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Packet Switched Streaming Service White Paper

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November 19, 2003

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Introduction

Streaming is a method for transferring data with real-time characteristics so that the recipient can start viewing the presentation before the entire contents have been transmitted. Currently, streaming is typically used in broadband Internet audio and video transmissions, but it is becoming feasible also in mobile networks.

The third generation partnership project (3GPP), established in 1998, is a collaboration agreement between several telecommunication standardization bodies. The original scope was to produce globally applicable technical specifications and reports for a third generation mobile system based on evolved GSM networks using the WCDMA radio technology. Currently, the scope also includes the maintenance and further development of the GSM system. There also exists a "sister project", 3GPP2, with similar goals but based on evolving CDMA technology (using CDMA2000 radio technology). More information can be found on the organizations' web pages at [1] and [2].

Transparent end-to-end packet switched streaming service (PSS) is a specification that defines a framework for an interoperable streaming service in 3GPP mobile networks. The framework reuses work done by organizations like IETF (protocols, payload formats), W3C (scene description), MPEG, ISO and ITU (codecs, media file format). PSS first appears in 3GPP Release 4. Similar work is being done in 3GPP2 under the term MSS – multimedia streaming service.

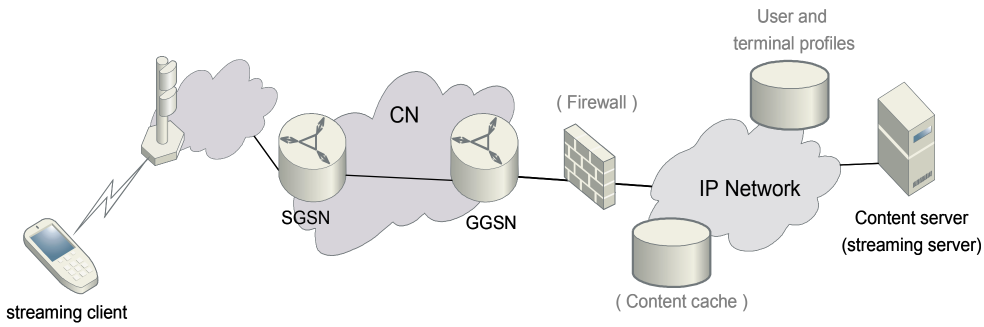

Figure 1. Network elements involved in 3GPP PSS [4]

PSS is an application level service; the specification mostly deals with the streaming client and server [Figure 1]. Although streaming can benefit from network support (e.g. Quality-of-Service, QoS), one requirement for PSS is that it should work over different (QoS) bearers. Thus, the idea lies in defining the service in such a way that it can adapt to the network.

Packet Switched Streaming Service Releases

The basic framework appears in 3GPP Release 4. Release 5 introduces features such as capability exchange; several new features are being incorporated in Release 6. In the following, the different releases are described in more detail.

PSS Rel-4

PSS in 3GPP Release 4 (frozen in March 2001) defines the basic framework: protocols, codecs and the 3GPP file format. The 3GPP protocol stack is illustrated in Figure 2. All the streaming related protocols utilize TCP and/or UDP as their transport. A summary of the protocols is listed in Table 1.

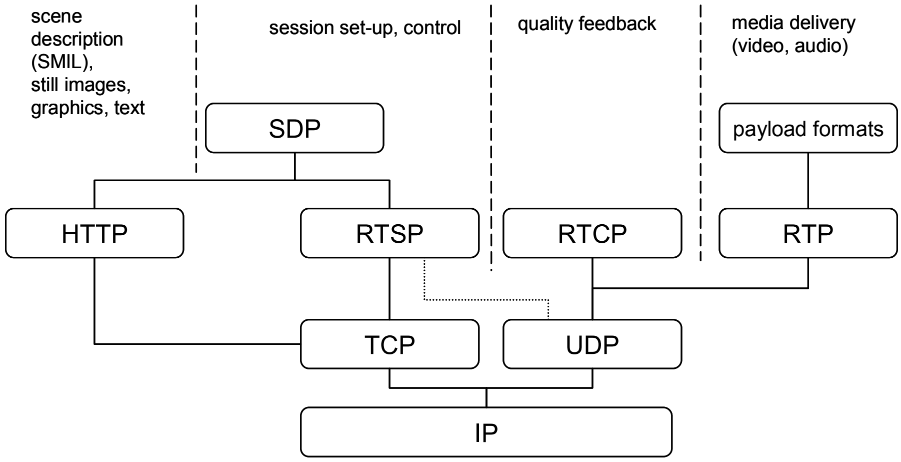

Figure 2. 3GPP PSS protocol stack [5].

Table 1. Summary of PSS streaming related protocols.

RTP	Real-time Transport Protocol [RFC 1889, RFC 1890]	RTP provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast network services. RTP does not address resource reservation and does not guarantee quality-of- service for real-time services [6]. It should be noted that in PSS, RTP is carried over UDP only.
RTCP	Real-Time Control Protocol [RFC 1889]	The primary function of RTCP is to provide feedback on the quality of data distribution. This is achieved by periodic "receiver report" packets sent by the receiver to the sender (reports contain e.g. inter-arrival jitter measured by the receiver and number of packets lost).
RTSP	Real-Time Streaming Protocol [RFC 2326]	RTSP is used to establish and control time-synchronized streams of continuous media. It acts as a "network remote control" for multimedia services. The protocol itself is textual and resembles HTTP, the main differences being that RTSP is stateful and the media data is (usually) delivered out-of-band using a separate transport protocol (normally RTP).[7]
SDP	Session Description Protocol [RFC 2327, RFC 2326]	The purpose of SDP is to convey information about media streams in multimedia sessions to allow the recipients of a session description to participate in the session [8]. It is a text based protocol with relatively simple and extendable syntax.

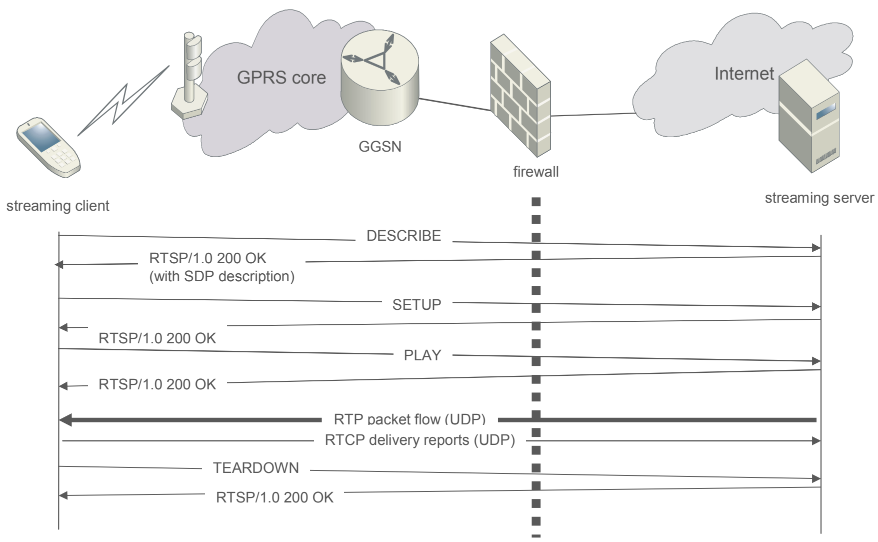

Figure 3. Basic RTSP/RTP operation.

A basic RTSP unicast operation is shown in Figure 3. The client learns the location of a media clip for example by browsing to a web page that has an RTSP URL. The streaming player connects to the streaming server and issues a RTSP DESCRIBE command. The server responds with an SDP description which includes information like number of streams, media types, and required bandwidth. After parsing the description, the client issues an RTSP SETUP command for each stream in the session. The SETUP command tells the server which ports the client uses to receive the media. When streams have been set up, the client issues a PLAY command after which the server starts sending the media streams as RTP packets over UDP to the client. Finally, the client issues a TEARDOWN command to end the streaming session.

The 3GPP PSS defined audio and video codecs are summarized in Table 2. The maximum bitrate is restricted by the codec specification, not by PSS. It should be noted that PSS also defines formats for some non-realtime media such as still images (JPEG), bitmap graphics (GIF), vector graphics (SVG-Tiny, Rel-5 onwards), text, and scene description (SMIL). In should also be noted that there is no mandatory audio codec (e.g. for music) in PSS Rel-4.

Table 2. 3GPP PSS audio and video codecs [5].

Type	Codec (Decoder)	Support	Max. bitrate	Notes
Speech	AMR-NB	Required	12.2 kbps	
Speech	AMR-WB	Required[1]	23.85 kbps	
Audio	MPEG-4 AAC-LC	Recommended	N/A	
Audio	MPEG-4 AAC-LTP	Optional	N/A	
Video	H.263 profile 0 level 10	Required	64 kbps	Max. frame size 176x144
Video	H.263 profile 3 level 10	Recommended	64 kbps	Interactive and wireless streaming profile --"Provides enhanced coding efficiency performance and enhanced error resilience for delivery to wireless devices." [H263]
Video	MPEG-4 Simple Visual Profile Level 0	Recommended	64 kbps	Max. frame size 176x144

[1] When wideband speech is supported by mobile terminals

The 3GPP file format for timed multimedia is based on the ISO base media file format (ISO standard 14496-12) which is also the basis of the MP4 file format and originally derived from the QuickTime file format. Rel-4 PSS actually references the MP4 file format, but that has been changed in Rel-5. The file format is flexible and supports both local playback and streaming delivery. The same file format is also used for MMSs, but there are some restrictions and additions to the ISO specification. For example, the 3GPP media file must be self contained; no references to external media are allowed. Also, the specification defines how the 3GPP specific media types (H.263 video and AMR audio) are represented in the file.

PSS Rel-5

The most notable change in PSS Rel-5 (frozen 03/2002) is the addition of capability exchange. The functionality is defined as an extension to User-Agent Profile (UAProf). UAProf is specified by Open Mobile Alliance and deals with capturing classes of device capabilities and preference information for content formatting [9].

Basically, during streaming session initiation, the client provides a capability profile to the server (an URL referring to the profile and possible differences). With the PSS capability profile, the client can signal information such as the number of audio channels, supported media types, rendering screen size, and bits per pixel. The server can use this information to, for example, select content that is best suited for the client.

Release 5 uses the same audio and video formats as Rel-4, but adds some new media types: vector graphics (SVG Tiny), synthetic audio (scalable polyphonic MIDI) and timed text (for e.g. subtitles). Also, scene description (SMIL) support has been extended.

PSS Rel-6

The PSS Release 6 specification is still a work-in-progress, the expected freeze date is in H1/04. Some notable new features are presented in the following. The information below is based on the draft 3GPP documents ([10], [11]) available at the time of the writing and is therefore subject to change.

End-to-end bit-rate adaptation

End-to-end bit-rate adaptation enables the streaming session to adapt to varying network conditions. This is important as PSS could be potentially used in networks with very different capabilities (WCDMA, EDGE, GPRS, QoS with guaranteed bit-rate or best-effort). In addition, the smooth operation of intra and especially inter-system handovers can potentially benefit from bit-rate adaptation. In the currently suggested mechanism, the streaming server is mainly responsible for adapting the sampling (stream bit-rate) and transmission rate and the client is responsible for providing the necessary feedback to the server. The goal is to keep the client pre-decoder buffer sufficiently full so that no gaps occur in the audio or video playback.

Ideally, bit-rate adaptation can ensure smooth and uninterrupted stream reception in most network conditions. However, it also means that the audio and/or video quality of the stream can change during reception. It also (usually) requires that the same content is available in several bit-rates, or that the server is able to "thin" the stream (for example, by only transmitting the key frames). The suggested implementation also makes the streaming server more complex. Currently, in fixed line Internet access, mechanisms where the client controls what bit-rate the server should send are more common (e.g. RealNetworks' SureStream).

Quality metrics

The purpose is to enable the PSS servers to receive client generated quality metrics, which could be used to determine the (subjective) quality of the client experience and monitor the service for improvements, for instance. The proposed metrics include information such as the number of corruptions, lost packets, gaps in reception etc. Quality metrics are not intended to be used for billing purposes.

Reliable streaming

Reliable streaming means streaming with reliable transport so that all the media is delivered to the receiver. It can be seen as an intermediate form between streaming and download:

- Delivers all the media without losses (download).
- Client can start viewing the presentation before the entire contents have been transmitted (streaming).

However, this is achieved by compromising (near) real-time, uninterrupted playback and favoring lossless reception even if it causes more interruptions. Therefore, reliable streaming is not particularly suitable for live streams.

Several mechanisms for reliable streaming have been proposed: Progressive download (over HTTP), RTSP tunneling (RTP interleaved with RTSP over TCP) and a resend mechanism. Currently, the relatively simple progressive download has been selected.

Digital Rights Management (DRM)

PSS Rel-6 should support DRM as specified in 3GPP TS 22.242: "Digital Rights Management (DRM)" [1]. It is assumed that the PSS Rel-6 DRM is in practice OMA DRM release 2.0, which is not publicly available at the time of this writing.

New codecs

The H.264 (MPEG-4 AVC) video codec is being considered for PSS Rel-6. Lately, Microsoft also proposed the Windows Media 9 video codec for this purpose. A mandatory audio codec is also being considered. The current audio codec contenders are aacPlus (MPEG-4 HE-AAC) and extended AMR-WB in the lower bit-rate (12-32kbps) range and aacPlus and MPEG-4 AAC in the higher bit-rate range (>32kbps).

Summary

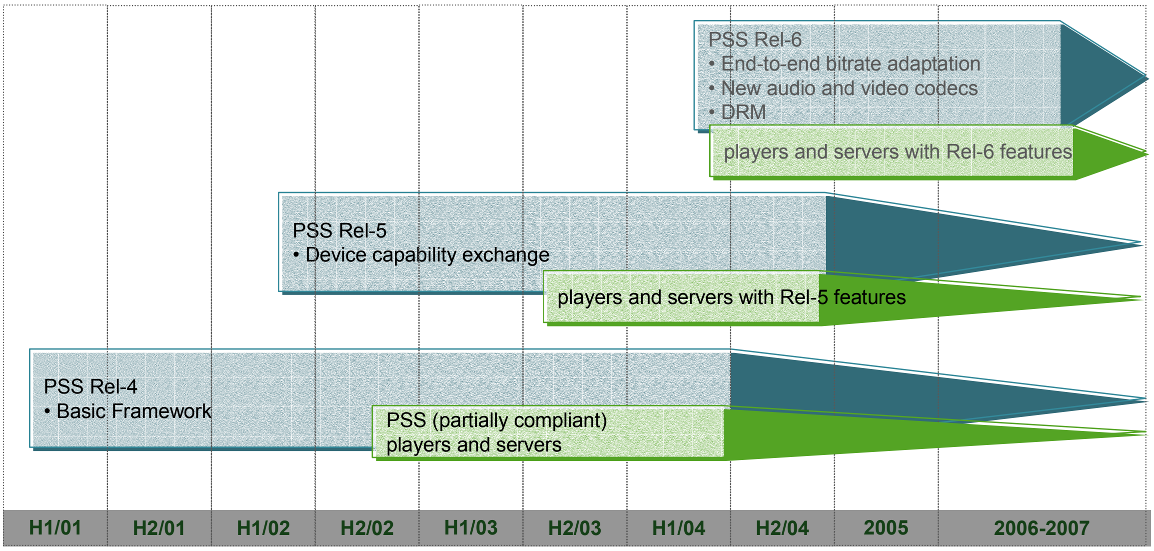

Figure 4. PSS Timeline (From the freeze date of each release. The REL-6 freeze date is an estimate).

The PSS specification timeline is presented in Figure 4. It should be noted that proprietary solutions for mobile streaming from RealNetworks and PacketVideo today offer some of the features (e.g. bitrate adaptation, resending of missing packets) that are being incorporated in release 6. Nevertheless, PSS is important as it defines a framework which enables an interoperable service and it is assumed that players and servers with release six features will emerge soon after the release has been frozen as it incorporates features that are useful in today's networks.

Charging for PSS services is a challenging issue and the exact mechanism is likely to be implementation dependent. In Release 5, service-user level (stage 1) requirements are introduced and these include charging requirements which state that "PSS should support various charging mechanisms, for example, time base charging, volume based charging, event based charging, content based charging" [12]. However, there is no technical implementation of the requirements: "Interworking with charging/billing services can be part of a future release of PSS" [13]. Status of charging in Rel-6 is still unknown.

References

[1] The 3rd Generation Partnership Project (3GPP). Available from http://www.3gpp.org/ [Accessed Aug 20,2003].

[2] The 3rd Generation Partnership Project 2 (3GPP2). Available from http://www.3gpp2.org/. [Accessed Aug 20,2003]

[3] 3GPP TS 22.242 Digital Rights Management (DRM). Available from: http://www.3gpp.org/ftp/Specs/latest/ [Accessed Jul 27, 2003]

[4] 3GPP TS 26.233 Transparent end-to-end packet switched streaming service (PSS); General description. Available from: http://www.3gpp.org/ftp/Specs/latest/ [Accessed Jul 27, 2003]

[5] 3GPP TS 26.234 Transparent end-to-end packet switched streaming service (PSS); protocols and codecs. Available from: http://www.3gpp.org/ftp/Specs/latest/ [Accessed Jul 27, 2003]

[6] RFC 1889. Real-time Transport Protocol. Available from: http://www.ietf.org/rfc/rfc1889.txt [Accessed Aug 19, 2003]

[7] RFC 2326 Real-Time Streaming Protocol. Available from: http://www.ietf.org/rfc/rfc2326.txt [Accessed Aug 19, 2003]

[8] RFC 2327 Session Description Protocol. Available from: http://www.ietf.org/rfc/rfc2327.txt [Accessed Aug 19, 2003]

[9] OMA User Agent Profile Version 2.0. Available from http://www.openmobilealliance.org/agreement/LicenseAgreement.asp?DocName=OMA-UAProf-v2_0-20030520-C1.zip [Accessed Sep 24, 2003]

[10] 3GPP TS 26.234 V0.2.6. Transparent end-to-end packet switched streaming service (PSS); protocols and codecs (Release 6) (working draft). Available from: http://www.3gpp.org/ftp/tsg_sa/WG4_CODEC/TSGS4_27/Docs/S4-030567.zip [Accessed Aug 20, 2003]

[11] Tdoc S4-030562. Draft Rel-6 PSS Quality Metrics Permanent Document version 0.04. Available from: http://www.3gpp.org/ftp/tsg_sa/WG4_CODEC/TSGS4_27/Docs/S4-030562.zip [Accessed Aug 20, 2003].

[12] 3GPP TS 22.233. Transparent end-to-end packet-switched streaming service; Stage 1. Available from http://www.3gpp.org/ftp/Specs/latest/ [Accessed Aug 20, 2003]

[13] 3GPP TS 26.233 V5.0.0. End-to-end transparent streaming service; General description (Release 5) . Available from http://www.3gpp.org/ftp/Specs/archive/26_series/26.233/26233-500.zip [Accessed Aug 20, 2003]

Additional Resources

- IETF transport area, especially audio/video transport (avt) group which has defined the RTSP, RTP, RTCP and SDP protocols and many payload formats for RTP. Also, dccp (Datagram Congestion Control Protocol) is interesting as it is potentially better suited for streaming transport than UDP (congestion control, better firewall traversal). See http://www.ietf.org/html.charters/wg-dir.html#Transport%20Area.
- W3C work in scene description (SMIL, XHTML), scalable vector graphics (SVG) and capability exchange (CC/PP, RDF). See http://www.w3.org/.
- ISO/IEC MPEG working group defines the MPEG standards. Currently the most relevant work for PSS are the MPEG-4 audio and visual as well as the ISO base media file formats. See http://www.iso.ch/ and http://mpeg.telecomitalialab.com/.
- ITU-T Study Group 16 (e.g. H.263 codec) and especially the Joint Video Team (JVT) which in co-operation with MPEG is responsible for the new H.264/MPEG-4 AVC video coding standard. H.264 is a potential new video codec in Release 6 time range. See http://www.itu.int/.
- Open Mobile Alliance (OMA) especially in the capability exchange (UAProf) and DRM area. See http://www.openmobilialliance.org/.
- Internet Streaming Media Alliance (ISMA) which may have significance in the DRM area and in (MPEG-4) streaming interoperability in general. See http://www.isma.tv/.

Definitions, acronyms and abbreviations

3GPP	Third Generation Partnership Project
CDMA	Code Division Multiple Access
DRM	Digital Rights Management
EDGE	Enhanced Data rates for GSM Evolution
GGSN	Gateway GPRS Support Node (a GPRS network element).
GPRS	General Packet Radio Service
IETF	Internet Engineering Task Force
ISMA	Internet Streaming Media Alliance
ISO	International Organization for Standardization
MMS	Multimedia Messaging Service
MPEG	Moving Picture Experts Group
PSS	Packet switched Streaming Service
QoS	Quality-of-Service
RTCP	Real-Time Control Protocol
RTP	Real-Time Transport Protocol
RTSP	Real-Time Streaming Protocol
SDP	Session Description Protocol
SGSN	Serving GPRS Support Node (a GPRS network element).
SMIL	Synchronized Multimedia Integration Language
SVG	Scalable Vector Graphics
WCDMA	Wideband Code Division Multiple Access

Appendix A. 3GPP PSS Specifications

The current PSS technical specifications and reports are listed in the table below. The PSS work is (mostly) conducted in the 3GPP Services and System Aspects TSG Codec working group (SA WG4). It should be noted that at the time of writing, the Rel-6 specifications are a work in progress as Release 6 has not yet been frozen.

Number	Title	WG	R97	R98	R99	Rel-4	Rel-5	Rel-6	active
TS 22.233	Transparent end-to-end packet-switched streaming service; Stage 1	S1					Y	Y	yes
TS 26.233	End-to-end transparent streaming service; General description	S4				Y	Y		yes
TS 26.234	Transparent end-to-end transparent streaming service; Protocols and codecs	S4				Y	Y	Y	yes
TS 26.244	Transparent end-to-end transparent streaming service; 3GPP file format (3GP)	S4						Y	yes
TS 26.245	Transparent end-to-end transparent streaming service; Timed text format	S4						Y	yes
TS 26.246	Transparent end-to-end transparent Packet-switched Streaming Service (PSS); 3GPP SMIL language profile	S4						Y	yes
TR 26.937	Transparent end-to-end packet switched streaming service (PSS); RTP usage model	S4					Y		yes

Links between PSS and other 3GPP work:

- Multimedia Messaging Service (MMS) has same codecs and file format for continuous media
- IP Multimedia Subsystem (IMS) can use PSS for streaming services
- Use of streaming (PSS) in Multimedia Broadcast/Multicast Service (MBMS)
- Digital Rights Management (DRM): PSS Rel-6 should also include DRM

The latest specifications are available on the 3GPP project web site: http://www.3gpp.org/ftp/Specs/latest/

