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Packet Switched Conversational Multimedia Applications;

Transport Protocols (Release 5)

Agenda Item: **7.4.3**

Presentation of Specification to TSG SA Plenary

Presentation to: **TSG SA Meeting #15**

Document for presentation: **TS 26.236, Version 2.0.0**

Presented for: **Approval**

Abstract of document:

This document contains updated draft transport protocol specification for the PS conversational multimedia service.

Changes since last presentation:

This is version 2.0.0, presented to TSG SA Plenary for approval.

The new version contains removal of editor’s notes, correction to references on RTP payloads, the restriction to the usage of multi-channel transport of AMR and AMR-WB and mapping of SDP parameters to UMTS QoS parameters.

Outstanding Issues:

None.

Contentious Issues:

None.

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Technical Specification

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Keywords

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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:

Version x.y.z

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.

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 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".

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
IETF	Internet Engineering Task Force
IM Subsystem	Internet protocol Multimedia Subsystem
ITU-T	International Telecommunications Union-Telecommunications
RFC	IETF Request For Comments
RTPCP	RTP Control Protocol
RTP	Real-time Transport Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol

4 General

3G PS multimedia terminals provide real-time video, audio, 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 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.

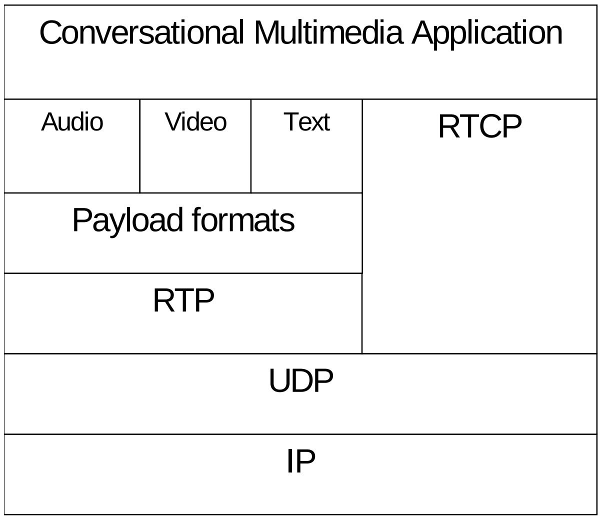

Figure 1 – User plane protocol stack for 3G PS conversational multimedia terminal

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 1890 [4].

5.1 Audio

5.1.1 RTP session description parameters

The IETF AMR and AMR-WB RTP payload format [19] offers different options. Here is the list of options and how they should be used by the transmitter. The receiver shall at least support the options as they are listed:

- the bandwidth efficient operation shall be used,
- only one speech frame shall be encapsulated in each RTP packet,
- the multi-channel session shall not be used,
- interleaving shall not be used,
- internal CRC shall not be used.

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 [27]. Redundant transmission provided by the RTP payload format is recommended in error prone channel.

6 Call control

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

The required signalling functions are specified in 3GPP TS 24.228 [6] and call control protocols 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.

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

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.

Table B.1 Mapping for conversational application

QoS parameter	Parameter value	comment
Delivery of erroneous SDUs	"no"	
Delivery order	Yes	
Traffic class	"Conversational class"	
Maximum SDU size	[TBD]	
Guaranteed bit rate for downlink	SDP media bw in downlink direction + 2.5% (media bw in downlink + media bw in uplink direction)	Per media type
Maximum bit rate for downlink	Equal or higher to guaranteed bit rate in downlink	
Guaranteed bit rate for uplink	SDP media bw in uplink direction + 2.5% (media bw in downlink + media bw in uplink direction)	Per media type
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	7*10-3 or less for AMR-NB and AMR-WB 10-4 for the rest	
Traffic handling priority	Subscribed traffic handling priority	Ignored
Transfer delay	100 ms AMR (NB and WB) 150 ms H.263 and MPEG 4 video [TBD] others	Target values

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