An End-to-End Quality of Service Management Framework for Streaming Services in Third Generation Mobile Networks

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***Abstract*- In this paper, an end-to-end Quality of Service (QoS) framework for streaming services in 3G mobile networks is considered. A solution based on a Public Land Mobile Network (PLMN) hosted multimedia streaming service is studied to avoid accessing through an external IP packet data network (IP-PDN) to streaming services. Under this scenario, the interaction between UMTS and IETF's protocols and mechanisms for a streaming session is analyzed. By signaling flowcharts, it is shown that both groups of protocols and mechanisms can co-operate to provide seamless end-to-end real-time services. Supported by this proposed framework, provisioning of audio streaming services over 3G mobile networks is also addressed.**

***Index terms*- multimedia streaming, quality of service, 3G mobile networks**

I. Introduction

Multimedia streaming services are receiving considerable interest in the mobile network business [1]. Supporting reliable real time services is decisive aspect for the increasing migration towards packet based telephony networks. For UMTS, deploying an all IP architecture is a promising standardization trend due to convergence between IP technologies and telephony services [2]. This service is also technically applicable over evolving second and third generation wireless networks, thus streaming clients will soon be deployed in advanced wireless communication devices.

Although a few proprietary streaming technologies rule the Internet today, proliferation of IETF standardized protocols, such as Real Time Streaming Protocol (RTSP), and aims to standardize an open streaming concept in major wireless standardization organizations (3GPP, 3GPP2) will bring a strong open standard based service to the wireless marketplace [3]. However, one important advantage of supporting an existing commercial service platform (such as a RealNetworks™ or QuickTime™ server) is to provide added value from the access to existing service/content provider, besides its brand awareness.

One key issue is how mobile networks can support these kind of services. In these "Pre-All-IP" service cases the used radio bearers can be chosen from either 2G or 3G circuit switched (CS) or packet switched (PS) bearer set. First commercial streaming services may well utilize existing CS bearer services but in 3G the services will be offered over PS bearers. Recent studies show that GPRS/EGPRS data bearers exhibit good performance when supporting streaming services [4]. Consequently, this may shorten the period needed for using CS bearers for providing streaming-like services on top of reliable transport channels. The PS wireless services will still utilize relatively low transmission bandwidths due to overall capacity restraints in the air link capacity. Thus, they should benefit from standardized and robust IP header compression methods while achieving an acceptable QoS for end users.

Providing end-to-end QoS for multimedia streaming services implies the harmonized interworking between protocols and mechanisms specified by IETF and 3GPP[5], both involved in QoS provisioning within the different 3G network subdomains and the external IP-PDN which the service is accessed through. In this paper, the end-to-end QoS management of streaming services in 3G mobile networks is considered. Particulary, the possibility of employing a PLMN-hosted multimedia streaming service is studied to avoid accessing through an external IP-PDN to streaming services. By this solution the mobile operator host a streaming server or a proxy server within the PLMN, allowing to provide sufficient QoS to users of wireless streaming terminals. More specifically, in this work provisioning of audio streaming services over 3G mobile networks is tackled. In addition, the presented analysis of the multimedia streaming session is chronologically divided in two phases: service activation and service utilization.

The remainder of this paper is organized as follows. In the first place, multimedia streaming service, mobile network architecture and protocol stack are overall described in section II. Secondly, service activation is detailed depicted, highlighting both application and UMTS level signaling procedures, in section III. Afterwards the mechanisms involved in QoS provisioning in the UMTS network while the service is ongoing are outlined in section IV. Finally, some conclusions in section V summarize the main ideas presented in this work.

II. Overall Scenario Description

A. Description of the service: Multimedia Streaming

A generic framework for a typical multimedia streaming service consists of content creation and retrieval system. The content creation system has one or more media sources, e.g. a camera and a microphone. In order to compose a multimedia clip consisting of different media types, the raw data captured from the sources are edited. Typically, the storage space required for raw media data is quite large. In order to facilitate attractive multimedia retrieval service over commonly available transport channels such as low bit rate modem connections, the media clips are also compressed in the editing phase, before they are handed to a server. Typically, several clients can access the server over a determined network. The server is able to respond to the requests presented by the clients and its main task is to transmit a desired multimedia clip to the client. Then, the client decompresses and plays the clip.

In the past, to view media on the Internet, users had to download the entire clip or file to their local hard disk drive before playing it. During the past couple of years streaming has matured and gained high user acceptance especially within users of Internet-enabled PCs. This technology is also seen as one of the new added value Internet services, and competition on the area has been and continues to be heavy.

By streaming, a media server opens a connection to the client terminal and begins to stream the media to the client at approximately the playout rate. During the media receiving, the client plays the media with a small delay or no delay at all. This technique does not only free up precious terminal memory, but also it allows for media to be sent live to clients as the media event happens.

The user needs a player, which is a special program that decompresses and sends video data to the display and audio data to the speakers. This client application must be able to control the streaming flows (control plane) and manage the media flows (user plane). In addition the client also has to interface with the underlying transport network technology, its specific protocols and data bearers dedicated to the service.

Nowadays, since the access to Internet services is moving fast to wireless devices, the available computing capacity in devices increases, and user rates for cellular subscribers are approaching those of wired terminals, streaming service is technically feasible also in wireless handsets.

Generally, multimedia streaming by definition is seen to include one or several media streamed or transported to the client over the network. Some example services are:

- Audio streaming (offering e.g. music playback at the terminal), that is the one studied in this paper.
- Streaming with an audio and a video component (news reviews, music videos)
- Audio streaming with simultaneous visual presentation comprising of still images and/or graphical animations, video clips presented at pre-defined order.

Statistical model of this kind of traffic depends quite a lot on the service [6]. In case of audio, the generated traffic is rather non-bursty whereas video traffic has a more bursty nature. This may make CS bearers suitable for audio streaming services whereas PS bearers gives more trunking gain and a better utilization of the resources.

B. *Architecture of the Mobile Network*

A general overview of the considered UMTS network architecture is depicted in fig. 1. Detailed descriptions of the entities, interfaces and protocols in UMTS are given in [7] and [8].

In addition to the User Equipment (UE), the main entities in fig. 3 that are involved in QoS management are: UMTS Terrestrial Radio Access Network (UTRAN) and GSM/EDGE Radio Access Network (GERAN), Serving GPRS Support Node (SGSN), Home Location Register (HLR), Gateway GPRS Support Node (GGSN) and Application Server and RTSP Proxy.

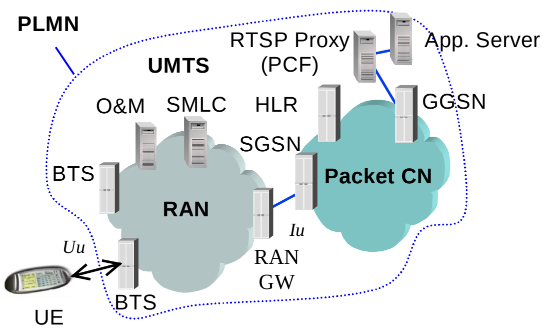

Fig. 1. End-to-end network architecture

End-to-end QoS in the UMTS Release 4 is based on the IP bearer service (IP BS) concept, which consists of the necessary extension of the UMTS BS defined in the UMTS release 1999 [5] to take into account the QoS in the external IP-PDN. In our model, the GGSN is connected to a RTSP proxy, which is also connected to the Streaming Server. Therefore, no external IP-PDN is involved in providing the streaming service.

The IP policy model allows to create a complete framework for management of IP BSs. Policies represent established Service Level Agreements (SLAs) between service providers and users. SLAs specify a set of agreed rules for performing admission control that are not only based on the availability of the requested resources (i.e. QoS, accessibility, security and other network performance issues expected by the UE can be considered). The entity in charge of the IP BS policy management is the Policy Control Function (PCF), which is co-located with the RTSP proxy. GGSN and RTSP proxy use Common Open Policy Service (COPS) protocol to interact and negotiate the IP BS [9].

C. *Protocol Stack: Control plane & User plane*

The 3GPP PS multimedia streaming service is being standardized based on control and transport IETF protocols as RTSP, Real-Time Transport Protocol (RTP) and Session Description Protocol (SDP), as fig. 2 shows. Codec standardization has not been evolving far, but as 3GPP in its history has not standardized audio codecs, this field is fairly open in 3GPP.

RTSP is an application level client-server protocol which is used to control the delivery of real-time streaming data [10]. It establishes and controls one or several streams of continuous media but does not convey the media streams itself. The media streams may be conveyed over RTP, but the operation of RTSP is independent of the transport mechanism of the media streams.

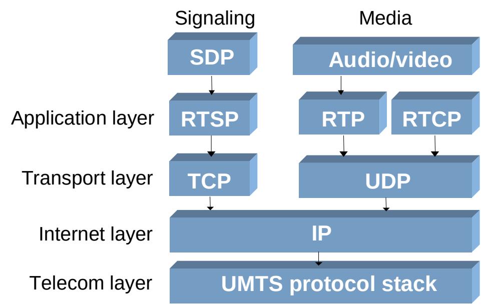

Fig. 2. Protocol Stack for signaling and media flows of streaming services

A presentation description defines the set of media streams controlled by RTSP. The format of the presentation description is not defined in [10], but one example of a presentation description is the session description format SDP, which is specified in [11]. The SDP includes information of the media encoding and port numbers used for the media streams. RTSP specification allows separate media streams to reside in different servers. RTSP may be sent over TCP while the media streams normally use UDP as transport mechanism. Thus, the continuity of the media stream is not affected by gaps in RTSP signaling. The RSTP request is a signaling message from the client to the server. The server sends responses back to the client by RTSP response status codes.

RTP transports media data flows over UDP, in the same way as its related control protocol called Real-time Transport Control Protocol (RTCP) [12]. RTP carries data that has real time requirements while RTCP conveys information of the participants and monitors the quality of the RTP session. The RTP and RTCP services together provide payload type identification, sequence numbering, timestamping and delivery monitoring. RTP defines a flexible framework for real-time data transport for multimedia services but it does not ensure timely delivery or provide any QoS guarantees. Furthermore it does not prevent out-of-order delivery or assume that the underlying network would deliver packets reliably and in sequence.

III. SERVICE ACTIVATION

This phase is described at three levels. In the first place, the service activation procedure from UE viewpoint is briefly outlined. Secondly, the signaling interchanges between application entities by using RTSP in order to establish the session is presented, as well as the media codec negotiation. Finally, all the signaling messages and mechanisms at lower layers (i.e. UMTS protocols) are detailed explained.

A. *User Equipment Operation*

The service activation from user viewpoint can be described as follows. At first, user initiates the streaming client application, which connects to the UMTS network by using a socket Application Program Interface (API). The application requests a primary Packet Data Protocol (PDP) context which is opened to an specific access point with interactive UMTS traffic class and other suitable UMTS QoS Rel'99 parameters. A socket is opened for RTSP negotiation and it is tied to the interactive PDP context. The user then selects an audio streaming content. The application activates a streaming handler to take care of the streaming content. When the RTSP negotiation reaches the SETUP phase, two secondary PDP contexts are activated, one with QoS parameters suitable for audio streaming (RTP traffic) and another for transport signaling (RTCP traffic). The new contexts must be activated before the RTSP PLAY command, because after that the RTP flow will start running through the streaming PDP context. New sockets are opened for RTP and RTCP traffic and they are tied to the corresponding PDP context.

The streaming handler launches a user interface to let the user control the audio stream, including e.g. play, pause and stop knobs.

B. *Application Layer Signaling*

The application layer signaling interchange between the UE and the streaming server is outlined in fig. 3. More detailed description is given in the following steps:

Step 1 – A primary PDP context is activated for the RTSP signaling between the terminal and the streaming server. By means the Access Point Name (APN), the UE finds out the address of the streaming server when the user selects a link that points to streaming content residing in the streaming server.

Step 2 – After creating a TCP connection to the streaming server, the UE sends an RTSP DESCRIBE request to the server. This request indicates that the server should send the UE information about the media it is going to send. This information includes the encoding of the media and the corresponding UDP port number.

Step 3 – The streaming server sends a 200 OK response containing a presentation description in the form of an SDP message. The SDP describes the streaming media the UE is about to receive. It should be noted that the RTSP specification as defined in the IETF [10] does not mandate the use of the DESCRIBE method for this media initialization phase.

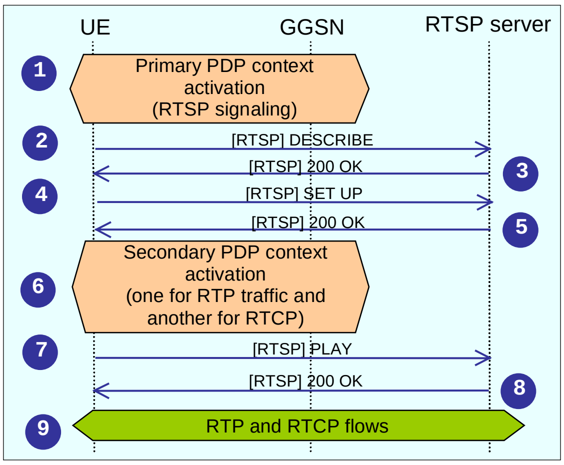

Fig 3. RTSP session initiation procedure in UMTS network

However, in order to function properly any RTSP-based system must receive the description of the media one way or other. The 3GPP standard [3], which defines the protocols and codecs for the transparent end-to-end packet switched streaming service in 3G networks, mandates the use of the DESCRIBE method for the conveyance of the media description.

Step 4 – The UE sends a SETUP request to the server. This message indicates the transport information of the stream including the UDP port numbers the UE is going to use for the RTP stream and the RTCP control traffic.

Step 5 – The server acknowledges the SETUP request by sending a 200 OK response back to the UE.

Step 6 – In this phase the two secondary PDP contexts for the streaming media (RTP and RTCP flows) are activated. This phase is presented in more detail later.

Step 7 – When the resources for the media are successfully reserved the UE sends the streaming server a PLAY request in order to start to receive the stream.

Step 8 – The server replies with a 200 OK response.

Step 9 – The server starts to send the stream in form of an RTP flow. Likewise, RTCP traffic is sent for the QoS control of the corresponding RTP data flow.

C. *UMTS Signaling Procedures*

Once the application level signaling procedure is presented, further insight about the UMTS signaling is provided. In UMTS, all signaling associated with service session establishment is carried out by the control plane through different QoS management functions (i.e. bearer service management, subscription, translation and admission&capability). Fig. 4 illustrates the different steps that form the service activation within the UMTS network.

In the first place, a primary PDP Context is activated, as aforementioned, for RTSP signaling using interactive UMTS traffic class [5]. The interactive traffic class has a priority based handling instead of guarantees based handling, being the reliability requirement the target in this case. The control plane functions are distributed in different layers of several network entities. Assuming that the service session establishment is successful, a detailed description of this phase is shown in the flowchart of fig. 4.

Step 1 – The QoS requirements of the application in the UE are mapped on 3G QoS attributes. Since the primary PDP context is used for RTSP signaling, a 3G QoS profile with interactive traffic class, high priority and low error rate is appropriate. A Session Management (SM) protocol message from the UE to the SGSN initiates the PDP context activation procedure.

Step 2 – After the SGSN has validated the service for that user by quering HLR, local admission control is performed (e.g. based on the state of the buffers, the CPU load, etc...). Then, the SGSN maps the 3G QoS attributes on Radio Access Bearer (RAB) QoS attributes and triggers a RAB assignment procedure in the RAN by using the Radio Access Network Application Protocol (RANAP).

Step 3 – In the RAN, admission control is basically based on the availability of radio resources. Once a new PDP context is accepted, RAB attributes are mapped on Radio Bearer (RB) parameters used in the physical and link layers (e.g. spreading codes, retransmission requirements, etc...). RB according to these parameters is established and it is reported to the SGSN, which employs GPRS Tunneling Protocol for Control Plane (GTP-c) to indicate the GGSN that a new PDP context has to be created.

Step 4 – As the primary PDP context is not intended for real time traffic, no resource reservations are needed in the Core Network (CN). The GGSN accepts to create the primary PDP context based on similar admission criteria to those employed by the SGSN. Thereafter, the GGSN notifies the SGSN that the primary PDP context for RTSP has been successfully created and the SGSN sends an SM message to the application in the UE.

Step 5 – Once Streaming Server accepts the RTSP connection request, the UE triggers two secondary PDP context activation procedures, one for unidirectional RTP traffic and one for bidirectional RTCP traffic. The reason for the use of different secondary PDP contexts is that RTCP traffic must separated from RTP if header compression is going to be applied for RTP/UDP/IP.

Step 6 – The UE converts user data application requirements into QoS profile for streaming class. Thus, table I shows an example of QoS profile for RTP data traffic. The QoS parameters requested for the PDP context take into account the full RTP/UDP/IP headers. Thus, no header compression is assumed in the IP level when requesting QoS. Some assumptions have been made for this proposed QoS profile. A bitrate of 64 kbps is assumed (e.g. MPEG-AAC codec), which achieves good stereo quality. The payload size from the streaming application in this example is assumed to be between 500-1000 bytes. The downlink bitrate of 72 kbps is calculated by including the impact of the following header sizes: RTP 12 bytes, UDP 8 bytes, and IPv6 40 bytes. As RTP flow is unidirectional, guaranteed bitrate for uplink is set to 0 kbps. The Transfer Delay requirement is not so stringent that re-transmissions are possible.

Table I. Proposed QoS Profile for RTP traffic

QoS99 Parameter name	Parameter value
Traffic Class	Streaming
Maximum bitrate for uplink	0 kbps
Maximum bitrate for downlink	90 kbps
Maximum SDU size	1060 bytes
Delivery of erroneous SDUs	No
SDU error ratio	10^{-2}
Transfer Delay	2 s
Guaranteed bitrate for uplink	0 kbps
Guaranteed bitrate for downlink	72 kbps

For a given *SDU Error Ratio*, the larger the SDU size, the smaller the BLER, what means that the reliability requirements for radio link are stringent. Since a more protective conding scheme must be used, the bitrate is lower (for the same radio blocks sent), implying larger delay. Therefore, *maximum SDU size* should be commonly considered with the required *SDU error ratio*. From network viewpoint, smaller SDUs allow easier compliance to reliability requirements by relaxing the radio link adaptation. Moreover, a trade off between the reliability and delay relevancy should be found. This compromise needs to be communicated from UE application to the network or the application criteria for SDU size should be always conservative. In a similar way, the UE converts transport control requirements into QoS profile for RTCP traffic.

Once the QoS profiles are derived, the secondary PDP contexts are activated. This procedure, that is performed for both secondary PDP contexts, is outlined in fig 4. The main steps of a secondary PDP context activation procedure are commented below. The RTP traffic PDP context is used for this example. A new Radio Resource Control (RRC) connection is not needed because every new connection the UE wants to activate can use the RRC connection established for the primary PDP context.

An *Activate Secondary PDP Context Request* SM message is sent from UE to SGSN. This message contains the requested QoS and Traffic Flow Template (TFT) parameters, which is transparently sent through SGSN to GGSN to enable packet classification for downlink data transfer.

Step 7 – The SGSN validates the service request by checking the subscription information stored in the HLR. Besides, the SGSN performs admission control. There are two configurable parameters that control the maximum amount of streaming traffic: real time bandwidth and streaming bandwidth. So here it is checked that there is enough bandwidth for the new PDP context. In affirmative case, the flow is accepted and the resource reservation is performed by decreasing the bandwidth quota by the guaranteed bitrate of the new PDP context. In addition to configuration parameter, the admission control procedure checks the CPU load so that there is processing capacity for an additional flow before accepting the PDP context activation. Allocation/Retention priority parameter is given by the HLR to SGSN. This parameter is used for admission precedence, i.e. to select which one is accepted among several users when streaming bandwidth is limited.

The 3G QoS profile is mapped to RAB QoS attributes. The 3G attributes are exactly the same as the RAB QoS attributes but the values are not typically the same for the following parameters: Residual BER, SDU error ratio and Transfer Delay, due to the packet lost and the delay inside the Core Network. The SGSN sends a *RAB Assignment Request* RANAP message to the RAN through the Iu interface. This message contains the RAB QoS attributes.

Step 8 – The RAN, both UTRAN and GERAN, performs an admission control. RAN must do the mapping between RAB and RB parameters. When requesting a QoS profile for a PDP context, the parameters should be requested for full headered IP packets. Since the header compression applies only at PDCP layer in radio Uu interface, the impact of the header compression is only taken into account in RAB to RB mapping, when the resources are requested. Therefore, RB resources should be reserved when RAN applies header compression according to the corresponding bitrate. Later, the RAN establishes the RAB with the selected cell. After that, the SGSN receives a *RAB Assignment Response* RANAP message.

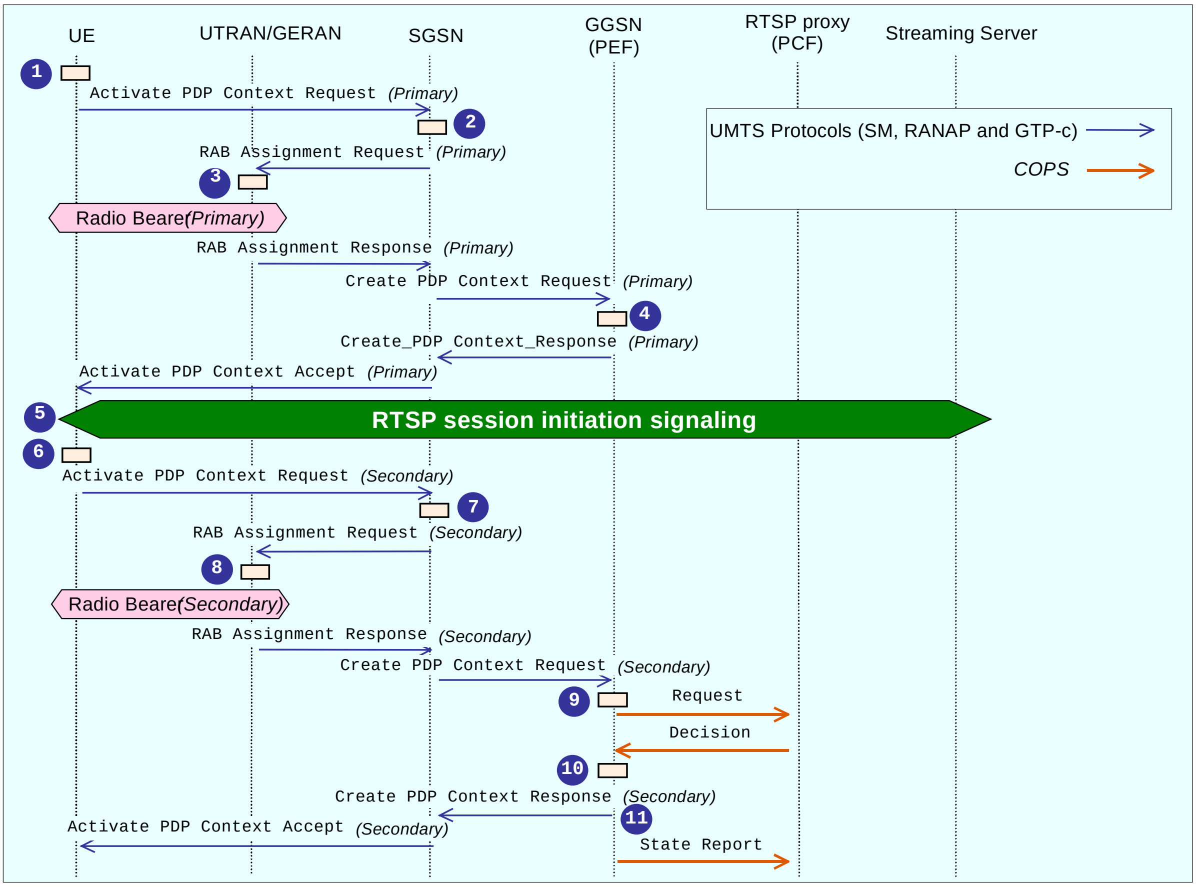

Fig 4. Multimedia streaming session initiation procedure in UMTS network

Step 9 – The SGSN sends a *Create PDP Context Request* message to the GGSN with the QoS negotiated, which generates a new entry in its PDP context table and stores the TFT. A local admission control and resource reservation are perfomed in the GGSN in the same way as SGSN. The *allocation/retention priority* parameter is also used in similar manner as for SGSN. Once the local admission control is performed in the GGSN, it outsources the admission control to the PCF in the RTSP proxy by sending a COPS message. The PCF applies appropriate rules to the service and sends its decision back to the GGSN.

Step 10 – The GGSN replies with a *Create PDP Context Response* GTP-c message to the SGSN. Likewise, the SGSN replies with an *Activate PDP Context Accept* SM message to the UE. The SGSN is now able to route PDP PDUs between the GGSN and the UE, and to start charging.

Step 11 – The GGSN reports the success of the secondary PDP context activation procedure to the SGSN and the PCF in the CSCF. Finally, the SGSN sends the corresponding SM message to the UE, so that it knows the end of the service session establishment.

IV. Service utilization

Once the connection is established, the RTP data flow needs an appropriate QoS provisioning. In this work, it is assumed that IP transport domain (i.e. both CN and IP based RAN) employs DiffServ mechanism [13][14], which is based on different Per Hop Behaviours (PHB). Each PHB consists of the rules used to treat packets in specific ways inside the network. More specifically, PHB denotes a combination of forwarding, classification, scheduling and drop behaviours at each hop. For streaming traffic, two groups of PHB can be applied: Expedited Forwarding (EF) or Assured Forwarding (AF). EF PHB target is to provide tools to build a low loss, low latency, low jitter, assured bandwith end-to-end service within the DiffServ domain, with the drawback of the complexity it introduces in the system. Due to the loose QoS requirements of streaming services, mainly in comparison with other real-time traffic like VoIP services, the AF PHB can be used. Inside AF PHB group there are a number of PHB delay classes, each with a number of drop precedence levels. For streaming traffic highest priority should be used.

In the radio subdomain there are basically two options: CS bearer or PS bearer. The CS approach has the inherent drawback of waste of resources, mainly in case of bursty traffic. Streaming traffic analysis shows its bursty nature. In fig. 5, the graph above presents the fluctuation of the bitrate for such a traffic source, whereas the graph below depicts the queue status. The source generates traffic at an average rate of 64 kbps. Due to the variation of the source rate, the existence of less activity periods (bitrate below the average one) is observed. Likewise, when these periods are large enough the queue gets empty (see highlighted parts in fig. 5). Therefore, in case a dedicated capacity of 64 kbps is allocated for such connection there is a waste of resources. In other words, if resources are shared, trunking gain is obtained.

Since 3G mobile networks are going to support multi-radio technologies, such as Wideband Code Division Multiple Access (WCDMA) and Enhanced Data rates for GSM Evolution (EDGE), in this work the QoS provisioning in both radio technologies is briefly outlined.

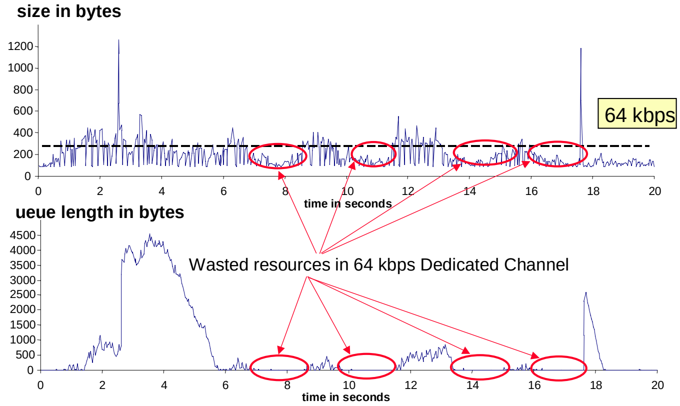

Fig 5. Bursty nature of streaming traffic

In UTRAN there are basically two types of bearers: Dedicated Channel (DCH) or Data Shared Channel (DSCH). Otherwise, GERAN provides different bearers which can support streaming services: Traffic Channel (TCH) like High Speed CS Data (HSCSD) and Enhanced CS Data (ECSD) from CS domain, or Packet Data Channel (PDCH) from PS domain.

As aforementioned, the use of shared resources gives to operators higher trunking gain. The challenge comes from the need of guaranteeing certain bandwidth on shared channels whose radio link capacity is continously varying, so enhanced QoS mechanisms are needed for that purpose. Hence, coordination between admission control and resource allocator as well as packet scheduling and link adaptation algorithms is required.

When the negotiated QoS during the service establishment can not be maintained by any network entity, different QoS control mechanisms have to be employed. Subsequently, some control plane signaling activity is needed to coordinate all these mechanisms, especially in order to provide a seamless end-to-end service bearer from the user point of view. The control plane activity when QoS degradation occurs can be divided into two different groups of mechanisms: preserving and renegotiation mechanisms.

The QoS preserving mechanisms are transparent to UE. For example, some RAN internal mechanisms are able to detect radio link degradation, so that specific control plane signaling procedures are triggered to successfully recover the negotiated QoS (e.g. by means of radio resources reallocation or cell reselection). When the first type of mechanisms can not successfully keep the negotiated QoS, it is possible to renegotiate a downgraded QoS profile with the UE. Hence, this group of mechanisms are not transparent to the UE.

V. CONCLUSIONS

Since supporting reliable real time services is decisive aspect for packet based telephony networks, an end-to-end QoS framework for streaming services in 3G mobile networks is considered. This work addresses a solution based on a PLMN-hosted multimedia streaming service. Signaling flowcharts have shown that UMTS and IETF's protocols can co-operate to provide seamless end-to-end real-time services. Thus, service activation have been described at three levels: the initiation from UE viewpoint, the RSTP signaling interchanges between application entities and the UMTS signaling procedures.

Provisioning of audio streaming services over 3G mobile networks have been also tackled in this paper. Results from traffic behaviour analysis have shown the convenience of using PS bearers. In case of shared channels, the challenge of assuring capacity for such traffic have also been pointed.

Nowadays, there are basically two trends among operators about the type of multimedia services to be supported in 3G mobile networks. In one hand, at the beginning operators may pilot the service with modified proprietary streaming technologies, which provide brand awareness and added value from the access to existing service provider. However, proliferation of IETF standardized protocols, such as RTSP, and aims to standardize an open streaming concept in major wireless standardization organizations (3GPP, 3GPP2) are bringing a strong open standard based service to the wireless marketplace, which provides a better environment for creating productive business with widespread wireless streaming services.

There remain a couple of issues that still need to be solved. So, audio streaming lacks a clearly identified codec for standardization and it is not yet sure that a standardized streaming service could effectively handle also scenarios where file downloading or HTTP streaming is used instead of pure streaming, what depends quite a lot on whether a standardized file format can be defined for multimedia content storage.

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