Technical Specification Group Services and System Aspects
Meeting #11, Palm Springs, CA, USA, 19-22 March 2001

TSGS#11(01)0079

3GPP TSG SA2#17
Gothenberg, Sweden

26/2-2/3/2001

S2-010748

Title: **LS on Packet Streaming Service Architecture**
Source: **SA 2**
To: **SA 4**
Cc: **TSG SA**

Contact Person: Chris Pudney
Email: chris.pudney@vf.vodafone.co.uk

TSG SA has tasked SA 2 to review SA 4's Packet Streaming Service Specifications in 26.233 and 26.234.

SA 2 has reviewed 26.233 v1.1.2 and has the attached suggestions for improvements: see S2-010738.

SA 2 has reviewed 26.234 version 1.2.0 and has no comments to it.

SA 2 ask SA 4 to try to incorporate the attached comments into 26.233.

3GPP TSG- SA2 ***S2-010738***
Gothenburg, Sweden, 26th February – 2nd March 2001

CR-Form-v3

CHANGE REQUEST

⌘ **26.233**	**CR**	**QX_1**	⌘ rev	**-**	⌘ Current version:	**1.1.2** ⌘

For ***HELP*** *on using this form, see bottom of this page or look at the pop-up text over the ⌘ symbols.*

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

Title: ⌘	Completion of Procedures for Packet Switched Streaming Service		
Source: ⌘	Vodafone		
Work item code: ⌘	SA2-SA4 PSS	***Date:*** ⌘	28 February 2001
Category: ⌘	B	***Release:*** ⌘	REL-4
	Use one of the following categories: ***F*** *(essential 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: ⌘	S4 need S2 to review their specifications for the packet switched streaming service. These are 26.234 (currently v1.2.0) and 26.233 (currently v1.1.2). From an SA 2 perspective, Vodafone believes that 26.234 requires no changes. For 26.233, Vodafone believes that SA 2 should recommend to SA 4 that the attached changes (or similar changes) are made to 26.233 v1.1.2. Assuming that SA 2 agree with Vodafone's analysis, then it is proposed to send this document with an appropriate liaison statement to SA 4, who are meeting this week in Sophia Antipolis.
Summary of change: ⌘	Figure 1 (section 4.2.1) shows "streaming RABs" being activated by the mobile. It is more accurate to show this as secondary PDP context activation/PDP context modification. Context deactivation ought to be shown as well. It seems sensible to mandate that the mobile attempts to obtain a bearer with streaming QoS (eg by the activation of an additional PDP context) and the text in 4.2.1 is modified to reflect this. From an SA 2 perspective, Vodafone suspects that charging and security are requirements and hence we suggest that they are marked as FFS (unless SA 5 and SA 3 have already been consulted). Reference 1 (26.234) only specifies the codecs and protocols for the Packet Switched Streaming service. It does not contain information relating to 23.060, 23.108, 24.008 etc and hence the text is modified in several places to indicate that 26.234 only specifies PSS protocols and codecs.
Consequences if not approved: ⌘	R4 features slip to R5.

Clauses affected:	⌘	Foreword, Scope, 4.2.1, 6.1; 8.3 and 9.		
Other specs Affected:	⌘	☐	Other core specifications	⌘
		☐	Test specifications	
		☐	O&M Specifications	
Other comments:	⌘			

3GPP TS 26.233 V1.1.2 (2001-02)

Technical Specification

3rd Generation Partnership Project;
Technical Specification Group Services and System Aspects;
Transparent End-to-End Packet Switched Streaming Services (PSS);
General Description
(Release 4)

Keywords

Streaming, multimedia, codec, protocols, packet-switched

3GPP

Postal address

3GPP support office address

650 Route des Lucioles - Sophia Antipolis
Valbonne - FRANCE
Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Internet

http://www.3gpp.org

Contents

Foreword

The 3rd Generation Partnership Project (3GPP) Technical Specification Group (TSG) Services and Systems Aspects has produced this Technical Specification.

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 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;

The 3GPP packet-switched streaming service (PSS) specification consists of two 3G TSs; 3GPP TS 26.234 "Transparent End-to-end Packet-switched Streaming Services (PSS); Protocols and Codecs" [1] and the present document. The present document provides an overview of the 3GPP PSS and [1] specifies the set of PSS protocols and codecs used by the service.

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.

Streaming over fixed-IP networks is already a major application today. While IETF and W3C have developed a set of protocols used in fixed-IP streaming services, no complete standardised streaming framework has yet been defined. For 3G systems, the 3G packet-switched streaming service (PSS) fills the gap between 3G MMS, e.g. downloading, and conversational services.

PSS enables mobile streaming applications, where the protocol and terminal complexity is lower than for conversational services, which in contrast to a streaming terminal require media input devices, media encoders and more complex protocols.

This document describes the transparent 3G packet-switched streaming services (3G PSS) on a general application level.

1 Scope

The present document contains a general description of a transparent packet-switched streaming service in 3G networks. In particular, it defines the usage scenarios, overall high level end-to-end service concept, and lists terminal related functional components. It also lists any identified service interworking requirements. PSS protocols and codecs are defined in [1].

2 References

[1] 3GPP TS 26.234, Packet-switched streaming services, Protocols and Codecs.

[2] H. Schulzrinne S. Casner et al., “RTP: A Transport Protocol for Real-Time Applications”, RFC1889, January 1996.

[3] H. Schulzrinne , A. Rao, R. Lanphier, “Real Time Streaming Protocol (RTSP)”, RFC2326, April 1998.

[4] M. Handley and V. Jacobson, “SDP: Session Description Protocol”, RFC 2327, April 1998.

3 Abbreviations

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

DRM	Digital Rights Management
FFS	For Further Study
GIF	Graphics Interchange Format
HTML	HyperText Markup Language
IETF	Internet Engineering Task Force
IP	Internet Protocol
MMS	Multimedia Messaging Service
PSS	Packet-switched Streaming Service
RAB	Radio Access Bearer
RFC	IETF Request For Comments
RTP	Real-time Transport Protocol
RTSP	Real-Time Streaming Protocol
SDP	Session Description Protocol
TCP	Transport Control Protocol
UDP	User Datagram Protocol
URI	Universal Resource Identifier
WAP	Wireless Application Protocol
WWW	World Wide Web

4 Usage scenarios

4.1 Applications

The streaming platform supports a multitude of different applications including streaming of news at very low bitrates using still images and speech, music listening at various bitrates and qualities, video clips and watching live sports events.

4.2 Use case descriptions

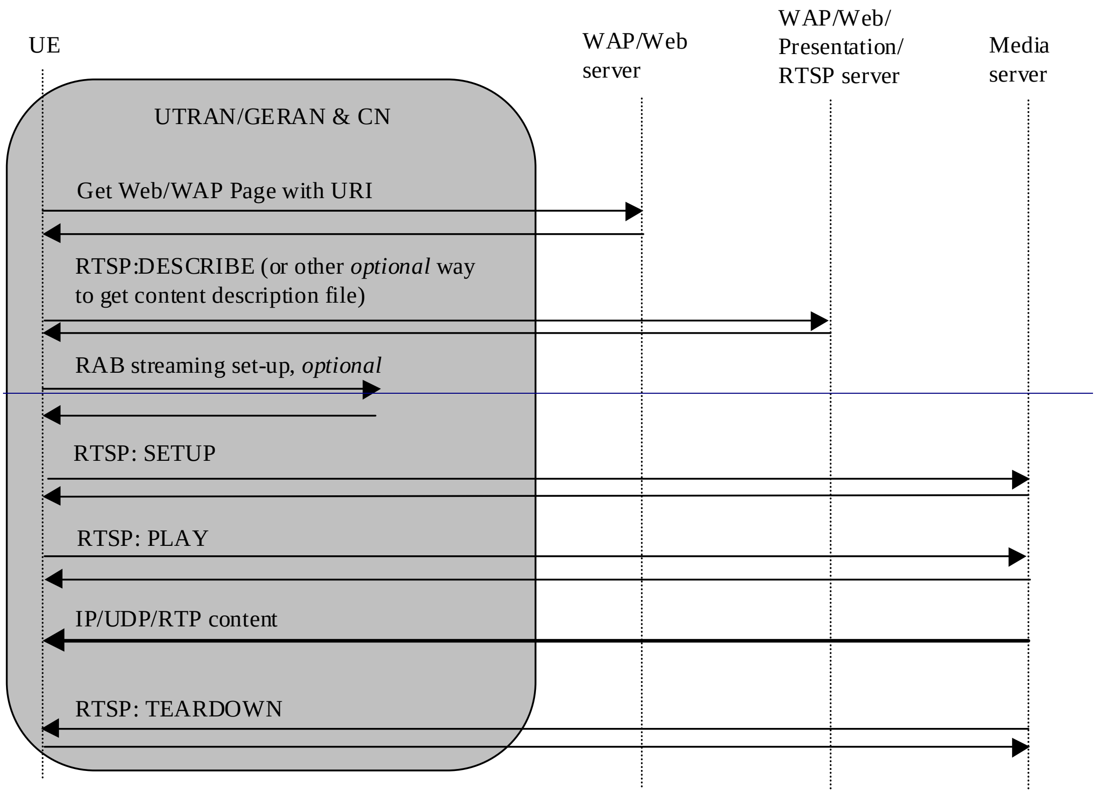
4.2.1 Simple streaming

The simple streaming service includes a basic set of streaming control protocols, transport protocols, media codecs and presentation and layout control protocol. In this simple case, there is neither explicit capability exchange, nor any encryption or digital rights management.

A mobile user gets a URI to specific content that suits his or her terminal. This URI may come from a WWW-browser, a WAP-browser, or typed in by hand. This URI specifies a streaming server and the address of the content on that server. An application that establishes the multimedia session should understand a Session Description Protocol (SDP) file. The SDP file may be obtained in a number of ways. It may be provided in a link inside the HTML page that the user downloads, via an embed tag. It may also be directly obtained by typing it as a URI. It may also be obtained through RTSP [3] signalling via the DESCRIBE method. The SDP file contains the description of the session (session name, author, …), the type of media to be presented, and the bitrate of the media.

The session establishment is the process in which the browser or the mobile user invokes a streaming client to set up the session against the server. The client may be able to ask for more information about the content. ~~It may also choose to set up a specific bearer for the streaming media.~~ The client shall initiate the provisioning of a bearer with appropriate QoS for the streaming media. The set up of the streaming service is done by sending an RTSP SETUP message for each media stream chosen by the client. This returns the UDP and/or TCP port etc. to be used for the respective media stream. The client sends a RTSP PLAY message to the server that starts to send one or more streams over the IP network.

This case is illustrated below in figure 1.



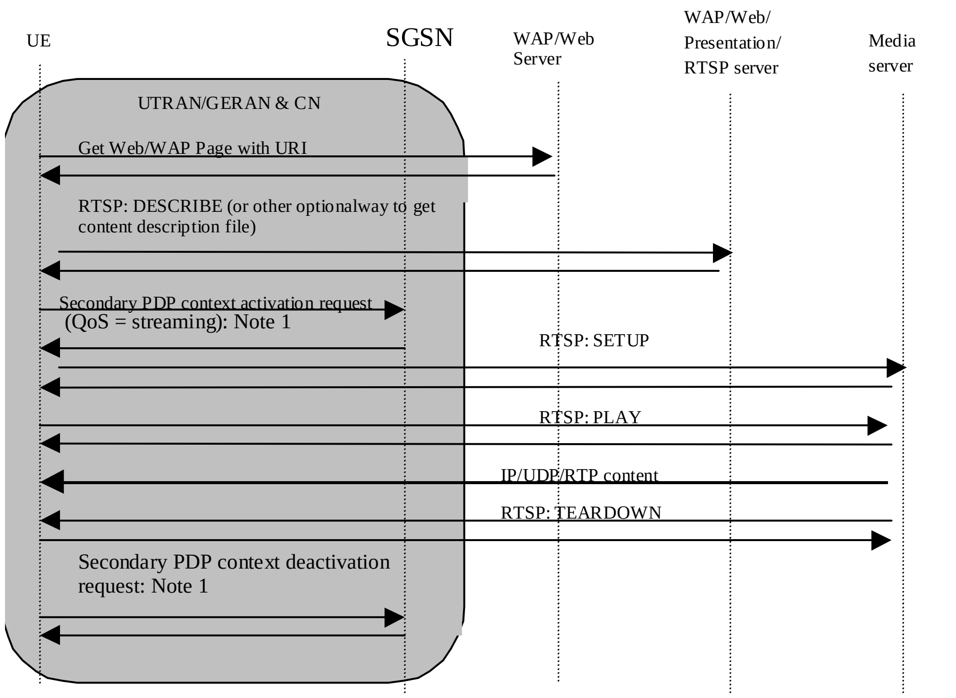


Figure 1: Schematic view of a simple streaming session

Note 1: These messages are one example of how to establish and terminate the bearer with the desired QoS. Other alternatives exist eg an existing PDP context might be modified.

4.2.2 Other streaming cases

[Extended streaming service will support all features defined for the simple streaming case and additionally includes capability exchange, interworking with core network services, security and Digital Rights Management. These cases are FFS.]

[Editor's Note: The two-phase approach for PSS has been agreed. Exact split of features between the two cases is to be decided.]

5 General service architecture

Figure 2 shows the most important service specific entities involved in a 3G packet -switched streaming service. A streaming service requires at least a content server and a streaming client. A streaming server is located behind the Gi interface. Additional components like portals, profile servers, caching servers and proxies located behind the Gi interface might be involved as well to provide additional services or to improve the overall service quality.

Portals are servers allowing convenient access to streamed media content. For instance, a portal might offer content browse and search facilities. In the simplest case, it is simply a Web/WAP-page with a list of links to streaming content. The content itself is usually stored on content servers, which can be located elsewhere in the network.

User and terminal profile servers are used to store user preferences and terminal capabilities. This information can be used to control the presentation of streamed media content to a mobile user.

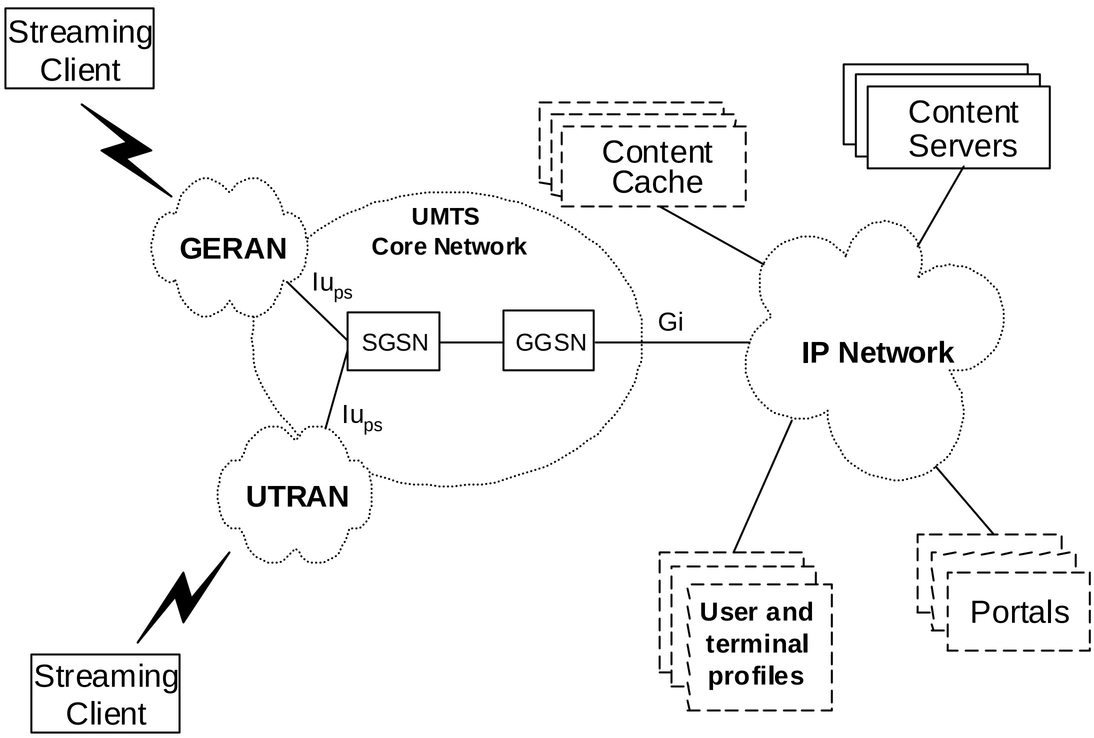


Figure 2: Network elements involved in a 3G packet switched streaming service

6 Functional components of a PSS terminal

This chapter lists the 3G packet-switched streaming service components, which belong to the terminal. Note that not all of the components need to be mandatory. The functional behaviour of the different components is discussed in the following.

6.1 Session protocols and data transport

Protocols are needed for PSS session establishment, session setup, session control, capability exchange, and data transport of streaming media and other data. The PSS protocols to be used are specified in [1].

6.2 Codecs

Codecs are needed for speech, audio, video, still images, bitmap graphics, vector graphics and text. The codecs to be used are specified in [1].

7 File format

The file format is an important element of the content manipulation chain. Conceptually, there is a difference between the coding format and the file format. The coding format is related to the action of a specific coding algorithm that codes the content information into a codestream. The file format is instead a way of organising the prestored codestream in such way that it can be accessed for local decoding and playback, or transferred as a file on different media, or streamed over different transport. Some file formats are optimised for one or more of these functions, others aim instead at achieving a higher flexibility.

When a single media type is involved, the coding and the file format are often considered, and referred to, as a single entity. When multimedia information is involved, instead, it is appropriate to maintain, at least conceptually, the distinction between these two instances. The file format can play an important role in facilitating the organisation and the access to the coded information, independently of the specific coding formats.

[Editor's note: SA4 may come up with a decision to recommend a specific (optional) file format to be used in the PSS compatible content servers providing prestored content, to enable maximum flexibility in serving the codestreams for the end user. There is no SA4 decision yet whether such recommendation should be already in the simple streaming (Release 4) or in the extended service (Release 5).]

8 Interworking with other core network services

[Editor's note: Relation to MexE, including UAProf work performed by TSG-T2 is to be clarified.]

8.1 Interworking with WAP

FFS

8.2 Interworking with MMS

[TS 23.140 defines a new optional feature for the MMS service, which enables streaming of the MMS messages by the message recipient. The MMS streaming option uses the codecs and protocols in accordance with TS 26.234.

Additionally, 23.140 mandates the use of the interchange format recommendation specified in 26.234, clause 9 for MMS purposes.]

[Editor's note: the square brackets are to be removed, and the right references are to be checked after SA4 receives the official T2 answer to LS S4-010078, and the target specification of the file format text has been agreed between the working groups.]

8.3 Interworking with charging/billing services

~~Not required.~~FFS.

9 Security

~~Not required.~~ FFS.

10 Digital Rights Management

Standardisation of 3G packet switched streaming services need to be aligned with standardised or industry solutions for media rights management. The specification of an appropriate DRM framework is FFS.

Annex A (informative): Change history

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

3GPP TS 26.234 V1.2.0 (2000-12)

Technical Specification

3rd Generation Partnership Project;
Technical Specification Group Services and System Aspects;
Transparent end-to-end Packet-switched Streaming Service (PSS);
Protocols and codecs
(Release 4)

Editor's note: Select keywords from list provided in specs database.

Keywords

<keyword[, keyword]>

3GPP

Postal address

3GPP support office address

650 Route des Lucioles - Sophia Antipolis
Valbonne - FRANCE
Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Internet

http://www.3gpp.org

Contents

Foreword

The 3rd Generation Partnership Project (3GPP) Technical Specification Group (TSG) Services and Systems Aspects has produced this Technical Specification.

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 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;

The 3GPP packet-switched streaming service (PSS) specification consists of two 3G TSs; 3GPP TS 26.233 "Packet-switched Streaming Services (PSS); General description" [2] and the present document. The first TS provide an overview of the 3GPP PSS and the present document the details of protocol and codecs used by the service.

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 PSS provides a framework for IP-based streaming applications in 3G networks.

1 Scope

The present document specifies the protocols and codecs for the PSS within the 3GPP system. Protocols for both control signalling, media transport and media encapsulations are specified. Codecs for speech, [audio,] video, still images, bitmap graphics, [vector graphics] and text are specified.

The present document is applicable to Internet Protocol (IP) based packet-switched networks.

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.

This specification may contain references to pre-Release-4 GSM specifications. These references shall be taken to refer to the Release 4 version where that version exists. Conversion from the pre-Release-4 number to the Release 4 (onwards) number is given in subclause 6.1 of 3GPP TR 41.001[1].

Editor's note: [<seq>] <doctype> <#>[([up to and including]{yyyy[-mm]|V<a[.b[.c]]>}[onwards])]: "<Title>".

[1] 3GPP TR 41.001: "GSM Release specifications".

[2] 3GPP TS 26.233: "3rd Generation Partnership Project; Technical Specification Group Services and Systems Aspects; Packet-switched Streaming Services (PSS); General description".

[3] 3G TR 21.905: "3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Vocabulary for 3GPP Specifications".

[4] IETF RFC 2326: "Real Time Streaming Protocol (RTSP)", Schulzrinne H., Rao A. and Lanphier R., April 1998.

[5] IETF RFC 2327: "SDP: Session Description Protocol", Handley M. and Jacobson V., April 1998

[6] IETF STD 0006: "User Datagram Protocol", Postel J., August 1980

[7] IETF STD 0007: "Transmission Control Protocol", September 1981

[8] IETF RFC 1889: " RTP: A Transport Protocol for Real-Time Applications ", Schulzrinne H. et al., January 1996.

[9] IETF RFC ????: "RTP payload format for AMR", Sjoberg J., et al., ?? 2001

[10] IETF RFC 3016: "RTP payload format for MPEG-4 audio/visual streams", Kikuchi Y. et al., November 2000.

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

[12] IETF RFC 2616: "Hypertext Transfer Protocol – HTTP/1.1", Fielding R. et al., June 1999.

[13] 3GPP TS 26.071: "3rd Generation Partnership Project; Technical Specification Group Services and Systems Aspects; Mandatory Speech Codec speech processing functions, AMR Speech Codec; General Description"

[14] 3GPP TS 26.171: "3rd Generation Partnership Project; Technical Specification Group Services and Systems Aspects; AMR Wideband Speech Codec; General description".

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

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

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

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

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

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

[21] "JPEG File Interchange Format", Version 1.02, September 1 1992.

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

[23] ISO/IEC 10646-1: "2000 Information technology – Universal Multiple-Octet Coded Character Set (UCS) – Part1: Architecture and Basic Multilingual Plane"

[24] The Unicode Consortium: "The Unicode Standard", Version 3.0 Reading, MA, Addison-Wesley Developers Press, 2000, ISBN 0-201-61633-5.

[25] W3C Working Draft Recommendation: "Synchronised Multimedia Integration Language (SMIL 2.0) Specification ", http://www.w3.org/TR/smil20/

[26] CompuServe Incorporated: "GIF Graphics Interchange Format: A Standard defining a mechanism for the storage and transmission of bitmap-based graphics information", Columbus, OH, USA, 1987.

[27] CompuServe Incorporated: "Graphics Interchange Format: Version 89a", Columbus, OH, USA, 1990.

[Editor’s Note: Reference [19], [9] needs to be updated.]

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply.

presentation description: as specified in clause 1.3 of RFC 2326 [4]

PSS client: a terminal for the 3GPP packet-based streaming service based on the IETF RTSP/SDP and/or HTTP standards, with possible additional 3GPP requirements according to the present document

PSS server: a server for the 3GPP packet-based streaming service based on the IETF RTSP/SDP and/or HTTP standards, with possible additional 3GPP requirements according to the present document

3.2 Abbreviations

For the purposes of the present document, the abbreviations given in TR 3G 21.905 [3] and the following apply.

AAC Advanced Audio Coding
DCT Discrete Cosine Transform
GIF Graphics Interchange Format
IANA Internet Assigned Numbers Authority

ITU-T	International Telecommunications Union – Telecommunications
JFIF	JPEG File Interchange Format
MIME	Multipurpose Internet Mail Extensions
MMS	Multimedia Messaging Service
PSS	Packet-switched Streaming Service
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
TBD	To Be Decided
UCS-2	Universal Character Set (the two octet form)
UTF-8	Unicode Transformation Format (the 8-bit form)

4 System description

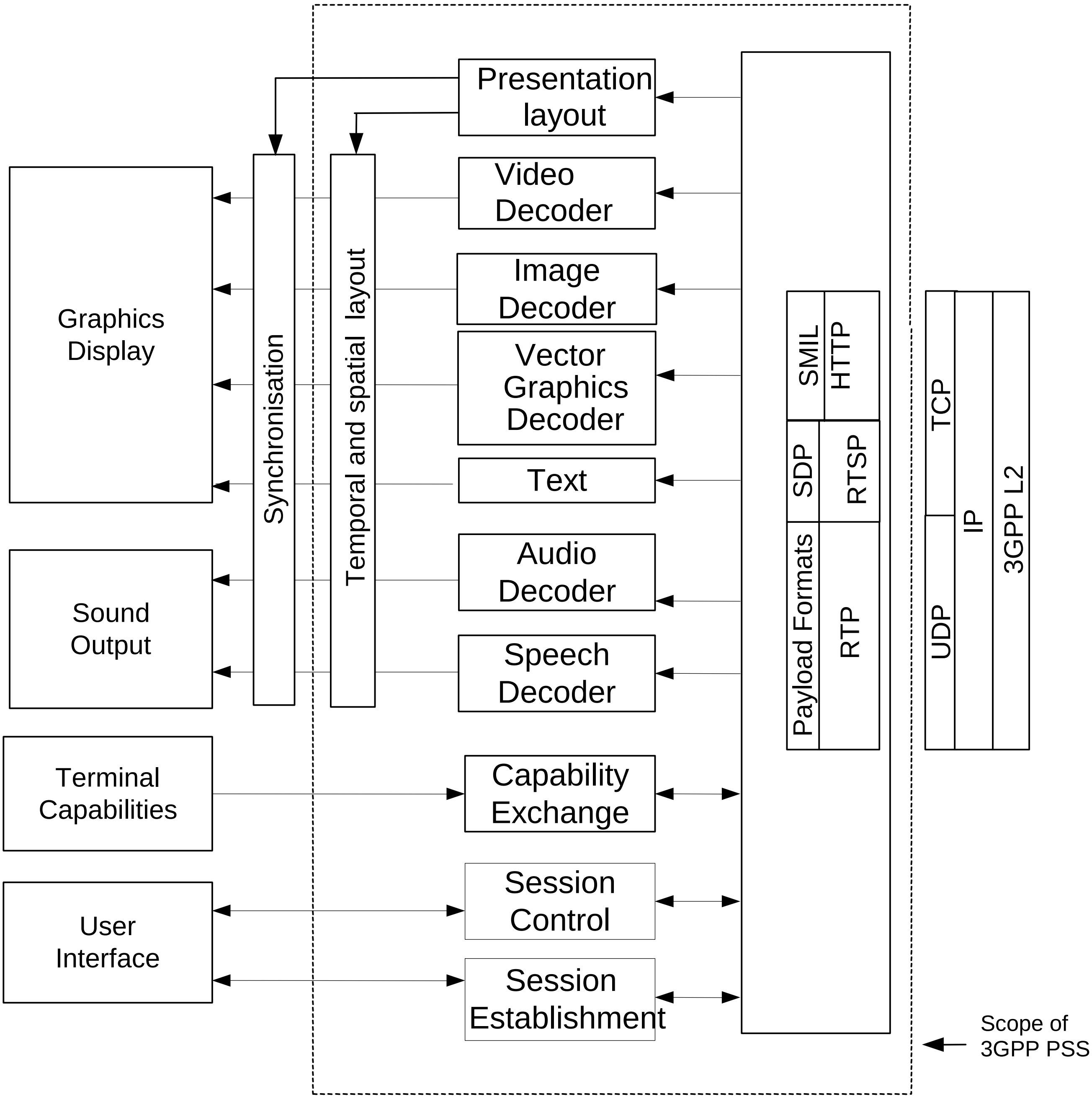


Figure 1: Functional elements of a PSS client

Editor's note: This figure must be redrawn with the appropriate drawing tool (Micrografx Designer or MS Draw 98). The description of the protocol stack should be removed from figure 1 and put into a separate figure. In this figure a box labelled e.g. payload depacketiser should instead describe the transport protocols.

Figure 1 shows the functional elements of a PSS client. The functional elements are either related to control, presentation layout, media codecs or the transport of media and control data.

The control related elements are session establishment, capability exchange and session control (see clause 5).

- Session establishment refers to methods to invoke a PSS session from a browser or directly by entering an URL in the terminal's user interface.

- Capability exchange enables automatic choice or adaptation of media streams depending on different terminal capabilities.
- Session set-up and control deals with the set-up of the individual media streams between a PSS client and one or several PSS servers. It also enables control of the individual media streams by the user. It may involve VCR-like presentation control functions like start, pause, fast forward and stop of a media presentation.

Presentation layout consists of spatial layout, a description of the relation between different media that is included in the media presentation and information for the synchronisation of different media (see clause 8).

The PSS includes media codecs for video, still images, [vector graphics,] bitmap graphics, text, [audio,] and speech (see clause 7).

Transport of media and control data consists of the encapsulation of the coded media and control data in a transport protocol (see clause 6)

5 Protocols

5.1 Session establishment

FFS

5.2 Capability exchange

FFS

5.3 Session set-up and control

5.3.1 RTSP

RTSP [4] shall be used for session set-up and session control. PSS client and servers shall follow the rules for minimal [on-demand playback] RTSP implementations in Appendix D of [4]. In addition to this,

- PSS servers and clients shall implement the DESCRIBE method (see subclause 10.2 in [4]);
- PSS servers and clients shall implement the Range header field.

5.3.2 SDP

SDP shall be used as the format of the presentation description for both PSS clients and servers. PSS servers shall generate and clients interpret the SDP syntax according to the SDP specification [5] and Appendix C of [4]. MIME types to be used in the SDP is described in subclause 5.3.3 of the present document.

PSS servers shall always include the following attributes in the SDP:

- "a=control:" according to clause C.1.1, C.2 and C.3 in [4];
- "a=range:" according to clause C.1.5 in [4].

[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.]

5.3.3 MIME types

The SDP delivered to the PSS client shall declare the media types to be used in the session using a codec specific MIME type for each media. The MIME types for the speech, [audio and] video (see subclause 7.3) codecs shall be according to the corresponding types registered by IANA.

The following MIME types shall be used:

- AMR narrow-band speech codec (see subclause 7.1) MIME type as defined in RFC TBD [9];
- MPEG-4 visual encoded video MIME type as defined in RFC 3016 [10];
- H.263 encoded video MIME type as defined in RFC 2429 [11].

Editor's note: MPEG-4 AAC encoded audio (see subclause 7.2) MIME type as defined in RFC 3016 [10]

6 Data transport

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

6.1 Packet based network interface

PSS clients and servers shall support an IP-based network interface for the transport of session control and media data. Control and media data can be sent either using TCP [7] or UDP [6].

6.2 RTP over UDP/IP

The IETF RTP [8] provides a means for sending real-time or streaming data over UDP (see [6]). The encoded media is encapsulated in the RTP packets with media specific RTP payload formats. RTP payload formats are defined by IETF. RTP also provides a protocol called RTCP for feedback about the transmission quality.

RTP/UDP/IP transport of the following media shall be supported:

- Speech (according to subclause 7.1);
- Audio (according to subclause 7.2);
- Video (according to subclause 7.3).

The following RTP payload formats shall be used:

- RFC TBD [9], for narrow-band AMR encoded speech;
- RFC 3016 [10], for MPEG-4 visual encoded video;
- RFC 2429 [11], for H.263 encoded video.

Editor's note: RFC 3016 [10], for MPEG-4 AAC encoded audio

6.3 HTTP over TCP/IP

The IETF TCP provides reliable transport of data over IP networks, but with no delay guarantees. It is the preferred way for sending the presentation layout and synchronisation description, text, bitmap graphics and still images. There is also need for an application protocol to manage the transfer. The IETF HTTP [12] provides this functionality.

HTTP/TCP/IP transport shall be supported for

- still images (according to sub clause 7.4);
- bitmap graphics (according to sub clause 7.5);
- text (according to sub clause 7.7);
- transport of the presentation layout and synchronisation description (according to clause 8).

Editor’s note: Working assumption. The list may be expanded

6.4 Transport of RTSP

Transport of RTSP shall be supported according to RFC 2326 [4].

7 Codecs

7.1 Speech

The AMR codec shall be supported for narrow-band speech [13]. The AMR wide-band speech codec [14] shall be supported when wide-band speech working at 16 kHz and above is supported.

Editor's note: Payload format for AMR-WB needs to be clarified.

7.2 Audio

Editor’s Note: The working assumption is to have support for MPEG-4 AAC [15]. The RTP payload format used shall be according to [10]. The following areas must be clarified: object types, IPMP, Scalability and transport protocols.

The contribution SA-010018 provides more information about the areas mentioned above. In SA-010018 it is proposed to support the object types MPEG-4 AAC Low Complexity (AAC-LC), Long-Term Prediction (AAC-LTP) and decoding of the AAC base layer of an MPEG-4 AAC Scalable object. The proposal was also that audio decoder may optionally support decoding of multiple AAC layers of an MPEG-4 AAC Scalable object. The maximum sampling rate to be supported by the decoder is proposed to be 48 kHz. The channel configurations to be supported are mono (1/0), dual mono (1+1) and stereo (2/0). Additional concerns raised in the discussion in the SA4 group where:

- Robustness against packet losses when using LTP, since there is a dependency between frames?
- Error robustness in general. Tests for UMTS channels requested. The error robustness depends on whether corrupted packets are delivered to the application or not, i.e. does the application just experience packet losses or are there packets with bit errors delivered to the application. This needs to be clarified (Liaison statement sent to RAN)

7.3 Video

ITU-T H.263 baseline [16] shall be supported. This is the mandatory video codec for the PSS. Additional, PSS clients and servers should support

- H.263 profile 3 level 10, [17];
- MPEG-4 Visual Simple Profile Level 0, [18] and [19].

Editors note: Need to include reference to FLC value.

7.4 Still images

ISO/IEC JPEG [20] together with JFIF [21] shall be supported. [The support for ISO/IEC JPEG only apply to the following two modes:

- baseline DCT, non-differential, Huffman coding, as defined in table B.1, symbol 'SOF0' in [20];
- progressive DCT, non-differential, as defined in table B.1, symbol 'SOF2' [20].]

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

7.5 Bitmap graphics

[The following bitmap graphics codecs should be supported:

- GIF87, [26];
- GIF89a, [27].]

7.6 Vector graphics

FFS

7.7 Text

[Text shall be formatted according to XHTML Basic [22] [23] [24].

The following character encoding shall be supported:

- UTF-8, [23];
- UCS-2, [24].]

8 Presentation layout and synchronisation

8.1 General

The 3GPP PSS use a subset of SMIL 2.0 [25] to describe presentation layout and synchronisation. This subset, or Profile, is defined in this clause through the specification of the SMIL 2.0 modules that a minimal 3GPP PSS client shall support. The present document also includes two informative Annexes that provide additional information on SMIL. Annex B provides guidelines for SMIL content authors and Annex C recommendations for optional SMIL modules to support.

NOTE: The interpretation of this is not that all streaming sessions are required to use SMIL. For some types of sessions, e.g. consisting of one single continuos media or two media synchronised by using RTP timestamps, SMIL may not be needed.

8.2 PSS SMIL profile

PSS clients shall support the following SMIL 2.0 modules:

- BasicInlineTiming
- EventTiming

BasicTimeContainers

- MinMaxTiming
- BasicContentControl

[- CustomTestAttributes]

Editor's Note: It is the working assumption to include the CustomTestAttributes module if any special 3GPP test attributes can be found.

- SkipContentControl
- BasicLayout
- BasicLinking
- BasicMedia
- MediaClipping
- MetaInformation
- Structure

9 Interworking with MMS

Editor's note: The SA4 group received a liaison statement (S4-000689) from TSG T2 requesting guidance from SA4 concerning a file format for MMS. In the response (S4-010078) sent to T2 the 3GPP S4 group recommends that the MP4 format be adopted for use in the Multimedia Messaging Service, according to the use cases described in S4-010078. The SA4 group will integrate this recommendation with a guideline addressing specific technical issues related to this choice (e.g. mandating optional elements or restricted support for elements that are mandatory in MP4). In the response T2 was also asked to inform SA4 if such guidelines should be included in T2:s specifications for MMS or in SA4:s specifications for PSS. Meanwhile any drafting of detailed technical specifications regarding this issue will be included in this document and kept here if T2 so decides. In this case T2 would only add a text in TS 23.140 mandating the format with a reference to this document for technical details. The following is a start for normative text.

[The MP4 file format is mandated in TS 23.140 to be used along the entire delivery chain envisaged by the Multimedia Messaging Service, independent on whether the final delivery is done by streaming or download, thus enhancing interoperability.

In particular, the following stages are considered:

- upload from the originating terminal to the MMS proxy
- file exchange between MMS servers
- transfer of the media content to the receiving terminal, either by file download or by streaming. In the first case the self-contained file is transferred, whereas in the second case the content is extracted from the file and streamed according to open payload formats. In this case, no trace of the file format remains in the content that goes on the wire/in the air.

Additionally, the MP4 file format may be used for the storage in the servers and the "hint track" mechanism may be used for the preparation for streaming. However, these use cases can be regarded as server implementation issues, and mandating them is beyond the scope of this specification.]

Annex A (informative): Protocols

A.1 SDP

This clause gives some background information on SDP.

The SDP 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.

Editor's note: This table needs to be formatted according to 3GPP drafting rules and a table heading added.

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

EXAMPLE 1:

```
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.

EXAMPLE 2:

```
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
```

A.2 RTSP

Editor's note: We should include examples of RTSP signalling.

Annex B (informative): SMIL authoring guidelines

B.1 General

Editor's Note: Type of presentations, number of simultaneous video sequences etc.

B.1 BasicLinking

Editor's Note: Recommendation for the usage of the area element should be described, terminals with different user interfaces etc.

B.2 BasicLayout

Editor's Note: Use of the "fit" attribute to scale video sequences should be restricted.

B.3 MetaInformation

Editor's Note: The inclusion of MetaInformation should be restricted…

B.4 XML entities

Entities are a mechanism to insert XML fragments inside an XML document. Entities can be internal, essentially a macro expansion, or external. XML entities should be avoided in PSS SMIL documents since all SMIL parsers may not support this.

Annex C (informative): Recommended additional SMIL 2.0 modules

C.1 PrefetchControl

C.2 MediaParam

Annex D (informative): Change history

Change history							
Date	**TSG #**	**TSG Doc.**	**CR**	**Rev**	**Subject/Comment**	**Old**	**New**
001019				0.0.1	Changes according to Streaming ad-hoc subgroup at TSG-SA#12	S4-000435	S4-000510
001026				0.0.2	Draft of agreed changes at TSG-SA#13	S4-000510	S4-000552
001027				0.0.3	Revised version after review by SA4	S4-000552	S4-000556
001117				0.0.4	Update before TSG-SA#14	S4-000556	S4-000569
001129				0.1.0	Draft of agreed changes at TSG-SA#14	S4-000569	S4-000654
001130				0.1.1	Changed accordingly to review by PSM SWG	S4-000654	S4-000658
001201				1.0.0	For information to SA#10	S4-000658	S4-000696, 26234-100
010116				1.0.1	Restructuring of layout, formatted accordingly to 3GPP drafting rules, clarified where to find MIME types, references updated.	26234-100	S4-010042
010123				1.1.0	Added Range header in RTSP subclause, Additional information about audio codecs, clarification on SMIL, interworking with MMS, WB speech codec.	S4-010042	S4-010081
010124				1.1.1	Minor changes after review by the PSM group	S4-010081	S4-010101
010125				1.1.2	Minor changes after review by SA#15 plenary	S4-010101	S4-010135
010208				1.2.0	Decisions on SMIL Profile at the PSM streaming Adhoc added	S4-010135	S4-AHP015