Presentation of Specification to TSG

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Technical Specification

3rd Generation Partnership Project;
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Contents

Foreword

The 3GPP has produced this Technical Specification.

The present document introduces the set of specifications for protocols and codecs, which apply to the packet-switched streaming service within the 3GPP system.

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 this TS, 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 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 specification;

Introduction

Streaming refers to the ability of an application to play synchronised media streams like audio and video streams in a continuous way while those streams are being transmitted to the client over a data network.

Applications, which can be built on top of streaming services, can be classified into on-demand and live information delivery applications. Examples of the first category are music and news-on-demand applications. Live delivery of radio and television programs are examples of the second category.

The 3GPP packet-switched streaming service (PSS) provides a framework for IP-based streaming applications in 3G networks.

Scope

This document specifies protocols and codecs of the 3GPP PSS. A general description of the 3GPP PSS is given in [1].

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.

[1] 3GPP TS 26.233: “General Description”

[2] AMR narrow-band ?

[3] IETF RFC 2326: “Real Time Streaming Protocol (RTSP)”, Schulzrinne H., Rao A. and Lanphier R.

[4] IETF RFC 2327: “SDP: Session Description Protocol”, Handley M. and Jacobson V.

[5] ITU-T Recommendation H.263: "Video coding for low bitrate communication"

[6] W3C Working Draft Recommendation: “SMIL 2.0”,http://www.w3.org/TR/smil20/

[7] W3C Proposed Recommendation: “XHTML Basic”, http://www.w3.org/TR/2000/PR-xhtml-basic-20001103/

[8] 3GPP TR 26.911: “Terminal Implementor’s Guide”

[9] ITU-T Recommendation H.263: "Annex X, Profiles and Levels Definition"

[10] IETF RFC 2429: “RTP Payload Format for the 1998 Version of ITU-T Rec. H.263 Video (H.263+)”, ??

[11] International Standard ISO/IEC 14496-3: “Information technology - Generic coding of audio-visual object – Part 3: Audio, 1999

[12] International Standard ISO/IEC 14496-2: “Information technology - Generic coding of audio-visual object – Part 2: Visual, 1999

[13] IETF RFC 3016: “RTP payload format for MPEG-4 Audio/Visual streams”, Kikuchi Y., et al.

[14] UCS-2/Unicode???

[15] ITU-T Recommendation T.81 “Digital compression and coding of continuous-tone still images – requirements and guidelines”, (9/91)

[16] IETF RFC XXX: “HTTP”

[17] IETF RFC XXXX: “RTP payload format for AMR”, ???

[18] JFIF???

[19] ISO/IEC JTC1/SC 29/WG11 N3670, La Baule, October, 2000

[Editor’s Note: Reference [19] needs to be updated]

Abbreviations

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

AAC	Advanced Audio Coding
AMR	Adaptive Multi-Rate codec
FFS	For Further Studies
GIF	Graphics Interchange Format
HTTP	Hyper Text Transport Protocol
IETF	Internet Engineering Task Force
ITU-T	International Telecommunications Union - Telecommunications
IP	Internet Protocol
JPEG	Joint Photographic Experts Group
MMS	Multimedia Messaging Service
MPEG	Moving Pictures Experts Group
PSS	Packet-switched Streaming Service
RFC	IETF Request For Comments
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
RTSP	Real-Time Streaming Protocol
SDP	Session Description Protocol
SMIL	Synchronised Multimedia Integration Language
TCP	Transport Control Protocol
UDP	User Datagram Protocol

System description

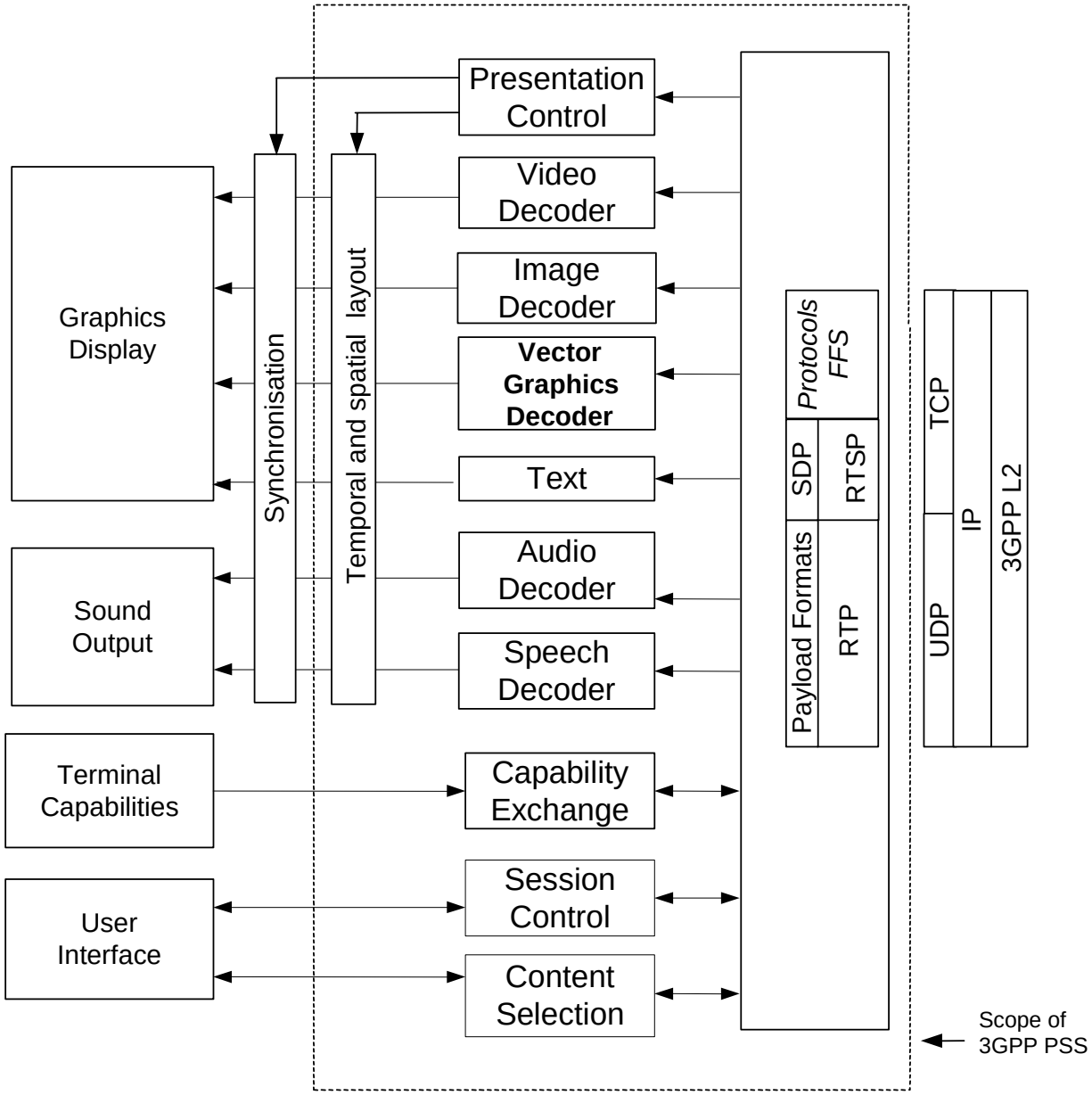

Figure 1: Functional components of a 3GPP packet switched mobile streaming terminal

Streaming services require some control components namely presentation control, capability exchange, session control and content selection.

Presentation control mechanisms are needed for specifying the user interface layout, for defining the particular media streams, which compose a media presentation, and for the synchronisation of those media streams.

Capability exchange methods should enable automatic choice or adaptation of media streams depending on different terminal capabilities.

Session set-up and control deals with the set-up of the streaming session between the streaming server and client and the control of media presentation by the user. It involves VCR-like presentation control functions like start, pause, fast forward and stop of a media presentation.

Session Establishment refers to method to invoke the streaming client from a browser or directly by the user..

Protocols

Packet based network interface

The transport of session control and media data in the 3GPP PSS is intended for IP-based networks.

Capability exchange

FFS

Session establishment

FFS

Session set-up and control

RTSP [3] shall be used for session set-up and session control. 3GPP PSS clients and servers shall follow the rules for minimal [on-demand playback] RTSP implementations in Appendix D of [3]. In addition to this, 3GPP PSS servers and clients shall implement the DESCRIBE method.

[Editors note: Additional support for headers and methods will be decided on later. A proposal is given in the document S4-000509.]

SDP shall be used as the format of the presentation description for both clients and servers. Servers shall generate and clients interpret the SDP syntax according to the SDP specification [4] and Appendix C of [3].

[Editor's note: Further studies will show if additional SDP attributes needs to be defined. It has also been suggested to require terminals to use the bandwidth information in SDP (both on the session and media level). The working assumption is that the terminal shall interpret the bandwidth information in SDP and the server should include this information in the SDP.]

Data transport

[Editor's note: For release 4 only unicast is considered.]

RTP over UDP/IP

RTP provides a means for sending real-time or streaming data over UDP. It is the preferred way of sending audio and video data. It also provides feedback to the server about the transmission quality. Each media has a payload specification defined or to be defined by IETF. RTP/UDP/IP transport of the following media shall be supported:

- Speech (according to section 6.1.2)
- Audio (according to section 6.1.3)
- Video (according to section 6.1.7)

HTTP over TCP/IP

TCP provides reliable transport of data over IP networks, but with no delay guarantees. It is the preferred way for sending some type of data, e.g. text and still images.

HTTP/TCP/IP [16] shall be used for the transport of the presentation layout and synchronisation description (according to section 7) and for static media. HTTP/TCP/IP transport of the following static media shall be supported:

- Text (according to section 6.1.1)
- Still images (according to section 6.1.6)

[Editor's note: Working assumption. The list may be expanded]

Codecs

Text

[Text shall be formatted according to XHTML Basic [7]. The following character encoding shall be supported:

- UTF-8.
- UCS-2/Unicode[14]]

Speech

[The AMR codec shall be used for narrow-band speech [2]. The RTP payload format defined in [17] shall be used.]]

Audio

[Editor's Note: The working assumption is to have support for MPEG-4 AAC. The RTP payload format used shall be according to [13]. The following areas must be clarified: object types, IPMP, Scalability and transport protocols]

Bitmap Graphics

[GIF89a and GIF87a should be supported]

Vector graphics codecs

FFS

Still images

ISO/IEC JPEG [15] and JFIF[18] shall be supported. [The following two modes of, ISO/IEC JPEG shall be supported:

- Baseline DCT, non-differential, Huffman coding, as defined in Table B.1, Symbol 'SOF0' [15]
- Progressive DCT, non-differential, as defined in Table B.1, Symbol 'SOF2' [15]]

[Editor's note: Exact mode to support is for further study].

Video

ITU-T H.263 baseline [5] shall be supported. H.263 profile 3 level 10 [9] and MPEG-4 Visual Simple Profile Level 0 [12] [19] should be supported. The RTP payload format defined for MPEG-4 Visual in [13] shall be used. The RTP payload format defined for H.263 in [10] shall be used.]

[Editors note: . Need to include reference to FLC value.]

Presentation layout and synchronisation

[SMIL 2.0 Basic profile

[6] shall be used for presentation layout and synchronisation.]

[Editor's note: Working assumption is that SMIL 2.0 Basic profile will be used as a starting point FFS.]

Annex A: Informative Annex on protocols

The section is intended for informational purposes only.

A.1: SDP

This section gives some background information on the Session Description Protocol [4].

The session description protocol has the following items that can be identified in a file.

Type	Description	Requirement	3GPP Recommendations
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	O	O
b	Bandwidth information	O	**R**
z	time zone adjustments	O	O
k	encryption key	O	O
a	session attributes	O	O
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	O	O
b	Bandwidth information	O	**R**
k	Encryption Key	O	O
a	Attribute Lines	O	O

R = Required, O = Optional

[Editor's note: The table will be aligned with the normative text.]

Below is an example SDP file, which can be transmitted to the client.

```
v=0
o=kgofron  2890844526 2890842807 IN IP4 192.168.10.10
s=3GPP SDP File Example
i=Example for 3GPP of a Session Description Protocol file
u=http://www.ccrl.mot.com/ae600/
e=gofron@labs.mot.com
c=IN IP4 228.1.3.3/64
t=0 0
m=video 1024 RTP/AVP 40
a=rtpmap:40 H263-1998/90000
a=recvonly
b=AS:128
```

A second SDP file example is given below for unicast data to a client

```
v=0
o=kgofron 2890844526 2890842807 IN IP4 192.168.10.10
s=3GPP Unicast SDP Example
i=Example of Unicast SDP file
u=http://www.ccrl.mot.com/ae600
e=gofron@labs.mot.com
c=IN IP4 192.168.30.29
t=0 0
m=video 1024 RTP/AVP 40
a=rtpmap:40 H263-1998/90000
a=recvonly
b=AS:128
```

Annex B: Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**