferences with Minimal Control in RFC 3551 [4].

5.1 Audio

The IETF AMR and AMR-WB RTP payload format [19] offers different options. Subclause 5.1.1 describes the use of those options for 3G PS endpoints. Subclause 5.1.2 describes the usage for PoC.

5.1.1 RTP session description parameters

The behaviour of the transmitter is defined below:

- bandwidth efficient and octet aligned operation shall be supported,
- codec mode changes shall be performed in integer multiples of 40 msec,
- codec mode changes should be performed to neighboring modes of the selected combination of codec modes,
- DTX signalling may be used,
- interleaving shall not be used,

The behaviour of the receiver is defined below:

- bandwidth efficient and octet aligned operation shall be supported,
- codec mode changes shall be accepted at any time,
- codec mode changes shall be accepted to any supported mode of the selected combination of codec modes,
- DTX signaling shall always be accepted,

5.1.1.1 Parameter usage in an SDP offer

When using SDP to signal the use of the AMR or AMR-WB payload format, a 3G PS endpoint shall include the following SDP parameters in an SDP offer:

- when redundant operation is offered:

- maxptime: 240

- when non-redundant operation is offered:

    - maxptime: 80

When using SDP to signal the use of the AMR or AMR-WB payload format with two or more modes in the mode-set, a 3G PS endpoint shall also include the following SDP parameter in an SDP offer:

- mode-change-period=2

Unless a 3G PS endpoint can support all possible configurations (= combinations of codec modes) for the codec, it should include in the SDP offer a separate payload type with a mode-set parameter for each configuration it can support.

A 3G PS endpoint should support one or more “preferred configurations” for the codec, as defined in 3GPP TS 26.103 [31].

A 3G PS endpoint should not include in an SDP offer any configuration that contains all but the highest codec mode(s) of another included configuration; the endpoint can support this configuration with the use of rate control to force the use of only the supported modes.

5.1.1.2 Construction of an SDP answer

When using SDP to signal the use of the AMR or AMR-WB payload format, a 3G PS endpoint performs all of the following procedures when constructing an SDP answer from a received SDP offer:

- An SDP answerer should select for a payload type in an SDP answer from a payload type in an SDP offer with the following parameters:

    - crc=0 or no crc parameter;

    - robust-sorting=0 or no robust-sorting parameter;

    - no interleaving parameter, and

    - channels=1 or no channels parameter.

- The SDP answer shall include the following parameters without change from the selected payload type of the SDP offer:

    - octet-align;

    - maxptime;

    - crc;

    - robust-sorting;

    - interleaving, and

    - channels.

- An SDP answerer should select from among payload types in the SDP offer regardless of the presence of the “mode-change-period=2” or “mode-change-neighbor=1” parameters. The SDP answer need not include either parameter.

- If there is no mode-set parameter for a payload type in an SDP offer, the SDP answerer may select any supported mode-set.

- When an SDP offer includes (different) mode-set parameter(s) in one or more payload types, the SDP answerer may select from among any supported mode-set in the SDP offer, including those mode-sets that can be supported with rate control. The mode-set in the SDP answer shall be identical to the mode-set selected from the SDP offer. The SDP answerer shall apply rate control immediately if necessary to limit the use of higher codec modes.

5.1.2 RTP session description parameters for PoC

For PoC services less restrictive IETF AMR and AMR-WB RTP payload format [19] options apply:

- the multi-channel session shall not be used,
- internal CRC shall not be used,
- the number of speech frames encapsulated in each RTP packet should not exceed 20,
- interleaving should not be used.
- The total packetization delay (including any interleaving delay) shall not exceed 500ms.

When the PoC Client uses the AMR-NB or AMR-WB RTP payload format, the PoC Client uses either the bandwidth-efficient mode or the octet-aligned mode of the IETF AMR-NB and AMR-WB RTP payload format. Therefore the PoC Client can construct an SDP offer and answer with the following parameters of RFC 3267 [19]:

- Octect-align=1 or no octet-align parameter;
- Maxptime: up to 400;
- crc=0 or no crc parameter;
- robust-sorting=0 (or no robust-sorting parameter);
- no interleaving parameter, and
- channels=1 or no channels parameter.

5.2 Video

Video packets should not be large to allow better error resilience and to minimize the transmission delay in conversational service. The size of each packet shall be kept smaller than 512 bytes.

5.3 Real time text

Real time text media type RTP payload format for ITU-T Recommendation T.140 is specified in [32]. Redundant transmission provided by the RTP payload format is recommended in error prone channel.

5.4 SES

The RTP payload for the DSR codec and AMR or AMR-WB used for SES are specified in [29, 19].

6 Call control

Functional requirements for call control are specified in 3GPP TS 23.228 [8].

The required signalling functions and call control protocols are specified in 3GPP TS 24.229 [7].

QoS authorization issues and interworking with the IM subsystem in general are covered in 3GPP TS 23.207 [10].

7 Bearer control

The media control is based on declaration of terminal media capability sets in SDP part of appropriate SIP messages. The usage of bearer bandwidth can be effectively controlled by adjusting the media type encoder bit rates.

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. Note that for RTP based applications, 'b=AS:' gives the RTP "session bandwidth" (including UDP/IP overhead) as defined in section 6.2 of [3].

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).

If the session described in the SDP is a point-to-point speech only session (see clause 7.4), the UE should request the deactivation of RTCP by setting its RTCP bandwidth modifier to zero.

If a UE receives SDP bandwidth modifiers for RTCP equal to zero from the originating UE, it should reply (via the SIP protocol) by setting its RTCP bandwidth using SDP bandwidth modifiers with values equal to zero.

7.2 QoS negotiation

The QoS architecture and concept is specified in 3GPP TS 23.107 [9]. The end-to-end QoS framework involving GPRS and UMTS is specified in 3GPP TS 23.207 [10]. The applicable general QoS mechanism and service description for the GPRS in GSM and UMTS is specified in 3GPP TS 23.060 [11].

7.3 RTP receiver

The RTP receiver implementation shall also include an RTCP implementation.

The RTP receiver implementation and functionality including lost and delayed packet processing as well as jitter buffer is out of scope of the present document.

7.4 RTP sender

The RTP sender implementation shall also include an RTCP implementation.

To facilitate traversal of NAT and Firewall gateways, RTP sender implementations should transmit their RTP stream from the same IP address and port on which it has advertised to receive RTP in its SDP. Similarly, RTCP sender implementations should transmit their RTCP stream from the same IP address port on which it has advertised to receive RTCP in its SDP a=rtcp attribute.

RTCP packets should be sent for all types of multimedia sessions except for point-to-point speech only sessions (i.e., using AMR and the AMR-WB codecs where synchronization with other RTP transported media or remote end-point aliveness information are not needed). For point-to-point speech only sessions, a UE should not send RTCP packets. Turning off RTCP can be done by setting to zero the SDP bandwidth modifiers (RR and RS) described in clause 7.1.When RTCP is turned off (for point-to-point speech only sessions) and the media is put on hold, the terminal should re-negotiate the RTCP bandwidth with SDP bandwidth modifiers values greater than zero, and send RTCP packets to the other end, following the rules given below. This allows the remote end to detect link aliveness during hold. When media is resumed, the resuming terminal should turn off the RTCP sending again through a re-negotiation of the RTCP bandwidth with SDP bandwidth modifiers (as described in clause 7.1) equal to zero.

When RTCP is turned off (for point-to-point speech only sessions) and if sending of an additional associated RTP flow becomes required and both RTP flows need to be synchronized, or if transport feedback due to lack of end-to-end QoS guarantees is needed, a terminal should re-negotiate the bandwidth for RTCP by sending an SDP with the RS bandwidth modifier greater than zero.

Note: For speech sessions where RTCP is not turned off, to reduce the potential disruption of RTCP onto the RTP flow, it is beneficial to keep the RTCP bandwidth and the size of RTCP packets as small as possible. RTCP packet size can be minimized by only using the optional parts of RTCP (according to [3]) which are required by the application. A practical size limit for the RTCP sender is in the order of 2 to 5 times the RTP packet size. Additionally, the RTCP sender can attempt to schedule RTCP packets during speech inactivity periods. For example, if an RTCP packet is scheduled at a future time and a silence period starts, this RTCP packet could be sent immediately. The subsequent RTCP packets would be scheduled according to the normal rules (i.e. as if the previous packet was sent as originally scheduled).

Annex A (informative): Optional enhancements

This annex is intended for informational purposes only. This is not an integral part of the present document.

A.1 Video enhancements

This clause gives informative recommendations for the video media type control.

The SDP attributes regarding the video frame rate and the quality of media encoding should be used to ensure good video service. The recommended usage of these attributes are FFS.

`a=framerate:<frame rate>` describes the maximum video frame rate attribute in frames/second. Fractional values of `<frame rate>` are allowed.

`a=quality:<quality>` describes the quality of media encoding attribute, where the `<quality>` is a value in [0..10] with 10 indicating the best quality.

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 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
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
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)	Guaranteed quality 10.2 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)	Non-guaranteed quality 12.2 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, H.264 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/H.264/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 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)	
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).

Annex C (informative): Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
2002-03	15	SP-020074			Version 2.0.0 presented for approval	2.0.0	5.0.0
2002-12	18	SP-020695	001	2	QoS profile parameters for conversational multimedia applications	5.0.0	5.1.0
2002-12	18	SP-020695	002	1	Clarification on SDP session bandwidth parameter	5.0.0	5.1.0
2003-03	19	SP-030092	003	2	SDP bandwidth modifier for RTCP bandwidth	5.1.0	5.2.0
2003-03	19	SP-030092	004		Correction on QoS profile parameters for conversational multimedia applications	5.1.0	5.2.0
2003-06	20	SP-030219	005		Examples of QoS profiles for conversational multimedia applications	5.2.0	5.3.0
2003-09	21	SP-030449	006		Correction of obsolete RTP references	5.3.0	5.4.0
2003-09	21	SP-030449	007	1	Correction of wrong reference	5.3.0	5.4.0
2004-06	24	SP-040356	010	3	Introduction of the DSR codec	5.4.0	6.0.0
2004-06	24	SP-040357	012		RTCP usage for IMS	5.4.0	6.0.0
2004-12	26	SP-040843	013	1	Inclusion of PoC support	6.0.0	6.1.0
2005-03	27	SP-050099	015	1	Introduction of AMR SDP parameters	6.1.0	6.2.0
2005-06	28	SP-050249	017	1	Clarification to the Introduction of AMR SDP parameters	6.2.0	6.3.0
2005-09	29	SP-050424	0018	1	Clarifications on RTP Session Description Parameters for PoC	6.3.0	6.4.0
2006-03	31	SP-060219	0019	3	Update of a RFC on real time text and DSR codec	6.4.0	7.0.0
2006-09	33	SP-060598	0021	1	Symmetric RTP and RTCP Port Usage	7.0.0	7.1.0
2007-09	37	SP-070627	0023		Correction of references	7.1.0	7.2.0
2008-03	39	SP-080009	0024	1	Correction of RTP usage for audio for PS conversational multimedia applications	7.2.0	7.3.0
2008-12	42				Version for Release 8	7.3.0	8.0.0
2009-12	46				Version for Release 9	8.0.0	9.0.0
2011-03	51				Version for Release 10	9.0.0	10.0.0
2012-09	57				Version for Release 11	10.0.0	11.0.0