

A Framework for Unified IP QoS Support Over UMTS and Wireless LANs

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Abstract

The framework for a unified QoS support mechanism for IP traffic over UMTS and Wireless LANs is discussed. The work addresses the potential issues that emerge when using UMTS and WLANs as access systems to an IP core and focuses on defining a consistent mechanism that will support QoS and resource reservation in a unified. The proposed approach utilizes the RSVP for negotiating QoS parameters and reserving resources in an end-to-end basis. A proper mapping of QoS parameters of RSVP parameters on UMTS and 802.11e QoS parameters is presented and the required signalling exchange is described in detail.

1. Introduction

As the Internet technologies evolve, more sophisticated and Quality of Service (QoS) demanding multimedia services are being requested by the users. The Internet Protocol (IP), together with its QoS enhancement frameworks (namely the Integrated Services - IntServ - and the Differentiated Services - DiffServ), is currently the main transport technology for supporting all these new services and in this respect the motto “Everything over IP and IP over everything” has become the trend of the day. On the other hand, both UMTS and Wireless LANs (WLANs) are already commercially available and become increasingly popular. The number of mobile users is growing rapidly and so does the demand for wireless access to the Internet services, imposing the need for a unified QoS support framework in both UMTS and WLANs.

Despite the initial impression, expressed by several network technology vendors, that UMTS and WLANs will be competing technologies it appears that they can be combined and complement each other in an effective way. The approach followed in this work is that both UMTS and WLANs can act as access systems to one common IP core network, efficiently covering both wide areas and hot-spots. One of the main requirements of this system is a unified QoS support for IP traffic. Assuming the use of IntServ in the IP layer, which is more suitable for access systems, the paper discusses how this can be supported by the lower layers of either the UMTS or the WLANs. As RSVP is considered the dominant signalling protocol of IntServ, the discussion focuses on the adoption of RSVP messages and parameters by UMTS or WLAN QoS mechanisms.

The rest of the paper is organized as follows. Section 2 describes the referenced system architecture. Section 3 starts with a brief description of the UMTS QoS system, and continues by describing the required parameters mapping and messages translation for IP QoS support over UMTS. In section 4, after a brief description of the

802.11e QoS system, the same mapping and translation is described for IP QoS support over WLANs. Finally, section 5 includes the conclusions and future plans.

2. System Architecture

The IntServ framework provides the ability for applications to choose among multiple, controlled levels of delivery of service for their packets [1]. So far in this framework two QoS Control services have been defined, namely the Controlled Load [2] and the Guaranteed QoS [3]. The Controlled-Load service is defined to provide an approximation of a lightly loaded network. It is intended to support applications that are highly sensitive to overload conditions. Such applications are the so-called “adaptive real-time applications”. These applications work well on unloaded networks but degrade quickly when the network is overloaded. The Guaranteed service is defined to provide guaranteed bandwidth and a bounded queuing delay. The Guaranteed QoS service guarantees that datagrams will arrive within the guaranteed delivery time and will not be discarded due to queue overflows, provided that the traffic stays within its predefined traffic parameters values. This service is intended for applications, which need a firm guarantee that a datagram will arrive no later than a certain time after it was transmitted by its source. Examples of such applications are streaming audio/video applications that are very sensitive to delay.

Figure 1 depicts the referenced architecture. While UMTS provides wireless access to Mobile Terminals (MTs) in a wider area, WLANs are designated to provide access within smaller areas with dense presence of mobile users, such as hot-spots (e.g., trade centers, metro stations, airports) or company premises. As shown in Figure 1, UMTS is connected to the IP core through a GGSN entity that implements the IP translation function and the IP Bearer Service manager as described in [4]. Each WLAN is also connected to the IP core through an Edge Router (ER) that acts as a gateway to/from the core network.

MTs are considered dual-mode, operating in both UMTS and WLAN areas. The applications running at the MTs should be able to access all services provided by the core IP network and request a certain level of QoS from both the UMTS and WLANs. The QoS requirements in the case of both UMTS and WLANs are requested and reserved with the use of RSVP [5]. The MTs should be able to roam seamlessly through the coverage areas of both UMTS and WLANs using the same subscription and profile information and maintaining the same level of service. Nevertheless, service degradation should be acceptable in transitions from WLANs to the UMTS, due to the difference in the available bandwidth. In order to achieve these requirements, UMTS and WLANs interwork in a loosely coupled manner as described in [6], sharing the same subscriber database that is stored in the core UMTS network. This allows for the Authentication, Authorization, and Accounting (AAA) functions in both networks. Further to this, both networks should be able to provide similar services and, more importantly, interpret the application and user requirements in a consistent way, so that an MT roaming from one network to the other will not experience a dramatic degradation of service quality.

These considerations raise the following issues that should be taken into account:

- a. Creation of a common AAA mechanism that will be accessible from both UMTS and WLANs.
- b. Unified IP QoS support over both UMTS and WLANs
- c. Consistent definition and translation of services and user/application requirements into QoS parameters in both UMTS and WLANs.
- d. Seamless handover/roaming between UMTS and WLANs.

The work presented here focuses mainly on issues (b) and (c). Unified QoS support is attained by the common use of RSVP in the IP layer, while the mapping of RSVP parameters in UMTS and WLANs parameters is analysed below.

3. IP QoS Support Over UMTS

The QoS architecture in UMTS is based on the concept of *Bearer Service (BS)*, following a layer-based approach. As stated in [7], in order to realize a certain level of QoS, a BS with clearly defined characteristics and functionality has to be set up from the source to the destination of the service. The QoS architecture of UMTS follows a layer-based approach. Each layer consists of one or more BSs that are concatenated in order to provide the desired QoS support on an end-to-end basis. The UMTS BS provides QoS support within the UMTS network, i.e., from the Mobile Terminal (MT) to the Core Network (CN) Gateway. UMTS BS is further divided into the *Radio Access Bearer Service*, which provides confidential transport between MT and CN Iu Edge Node, and the *Core Network Bearer Service*, which connects the CN Iu Edge Node with the CN Gateway and through this with the external network.

Four different QoS classes are defined for the UMTS BS, taking into account the nature of the traffic produced by the various applications that are used over UMTS, as well as the restrictions and characteristics of the wireless interface.

The four classes are defined as follows [7]:

- **Conversational Class.** It includes real-time services such as speech and videoconferencing. Services belonging to the conversational class have strict transfer delay constraints.
- **Streaming Class.** This class applies to services where a stream of real-time data is destined to a user (e.g., video or audio streaming). The flow of data is unidirectional while the destination is usually a human. In contrast to the conversational class, delay constraints are not very strict, but the need for preserving the time relation (delay variation) remains important.
- **Interactive Class.** Applications such as web browsing or database retrieval fall into this class. There is an interactive scheme involved with the end-user requesting data from a remote host or equipment.

In this case, a key aspect is the round-trip delay, which should be kept within reasonable limits.

- **Background Class.** This class includes services such as e-mail, or file downloading. All these services are characterized by the fact that the end-user is expecting the data within a large period of time; therefore time constraints do not normally apply. However the content of the data packets should be preserved so the bit error rate should be kept low.

The QoS attributes for UMTS BS [7] are briefly described below.

Traffic Class: It defines the type of application (class) for which the bearer service is optimized. UMTS defines four traffic classes, as described earlier.

Maximum Bit-rate: It is the maximum number of bits delivered by UMTS and to UMTS at a particular Service Access Point (SAP) within a certain period of time divided by the duration of the period. The traffic is conformant with the maximum bit-rate, as long as it follows a token bucket algorithm where token rate equals maximum bit-rate and bucket size equals maximum SDU size.

Guaranteed Bit-rate: It is the guaranteed number of bits delivered by UMTS at a particular Service Access Point (SAP) within a certain period of time divided by the duration of the period (provided that there are data to deliver). The traffic is conformant with the guaranteed bit-rate as long as it follows a token bucket algorithm where token rate equals guaranteed bit-rate and bucket size equals Maximum SDU size.

Delivery Order: Indicates whether the bearer service should provide in-sequence delivery of application data packets, referred to as Service Data Units (SDUs).

Maximum SDU Size: Indicates the maximum SDU size.

SDU Format Information: Defines a list of possible exact SDU sizes. This information is usually used by the UMTS radio access network in order to achieve spectral efficiency and lower transfer delay.

SDU Error Ratio: It indicates the acceptable fraction of SDUs that can be lost or detected as erroneous.

Residual Bit Error Ratio: Indicates the undetected bit error ratio in the delivered SDUs. If no error detection is requested, residual bit error ratio indicates the bit error ratio in the delivered SDUs.

Delivery of Erroneous SDUs: Indicates whether SDUs detected as erroneous shall be delivered or discarded.

Transfer Delay: Defines maximum delay for the 95th percentile of the distribution of delay for all delivered SDUs during the lifetime of a bearer service.

Traffic Handling Priority: Specifies the relative priority of an interactive-class bearer with respect to other interactive-class bearers.

Allocation/Retention Priority: Specifies the relative importance of allocating and retaining a UMTS bearer compared to other UMTS bearers. It is used for admission control during congestion periods.

For each traffic class within UMTS a different set of QoS attributes is applicable. Table 1 shows which parameters are applicable to each traffic class.

The interworking of UMTS with IP for End-to-End QoS support is described in [4]. The proposed interworking architecture is shown in Figure 2. The main functional entities regarding the QoS support are the IP Bearer Service (BS) Manager and the Translation/Mapping function that are present both in GGSN as well as in the User Equipment (UE). The IP BS Manager uses standard IP mechanisms to manage the IP bearer services. Its main function is to interface UMTS with the IP core and the mechanisms used may be different from the internal mechanisms used within the UMTS. The Translation Function interacts with the IP BS Manager and provides the interworking between the mechanisms and service parameters used within UMTS BS and those within the IP Bearer Service. In the GGSN the Translation Function maps the IP QoS parameters to adequate UMTS QoS Parameters while in the UE the user/application QoS parameters are mapped to either PDP context parameters or IP layer parameters (e.g., RSVP). It is evident that the translation function is an integral part of the UMTS - IP QoS interworking since, without proper mapping, UMTS will fail to provide a bearer service that will be consistent and aligned with the corresponding IP BS of the IP core.

3.1 Proposed Mapping of RSVP Parameters to UMTS QoS Parameters

The QoS parameters for Controlled-Load and Guaranteed QoS Services are described in detail in [2] and [3] respectively. Both these services are invoked by specifying the desired Traffic Specification (TSpec) parameters to the network elements. The TSpec takes the form of a token bucket specification plus a **peak-rate (p)**, a **minimum policed unit (m)**, indicating the minimum packet size generated by the application, and a **maximum packet size (M)**. The token bucket specification includes a **bucket rate (r)** and a **bucket length (b)**. When invoked using RSVP, both services make use of the TOKEN_BUCKET_TSPEC parameter included in SENDER_TSPEC and FLOWSPEC RSVP objects as described in [1]. Additionally, for Guaranteed QoS service the desired service parameters (RSpec) are required. The desired service parameters consist of a **desired rate (R)** and a **slack term (S)**, which signifies the difference between the desired delay and the delay obtained by using the reservation level (R).

In order to achieve the same level of QoS within UMTS as in the IP Core, all these service parameters should be adequately mapped to appropriate UMTS QoS Parameters. A possible mapping would be as follows:

- **Bucket rate (r):** Since it reflects the mean rate of the application data or the desired reservation rate from the receiver's perspective, it can be mapped to the **Guaranteed Bit-Rate** QoS attribute of the UMTS
- **Peak-Rate (p):** In a similar manner with bucket-rate (r) it can be mapped to **Maximum Bit-Rate** parameter of UMTS.
- **Bucket length (b):** It can be used to set the size of the bucket size for the token bucket algorithm of UMTS. It should be noted however that, regardless of the bucket length, within UMTS, conformance of traffic with Guaranteed and Maximum Bit-Rate is measured against a bucket size that equals the Maximum SDU Size

- **Maximum packet size (M):** It can be directly mapped to the respective **Maximum SDU Size** parameter of UMTS. Based in the previous remark for Bucket Length parameter, maximum Packet Size can also be used to define the bucket size for the traffic conforming to both Maximum and Guaranteed Bit Rate.
- **Minimum policed unit (m):** This parameter is used to calculate minimum resource allocation requirements for a flow and for polishing purposes. This parameter is very useful since it provides a lower bound for the minimum packet size thus it facilitates a more realistic estimation of bandwidth overhead due to the link-level headers added to IP datagrams. Although Minimum policed unit (m) cannot be directly mapped to a UMTS QoS parameter it can be utilized by the UMTS BS Service and Admission control for reserving the correct amount of bandwidth for a flow.

In case of QoS Guaranteed Service request, the additional parameters **desired rate (R)** and **slack term (S)** of RSpec should also be utilized as follows:

- **Desired Rate (R):** Since this parameter indicates the desired mean rate of a data flow, it can be mapped on the Guaranteed Bit-Rate (instead of the Bucket Rate (r) parameter) providing a more accurate value.
- **Slack Term (S):** This parameter signifies the difference between the desired delay and the delay obtained by using a reservation rate R. This term allows the network, in case of congestion, to reduce the resources allocated to a flow without violating the delay constraints of the flow. Within UMTS this parameter can be used in conjunction with the **Transfer Delay** parameter of UMTS and be used by UMTS BS Manager when re-allocating resources to a bearer.

The signalling exchange during the establishment of a new service, using RSVP, is shown in Figure 3 and Figure 4. The general description of the signalling sequence can be found in [4], leaving however enough scope for modifications and adjustments depending on the specific requirements of the requested service. The proposed approach here is that PDP Context activation takes place after the reception of the RSVP-RESV message indicating the desired traffic and QoS parameters of the receiver. In this way, the signalling sequence for outgoing (downlink) flows is as follows (Figure 3):

1. User Equipment (UE) issues the RSVP-PATH through the GGSN. The GGSN may or may not process the RSVP-PATH message before forwarding it to the next hop.
2. GGSN receives the RSVP-RESV message. The IP Bearer Service entity of the GGSN processes the QoS parameters of the message, which is then forwarded to the UE.
3. UE receives the RSVP-RESV message. IP Bearer Service entity processes the QoS parameters, which are then translated to UMTS parameters by the Translation function.
4. UE invokes PDP Context activation by issuing the Activate PDP Context message to SGSN.
5. The SGSN send the corresponding Create PDP Context to GGSN.

6. The GGSN authorizes the PDP context activation request according to the local IP Bearer Resource Policy and the QoS parameters authorized by the IP Bearer Service entity upon reception of the RSVP-RECV message (see step 2).
7. Radio Access Bearer (RAB) Setup is performed by the RAB Assignment Procedure.
8. The SGSN sends an Activate PDP Context Accept message to the UE.
9. The UE sends a RSVP-RESV-CONF message to the next hop confirming the resource reservation.

Notice that, unlike the normal case for RSVP in wired links, the last RSVP-RESV message in the wireless hop does not in fact reserves the resources on the wireless link but rather triggers the UE to request a new reservation. In the PDP Context Activation message the UE has to make use of the translated QoS parameters provided by the Translation Function while the GGSN has to store the QoS parameters derived from the RSVP-RESV message and reserve resources based on these parameters.

In case of an incoming (downlink) flow the signalling exchange is as follows (Figure 4):

1. The UE receives the RSVP-PATH through the GGSN. The GGSN may or may not process the RSVP-PATH message before forwarding it to the UE.
2. The IP Bearer Service entity of the UE processes the QoS parameters of the message and the Translation Functions translates them to appropriate UMTS QoS Parameters. The UE then invokes PDP Context activation by issuing the Activate PDP Context message to SGSN.
3. The SGSN send the corresponding Create PDP Context to GGSN.
4. The GGSN authorizes the PDP context activation request according to the local IP Bearer Resource Policy and the QoS parameters indicated by the UE IP Bearer Service entity upon reception of the RSVP-PATH message (see step 2).
5. Radio Access Bearer (RAB) Setup is performed by the RAB Assignment Procedure.
6. The SGSN sends an Activate PDP Context Accept message to the UE.
7. The UE issues a RSVP-RESV message through the GGSN. The GGSN may or may not process the RSVP-RESV message before forwarding it to the next hop.
8. The UE receives a RSVP-RESV-CONF message confirming the resource reservation.

4. IP QoS Support Over WLANs

Recent initiatives try to extend traditional WLANs to provide efficient QoS. A major such initiative is the IEEE 802.11e standard [8], which forms a QoS extension for legacy IEEE 802.11. In 802.11e the QoS mechanism is supervised by the Hybrid Coordinator (HC) entity, which implements the Hybrid Coordination Function (HCF). The HC is typically located at the Access Point (AP) and utilizes a combination of a contention-based, called Enhanced Distributed Coordination Function (EDCF), and a polling-based scheme to provide QoS-enhanced access to the wireless medium. The EDCF implements QoS disciplines by introducing classification and

prioritisation of the different traffic categories. In a similar manner, the polling mechanism implemented by the HCF allows the HC to assign Transmission Opportunities (TXOPs) to the Mobile Stations (STAs). A TXOP is a period of time in which a STA or the HC can transmit a burst of data frames. The TXOPs are always initiated and assigned by the HC. The assignment of TXOPs to a STA may again be based on traffic classification as well as on current traffic requirements of the STA.

The polling-based QoS mechanism is applied per Traffic Stream (TS). A TS is a set of MSDUs to be delivered subject to the QoS parameters values provided to the MAC layer in a particular traffic specification (TSPEC) element. The main recommended parameters of TSPEC as described in [9] are:

Mean data rate (ρ): average bit rate for transfer of the packets, in units of bits per second.

Delay bound (D): maximum delay allowed to transport a packet across the wireless interface (including queuing delay), in milliseconds.

Nominal MSDU size (L): nominal size of the packets, in octets.

User priority (UP): priority to be used for the transport of packets in cases where relative prioritisation is required (e.g., it can be used for admission control). It is based on IEEE 802.1d priority levels, which go from 0 (lowest) to 7 (highest).

Maximum MSDU size (M): maximum size of the packets, in octets.

Maximum Burst Size (MBS): maximum size of the data burst that can be transmitted at the peak data rate, in octets.

Minimum PHY rate (R): physical bit rate assumed by the scheduler for transmit time and admission control calculations, in units of bits per second.

Peak data rate (PR): maximum bit rate allowed for transfer of the packets, in units of bits per second.

These parameters are incorporated by the so-called *Hybrid Coordinator (HC)*, which dynamically decides on the allocation of the radio resources per TS.

4.1 Proposed Mapping of RSVP Parameters to WLAN QoS Parameters

Since the proposed 802.11e TSPEC parameters are quite similar with the IP parameters defined for both IntServ and DiffServ the mapping of the RSVP QoS parameters to the 802.11e TSPEC parameters is as follows:

- **Bucket rate (r):** It can be mapped to the **Mean Data Rate (ρ)** parameter of the 802.11e TSPEC.
- **Peak-Rate (p):** Similarly it can be mapped to **Peak Data Rate (PR)** parameter of the 802.11e TSPEC.
- **Bucket length (b):** This parameter can be correlated with the **Maximum Burst Size (MBS)** or **Maximum MSDU Size (M)** parameters of the 802.11e TSPEC. Both of these parameters can be used to define the minimum bucket length required for the conforming traffic of a TS or, in reverse, in absence of these two parameters, the Bucket length (b) can provide a rough estimate of their values.
- **Maximum packet size (M):** This parameter directly corresponds to the **Maximum MSDU Size** parameter of 802.11e TSPEC.

- **Minimum policed unit (m):** It corresponds to the **Nominal MSDU Size (L)** parameter of 802.11e TSPEC and can be used to calculate minimum resource allocation requirements for a flow and for polishing/Call Admission Control purposes.

In case of QoS Guaranteed Service request, the additional parameters **desired rate (R)** and **slack term (S)** of RSpec can also be mapped as follows:

- **Desired Rate (R):** It can be mapped on the **Peak Data Rate (PR)** parameter of 802.11e TSPEC (instead of the Bucket Rate (r) parameter) providing a more accurate value for this parameter.
- **Slack Term (S):** Likewise the UMTS case, this parameter can be used in conjunction with the **Delay Bound (D)** parameter of 802.11e TSPEC. Each time the **Desired Rate (R)** and **Slack Term (S)** parameters are updated due to resource re-allocation for a TS, the value of S can be used for updating the Delay Bound (D) parameter.

The signalling exchange during a TS setup with the use of RSVP is similar with that of UMTS and is depicted in Figure 5 (incoming TS) and Figure 6 (outgoing TS). In the case of an outgoing TS setup the signalling exchange has as follows (Figure 5):

1. The STA Signalling Management Entity (SME) issues the RSVP-PATH through the Gateway. The Gateway may or may not process the RSVP-PATH message before forwarding it to the next hop.
2. Gateway receives the RSVP-RESV message and forwards it to HC SME. The HC SME processes the QoS parameters of the message, which are then forwarded to the STA SME.
3. The STA SME receives the RSVP-RESV message and processes the QoS parameters, which are then translated to IEEE 802.11e QoS parameters.
4. The STA SME invokes the TS setup procedure by issuing the MLME-ADDTS.request primitive to the STA MAC.
5. The STA MAC issues a ADDTS QoS Action Request frame to the HC MAC requesting a TS setup.
6. The HC MAC forwards the MLME-ADDTS.indication primitive to the HC SME.
7. The HC SME authorizes the TS setup according to the QoS parameters authorized upon reception of the RSVP-RECV message (see step 2) and replies with a MLME-ADDTS.response primitive to HC MAC.
8. The HC MAC in turn forwards a ADDTS QoS Action Response frame to the STA SME.
9. The STA MAC sends a MLME-ADDTS.confirm to the STA SME confirming the successful TS setup.
10. The STA SME sends a RSVP-RESV-CONF message to the next hop through the Gateway confirming the resource reservation.

Note that, much like the case with UMTS, the RSVP-RESV (step 3) does not reserve resources over the wireless hop but triggers the TS setup at the STA SME. This is again necessary since according to [8] the initiation of a TS setup is always a responsibility of the non-AP STA regardless of the TS direction.

In a similar manner with outgoing TS setup, the signaling exchange during the setup of an incoming TS is shown in Figure 6:

1. The STA SME receives the RSVP-PATH through the Gateway. The Gateway may or may not process the RSVP-PATH message before forwarding it to the STA SME.
2. The STA SME processes the QoS parameters of the message and translates them to appropriate 802.11e QoS Parameters. The STA SME then invokes the TS setup procedure by issuing the MLME-ADDTS.request primitive to the STA MAC.
3. The STA MAC issues a ADDTS QoS Action Request frame to the HC MAC requesting a TS setup.
4. The HC MAC forwards the MLME-ADDTS.indication primitive to the HC SME.
5. The HC SME authorizes the TS setup according to the QoS parameters authorized upon reception of the RSVP-RECV message (see step 2) and replies with a MLME-ADDTS.response primitive to HC MAC.
6. The HC MAC in turn forwards a ADDTS QoS Action Response frame to the STA SME.
7. The STA MAC sends a MLME-ADDTS.confirm to the STA SME confirming the successful TS setup.
8. The STA SME issues a RSVP-RESV message through the Gateway. The Gateway may or may not process the RSVP-RESV message before forwarding it to the next hop.
9. The STA SME receives a RSVP-RESV-CONF message confirming the resource reservation.

Although important, the aforementioned mapping and signaling translation is not the only thing required for IP QoS support in WLANs. In contrast to the stable links used for fixed networks, bandwidth of wireless links is subject to variations, resulting in low performance of static resource reservation based schemes such as RSVP. For example, if available resources are reduced after admission control, the network may not be able to meet commitments for flows that have been successfully admitted. To handle this problem, extra mechanisms are required for the wireless link. These mechanisms include:

- An admission control algorithm, that considers not only the current available bandwidth, but also the expected instabilities of the wireless link. In this way, it might choose not to accept flows, even though there are available resources at the time of the request, if quality reductions are expected.
- A traffic-scheduling algorithm for the wireless medium, that is capable of arranging transmissions in such a way that the QoS is maintained for all flows. This algorithm should take as input the RSVP parameters (basically Tspec and Rspec) and try to maintain the agreed values under all traffic and channel conditions.
- A flow rejection algorithm that detects the instances where the overall bandwidth is reduced below the required limit and rejects a number of flows, to allow the rest to operate as required. The critical question for the algorithm is how many and which in particular should be rejected, that can cause as less inconvenience to the users and the network as possible.

All these mechanisms form an interesting research area for further extensions of WLANs.

5. Conclusions

A framework for unified QoS support for IP traffic over both UMTS and WLANs has been described. The presented conceptual study shows that it is feasible to provide a common QoS support mechanism for both platforms. The proposed mechanism utilizes the RSVP signaling for negotiating and reserving resources on an end-to-end basis with the RSVP signals and parameters adequately mapped to UMTS and WLANs QoS mechanisms. The merit of this approach is that it allows the mobile terminals and users to roam between the UMTS and the WLANs, using the same RSVP-based QoS system. The future plans include the development of an effective admission control and traffic-scheduling algorithm for the Hybrid Coordinator of 802.11e, and the definition of an efficient handover procedure that will allow the seamless transition of active users between the coverage areas of UMTS and WLANs.

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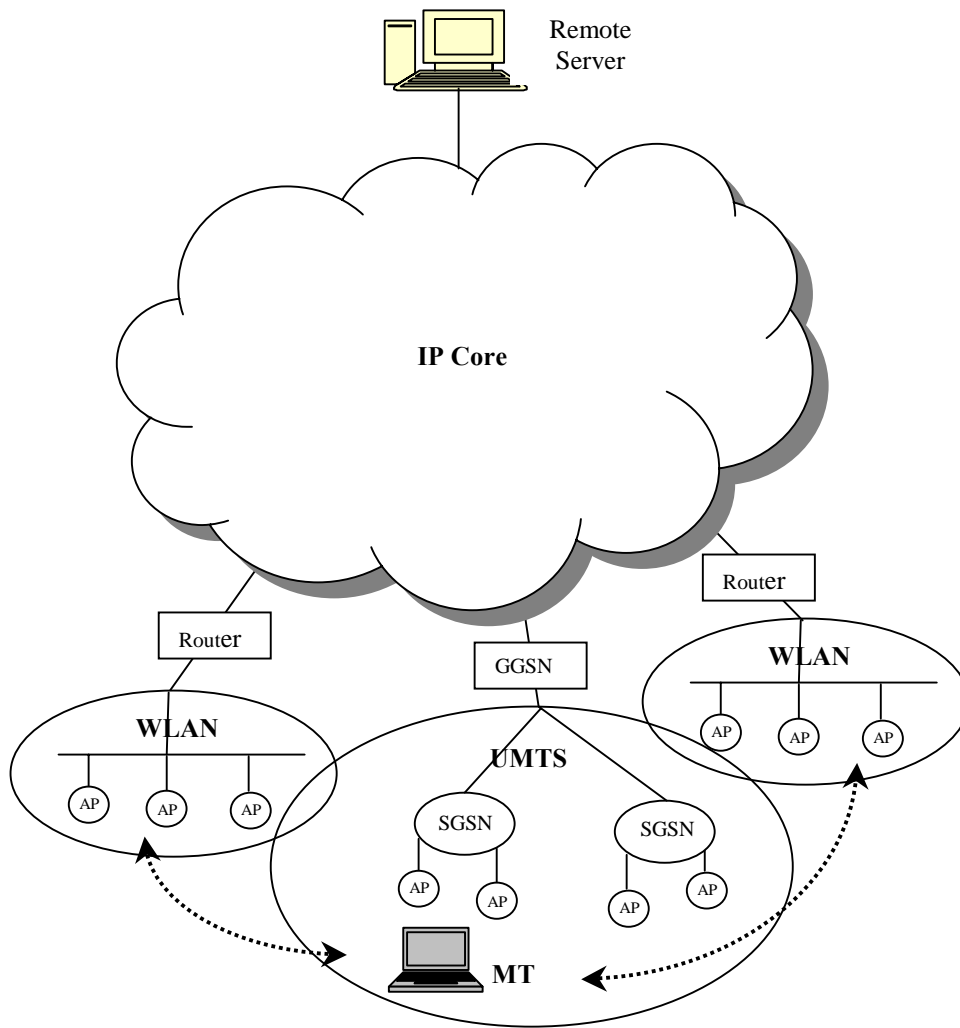


Figure 1. System Architecture

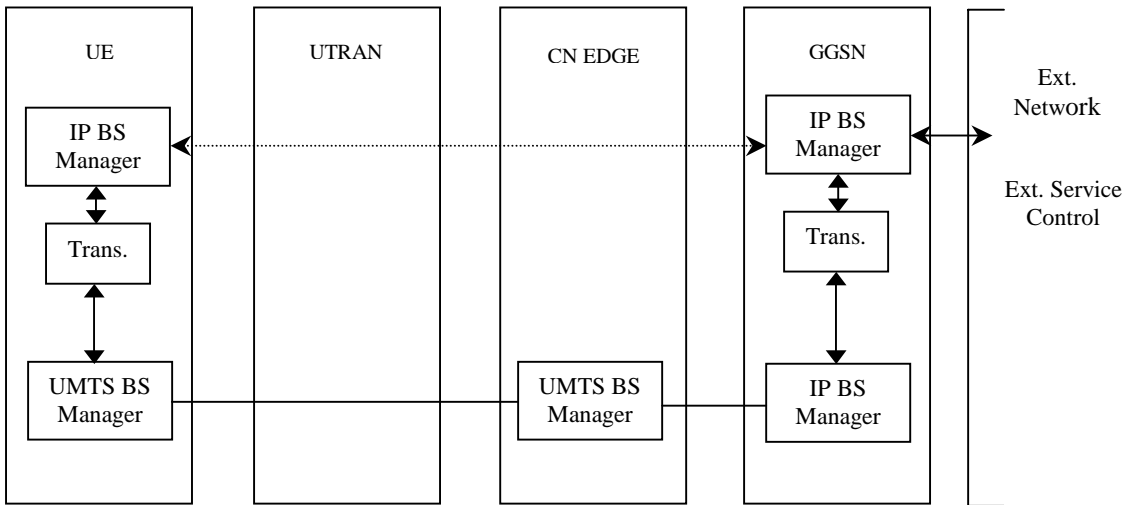


Figure 2. QoS Management Functions for UMTS-IP Interworking

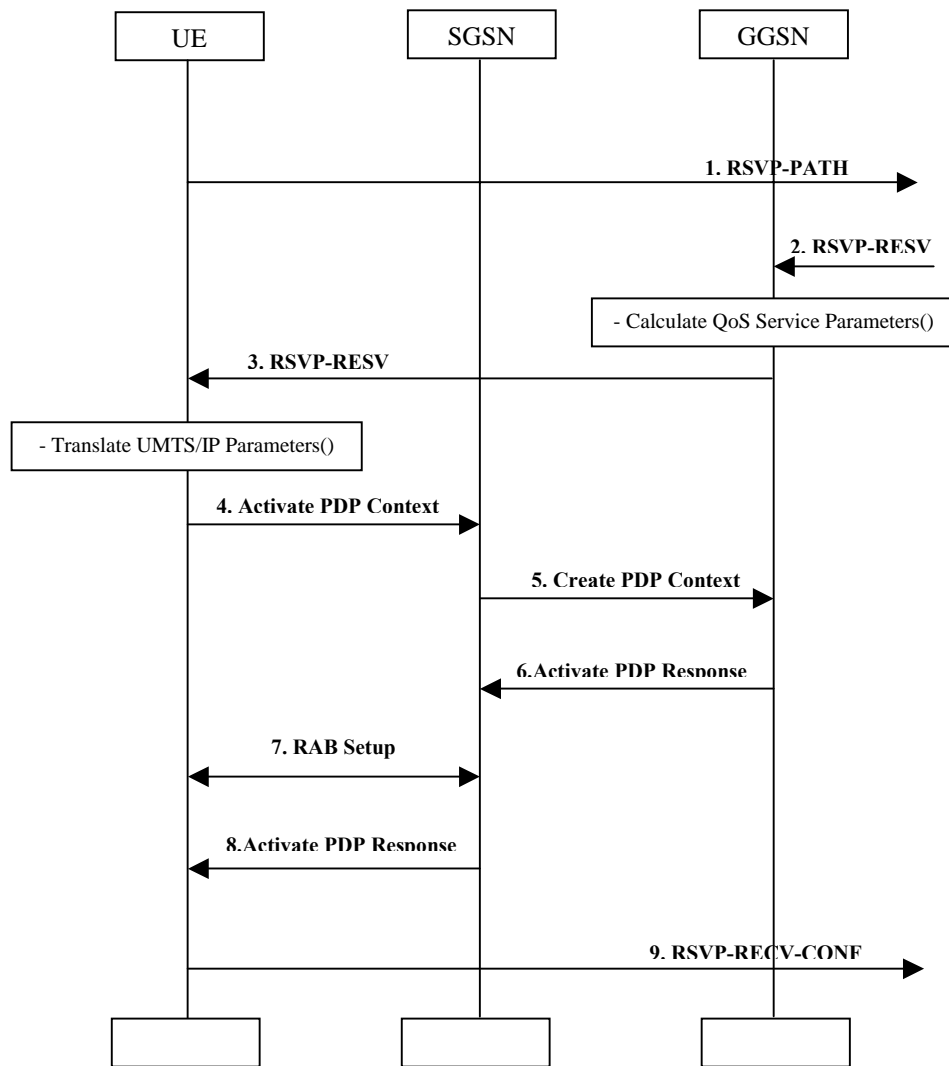


Figure 3. UMTS Resource Reservation using RSVP for Outgoing(uplink) flows.

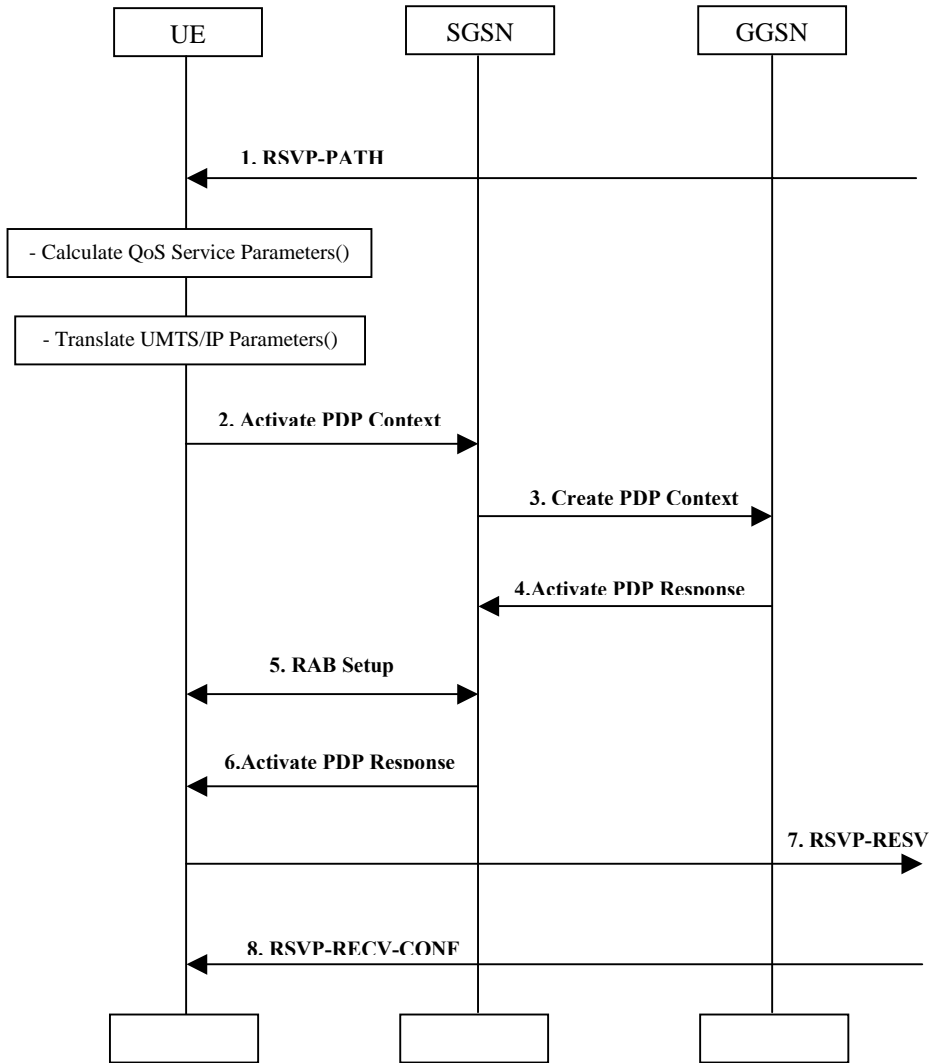


Figure 4. UMTS Resource Reservation using RSVP for Incoming (downlink) flows.

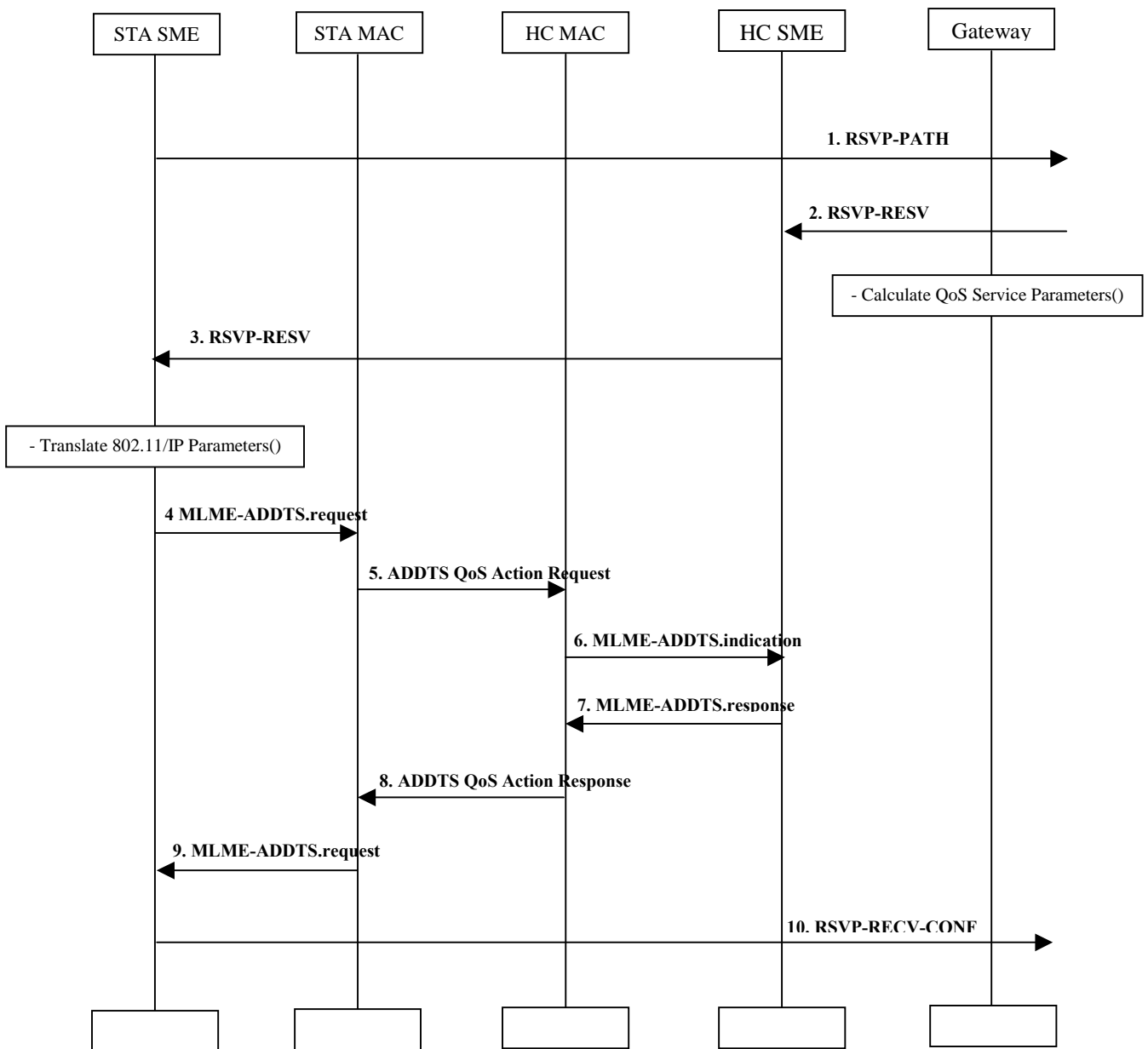


Figure 5. IEEE 802.11e Resource Reservation using RSVP for Outgoing(uplink) flows.

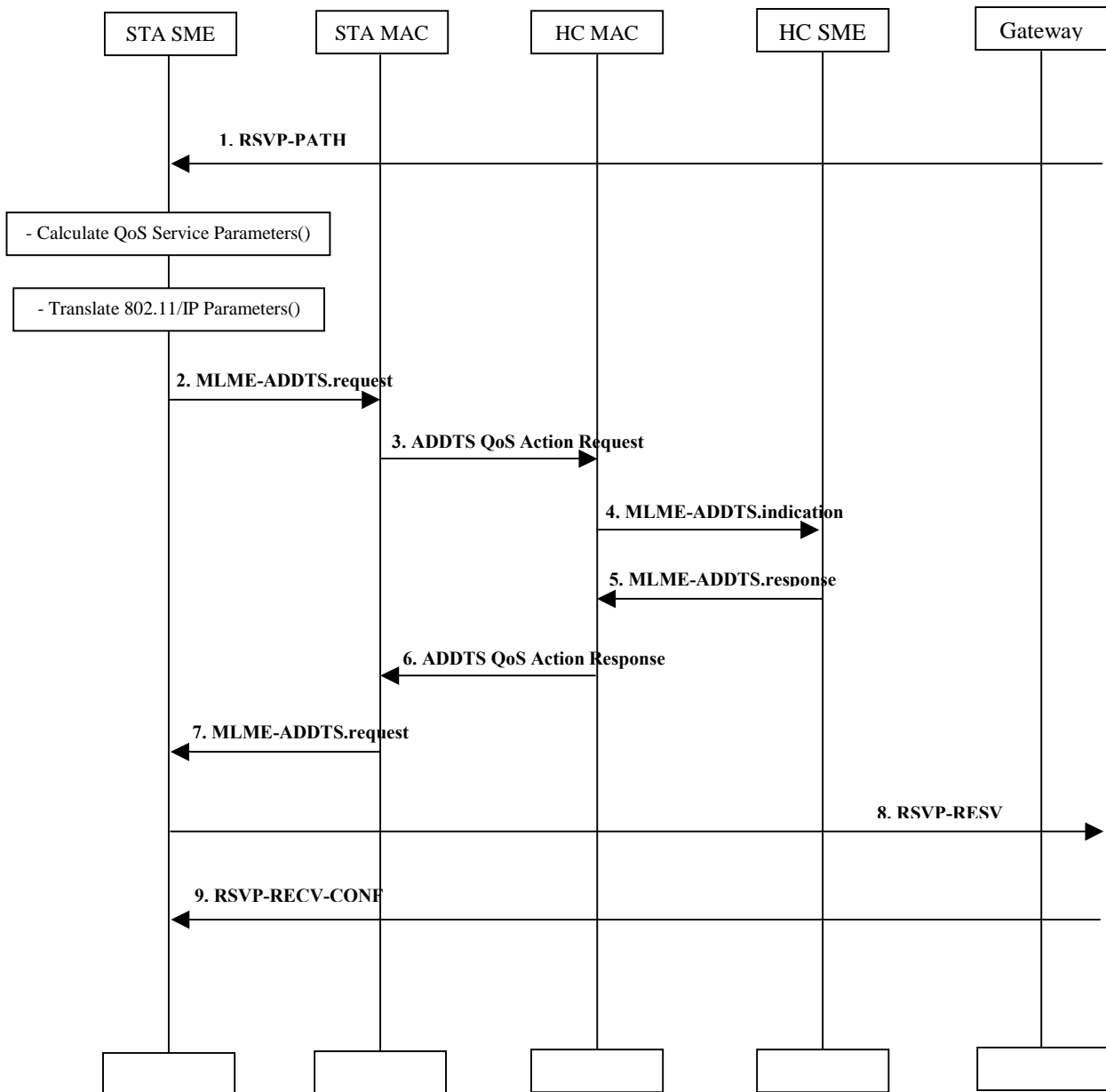


Figure 6. IEEE 802.11e Resource Reservation using RSVP for Incoming (downlink) flows.

QoS Attributes	Conversational Class	Streaming Class	Interactive Class	Background Class
Maximum Bitrate	X	X	X	X
Guaranteed Bitrate	X	X		
Delivery Order	X	X	X	X
Maximum SDU Size	X	X	X	X
SDU Format Information	X	X		
SDU Error Ratio	X	X	X	X
Residual Bit Error Ratio	X	X	X	X
Delivery of erroneous SDUs	X	X	X	X
Transfer Delay	X	X		
Traffic Handling Priority			X	
Allocation/Retention Priority	X	X	X	X

Table 1. QoS Attributes defined for each Traffic Class