QoS-Enabled Broadband Mobile Access to Wireline Networks

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ABSTRACT

Third-generation wireless systems, known as IMT-2000 within the ITU, offer opportunities to support a wide range of multimedia services. Packet data services will play a major role in these new multimedia services. A key component of packetized data services is to ensure end-to-end QoS requirements through efficient management of the network's resources. In this article we present an overview of radio resource scheduling schemes including architecture, radio interface protocol, and interactions in a wideband CDMA environment. We then present an example of QoS architecture followed by a discussion on end-to-end provisioning and interworking from wireless to fixed networks.

INTRODUCTION

During the last decade, there has been tremendous growth in two technological sectors: the Internet and wireless communications. The International Telecommunication Union (ITU) projects that by the end of 2002 there will be approximately 600 million Internet users. Running almost concurrently with this growth in the Internet has been the equally extraordinary growth in the number of mobile cellular networks and subscribers. It is expected that the number of mobile phones will be more than a billion by 2003. The Internet and wireless communications have conventionally been regarded as separate technologies. This is because originally the Internet was designed to carry mainly data traffic, while wireless networks were designed to carry mainly voice traffic. This boundary has become increasingly blurry in recent years, especially with the introduction of several proposals for International Mobile Telecommunications 2000 (IMT-2000) in the ITU [1, 2]. These third-generation (3G) mobile communications systems are poised to be multiservice platforms supporting voice, video, and data services with bit rates up to 2 Mb/s.

An important part of this evolution is quality of service (QoS), which is simply a set of service requirements to be met by the network while transporting a traffic stream from source to destination. QoS attributes are usually specified in terms of bit error rate (BER), guaranteed bit rate (R^G), delay, and so on. Essential to the notion of QoS is resource management that resolves users' competition to utilize network resources according to their QoS agreements.

According to major 3G partnership projects, code-division multiple access (CDMA) is the dominant air interface in 3G systems. That is why we focus in this article on resource allocation in a CDMA environment. We first present a brief overview of UMTS architecture. Next, we review CDMA radio resources scheduling followed by different QoS classes' requirements and some thoughts on QoS mapping from radio access to fixed core networks. We then close with some concluding remarks.

SYSTEM ARCHITECTURE

A simplified architectural model of the Universal Mobile Telecommunication System (UMTS) [1] is shown in Fig. 1. It mainly consists of three components: a wireless mobile station (MS),[1] a radio access network (RAN), and a core network (CN). RAN, which performs all radio-access-specific procedures, can be viewed as a network extension providing wireless access to the core network (CN).

This wireless extension essentially consists of a set of interconnected radio network systems (RNSs). Each RNS is responsible for the resources of its set of cells and contains several base stations (BSs) connected through a radio network controller (RNC). Each BS serves a group of MSs currently residing in a cell and is responsible for intracell control, while RNC is in charge of the intercell operations like handover decisions. On the fixed network side, CN consists mainly of some edge nodes (ENs) and gateway nodes (GNs) that interconnect with external networks like Internet Protocol (IP), integrated services digital network (ISDN), public switched telephone network (PSTN), or old mobile networks. Thus, in

[1] *The mobile terminal (MT) consists of terminal equipment (TE) in addition to a wireless mobile station (MS).*

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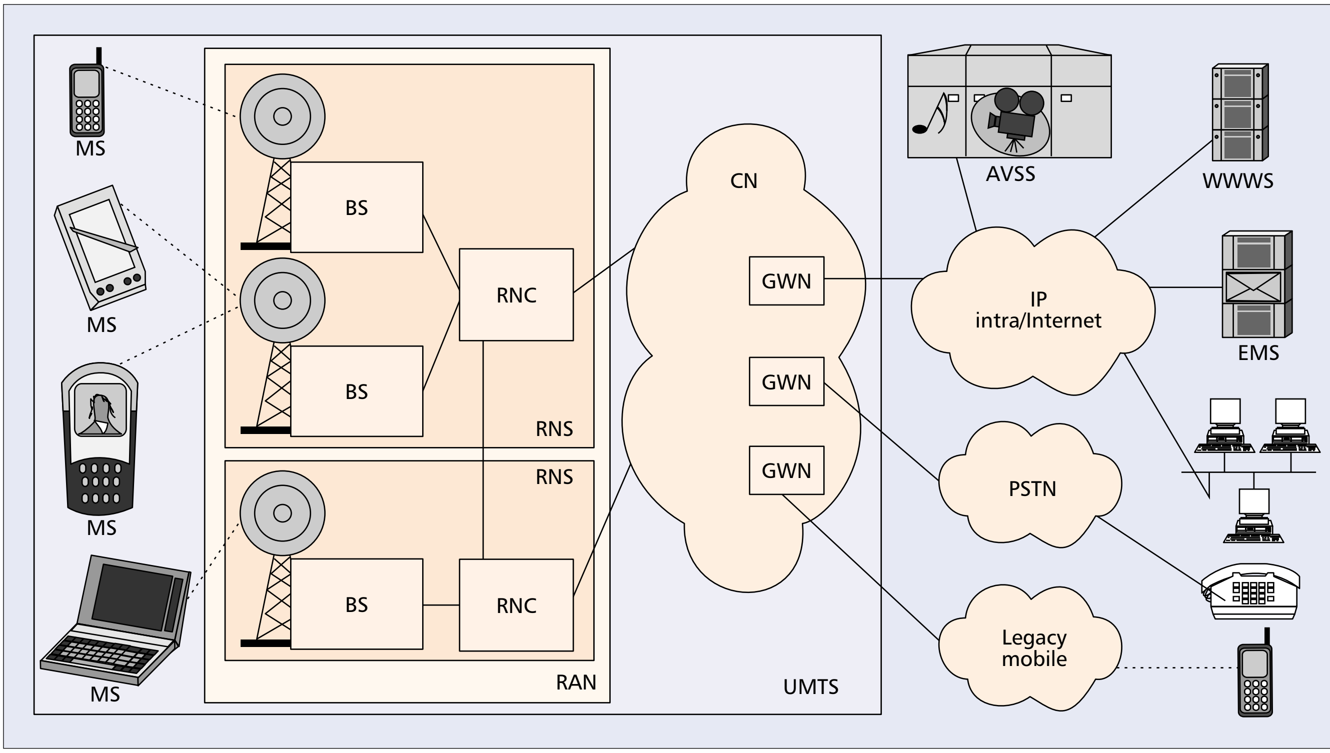


■ **Figure 1.** *Simplified UMTS architecture. MS: mobile station. BS: base station. RNC: radio network controller. RNS: radio network system. RAN: radio access network. CN: core network. GWN: gateway node. PSTN: public switched telephone network. AVSS: audio/video streaming server. WWWS: World Wide Web server. EMS: email server.*

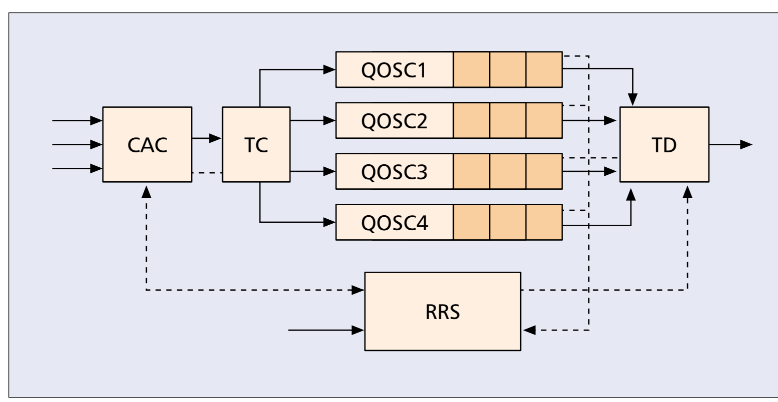
addition to being able to place voice calls, an MS has the opportunity to establish communication links with any type of network, including the Internet, to exchange multimedia contents, download emails, browse the Web, and so on.

RADIO RESOURCE MANAGEMENT

Today the Internet does not provide any QoS guarantees; however, this will change through mechanisms that provide some form of allocation of resources. In the UMTS architecture, each BS has a radio resource management module that attempts to preserve the traffic's QoS requirements across the RAN [3].

The RRM main role is to assign resources to users according to their QoS requirements. As Shown in Fig. 2, RRM mission starts by performing connection admission control (CAC). Since the decision is based on resource availability, CAC consults the radio resource scheduler (RRS) before accepting or rejecting the requested call. Upon call acceptance, the traffic classifier (TC), another RRM component, categorizes the incoming traffic according to its QoS specification, which is typically included in each packet header (e.g., IETF Diffserv service code point [4]). Data flows are then directed to a corresponding queue according to its Qos field. Each Qos class is represented by at least one queue. Finally, the traffic dispatcher (TD) drains the multiple queues according to some priority logic after getting the assigned radio resources from the RRS, which relies on the channel conditions and the requested QoS in its response.

Based on the above, it is evident that RRS bears great responsibility in having a successful RRM. That is why the next section is dedicated to it.



■ **Figure 2.** *Radio resource manager (RRM) [3]. CAC: connection admission controller, TC: traffic classifier, RRS: radio resource scheduler, η: interference and noise measurements, TD: traffic dispatcher, QOSCx: packets queue for QoS class x.*

RESOURCE SCHEDULING

An important goal in a multiple-access system, such as in the IMT-2000, is to maximize the number of simultaneous users it can accommodate. If each MS is assigned the minimum resources necessary for meeting or exceeding its QoS requirements, the capacity of the system will be maximized. That is why RRS is an essential component in QoS-aware BS, as indicated in the previous section and shown in Fig. 2. In a CDMA environment, RRS has two important radio resources to control: MS transmitting power P, and bit rate R. In this section we review some resource scheduling techniques.

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Standard Power Control

Power control is a means primarily designed to compensate for the loss caused by propagation and fading. It aims at preserving the BER as an important QoS measure, which depends in CDMA networks, on the received bit energy-to-noise density ratio E_b/N_0 given by

$$\left(\frac{E_b}{N_0}\right)_i = \frac{G_{bi}P_i / R_i}{\left(\sum_{j \neq i}^{M} G_{bi}P_j + \eta\right) \Big/ W} \tag{1}$$

W is the total spread spectrum bandwidth occupied by the CDMA signals. G_{bi} denotes the link gain on the path between MS i and BS b. η denotes background noise due to thermal noise contained in W, and M is the number of mobile users. MS i transmits information at bit rate, R_i, and power level, P_i, which is usually limited by a maximum value $P^{\max}$. Any increase in mobile power raises its E_b/N_0, but increases the interference to other users, causing a decrease in their E_b/N_0s and a compromise in the overall system capacity. Therefore, a dynamic scheme must be used to control power and prevent capacity degradation.

An example of a dominant RRS is the transmitter power control (TPC) specified within the Interim Standard 95 (IS-95) [5]. In fact, this scheme is currently operating in all CDMA-based systems and included as well in all wideband CDMA 3G proposals [1, 2]. In the closed loop mode (CLPC) of TPC, the BS measures the received E_b/N_0[2] over a 1.25 ms period,[3] and compares that to a target Θ. If the received $E_b/N_0 < \Theta$, a $\Delta P = 0$ is generated to instruct the mobile to increase its power, otherwise, a $\Delta P = 1$ is generated to instruct the mobile to decrease its power. These commands instruct the MS to adjust transmitter power by a predetermined amount, usually 1 dBm. Therefore, CLPC incrementally removes each MS excess power that primarily increases interference in the face of other users.

Nevertheless, as any other power-only control scheme, CLPC has a major shortcoming. Specifically, when link gain G_{bi} is too low at bad radio channel conditions, CLPC advises MS i to use high power levels causing excessive interference to other users. Moreover, the MS can reach its maximum transmitting power level, $P^{\max}$, without reaching its required Θ. This situation always leads to the drop of this MS connection when bad conditions persist for some time.

Combined Power and Rate Control

Giving Eq. (1) a closer look, we can deduce that bit rate control gives the RRS an additional tool to preserve QoS (BER), (i.e., E_b/N_0), and hence to overcome the CLPC limitation. Simply, if the BS RRS advises MS i to decrease its rate R_i at bad channel conditions instead of increasing its power P_i, the outage probability of MS i shall decrease. The outage probability, a measure of the user's satisfaction, is defined as the probability of having MS E_b/N_0 falling below its threshold Θ. Several recent studies have investigated this possibility. We discuss here some of them.

The authors in [6] proposed manipulating spreading gain in CDMA data networks besides power control. Spreading gain is simply the ratio of the system bandwidth (W) to the transmitter bit rate (R). Each BS appropriately balances a user's desire for high transmission power with the amount of interference it will generate to other users. However, the optimization problem was formulated to handle only data streams as best effort traffic. Therefore, this method does not preserve a prespecified bit rate QoS constraint (R^G).

Another scheme, proposed in [7] and indicated hereafter as (PRA), considers power and rate adaptation one at a time. The authors presented two alternatives for *P-R* adaptation. In first mode, *Mode-A*, MS transmitting bit rate is left uncontrolled while power can be increased up to some value ($P^{\text{Limit-A}} < P^{\max}$) to preserve the E_b/N_0. In case of bad channel conditions, transmitting power is limited to $P^{\text{Limit-A}}$ while bit rate is reduced to meet the Θ requirement. On the other hand, *Mode-B* manipulates only bit rate. It completely suspends transmitting data when the channel gain is below a threshold. Otherwise, it reduces bit rate while having constant MS transmitting power at some level ($P^{\text{Limit-B}} < P^{\max}$). Evidently, PRA provides more flexibility to the resource scheduling problem over TPC by allowing bit rate manipulation. This flexibility can easily be noticed by visualizing the constrained search region in *P-R* space. As depicted in Fig. 3a, TPC set is restricted to simply horizontal lines search since R is not controlled. In the meantime, PRA has more permissible area by including vertical lines corresponding to R adaptation when P reaches the threshold level PLimit. As shown in Fig. 3b, PRA also embraces the space origin, which corresponds to transmission suspension of *Mode-B*. Clearly, *Mode-B* provides higher power gain but introduces time delay. Besides lack of commitment to the QoS guaranteed bit rate level R^G, the major drawback of PRA is its high dependence on the power level limit P^{Limit}, especially since the authors have not proposed a procedure for defining or optimizing it [7].

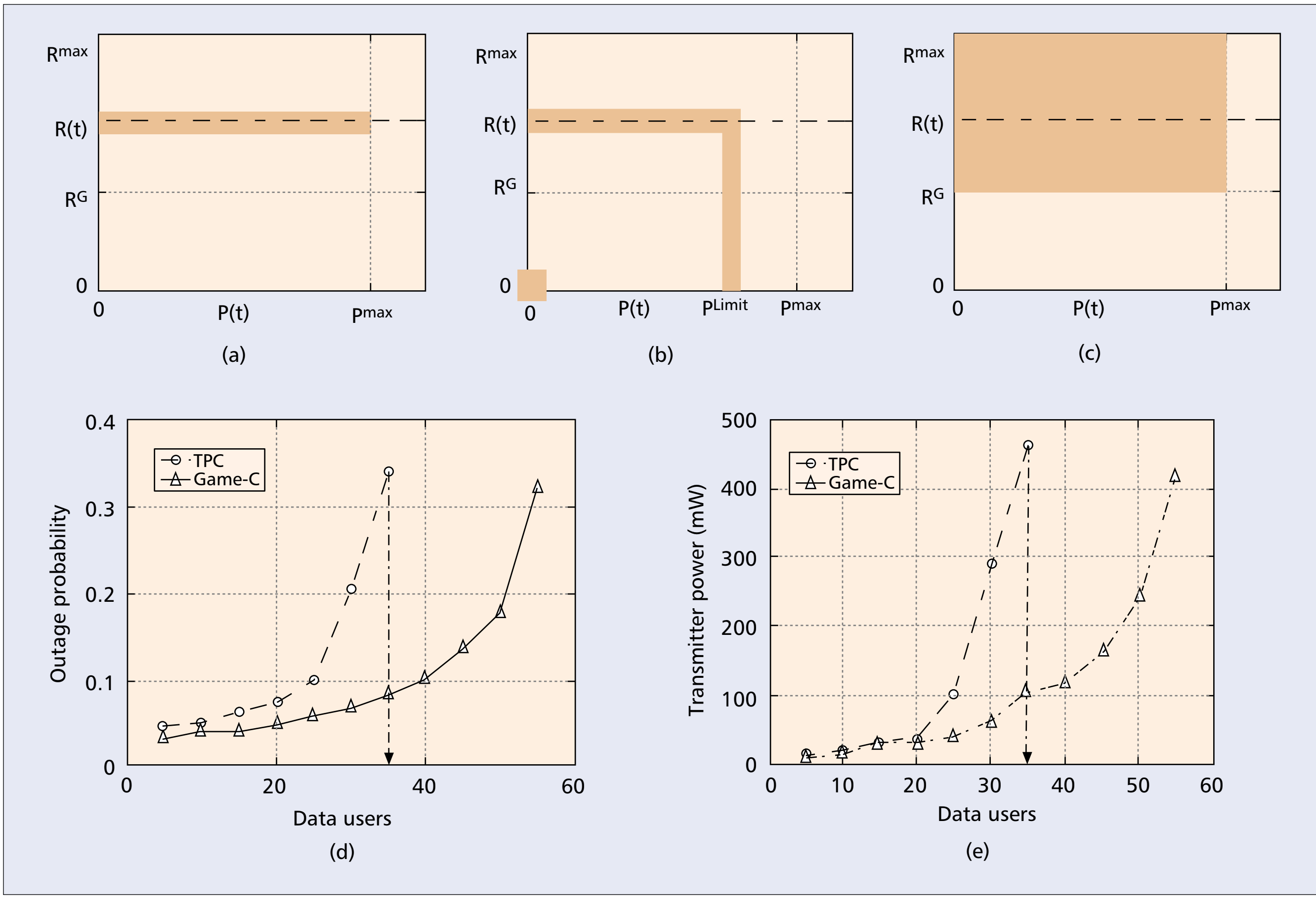
Another recent RRS combining power and rate control is the Genetic Algorithm for Mobiles Equilibrium (GAME) proposed in [8]. It relies mainly on a fast evolutionary computational model trying to find the optimum power P^* and optimum bit rate R^* for each MS. A solution is optimal in the sense that each MS gets merely enough power, P^*, to fulfill its Θ with the maximum possible rate, R^*, where $R^* \geq R^G$. First, the problem has been formulated by defining the *P-R* search space for the power and rate values of each MS. Then some restrictions have been set on the search space by excluding P values above maximum capability of the transmitter $P^{\max}$. Also, in order to preserve QoS, R values below the guaranteed bit rate R^G have been excluded. In other words, the search space has been trimmed to ensure QoS (R^G) and to be within the MS transmitting power capabilities. A fitness function has been defined to reflect the following objectives:

- Maximize the number of QoS (BER) satisfied users by having $E_b/N_0 > Q$
- Prefer solutions with lower transmitting power P
- Prefer solutions with higher transmitting bit rate R

Finally, a steady state genetic algorithm is used to browse the constrained *P-R* search space

[2] *The IS-95 standard suggests that the received signal strength should be measured. However, in practice usually the SIR or* E_b/N_0 *are used, since they have direct impact on the BER.*

[3] *In UMTS, the power control period is 0.625 ms.*



■ **Figure 3.** *(a, b, c)* P–R *space with shaded permissible search regions for TPC [5], PRA [7], and GAME [8] schemes respectively. (d, e) performance of GAME compared to standard TPC for packet data users with peak bit rate of 144 kb/s, guaranteed bit rate (RG) of 48 kb/s and required* E_b/N_0 *of 3 dB.*

using the proposed fitness function and extracting the fittest solution that maximizes the number of QoS (BER and R^G) satisfied users while using the lowest possible transmitting power.

Looking again at the *P-R* space, Fig. 3c reflects GAME flexibility by having a large selection area. The authors demonstrated the gain provided by a GAME-based scheme combining power and rate control, called GAME-C, over the standard TPC method through several experiments. As shown in Fig. 3d, GAME-C surpassed standalone TPC in user satisfaction by yielding lower outage probability for the same number of MSs. Moreover, GAME-C consistently took the QoS lead by supplying its users with more extra E_b/N_0 than the classic TPC scheme. Additionally, GAME-C was able to reduce MS transmitter power consumption as expected and depicted in Fig. 3e. The major reason for these savings is again the bit rate manipulation that gave the base station another degree of freedom in the restricted power allocation problem. Meanwhile, the granted bit rate was always above the promised one R^G, hence preserving QoS. In short, GAME manipulation of both transmitter power and rate increases the number of QoS satisfied users in a cell.

Radio Interface Protocol Architecture

The MS-RAN radio interface is layered into three protocol layers: physical (L1), data link (L2), and network (L3) layers. The physical layer offers information transfer services to higher layers, including:

- Forward error correction (FEC) encoding/decoding and interleaving/deinterleaving of transport channels
- Modulation and spreading/demodulation and despreading of physical channels
- Measurements and reporting to higher layers (e.g., BER, SIR, interference power, transmit power)
- Closed loop power control (CLPC)

As illustrated in Fig. 4a, in the control plane (C-plane), layer 2 is split into two sublayers: medium access control (MAC) and radio link control (RLC). Layer 3 is partitioned into sublayers where the lowest sublayer, radio resource control (RRC), interfaces with layer 2 and terminates in the RAN.

Upon RRC request, MAC executes radio resource reallocation and reports measurements such as traffic volume and quality indication to RRC. MAC is responsible also for priority handling between MSs by means of dynamic scheduling.

RLC provides error correction in protocol data units (PDUs) by retransmission (e.g., Selective Repeat, Go Back *N*, or Stop-and-Wait automatic repeat request, ARQ) and ciphering to prevent unauthorized acquisition of data. Besides, it is in charge of detection of duplicated

Since next generation mobile telecommunication systems are not devoted to either voice or best effort services, it is important to address Quality of Service provisioning, which is essential for the success of packet data services, especially in a bandwidth-constrained and error prone environment such as cellular networks.

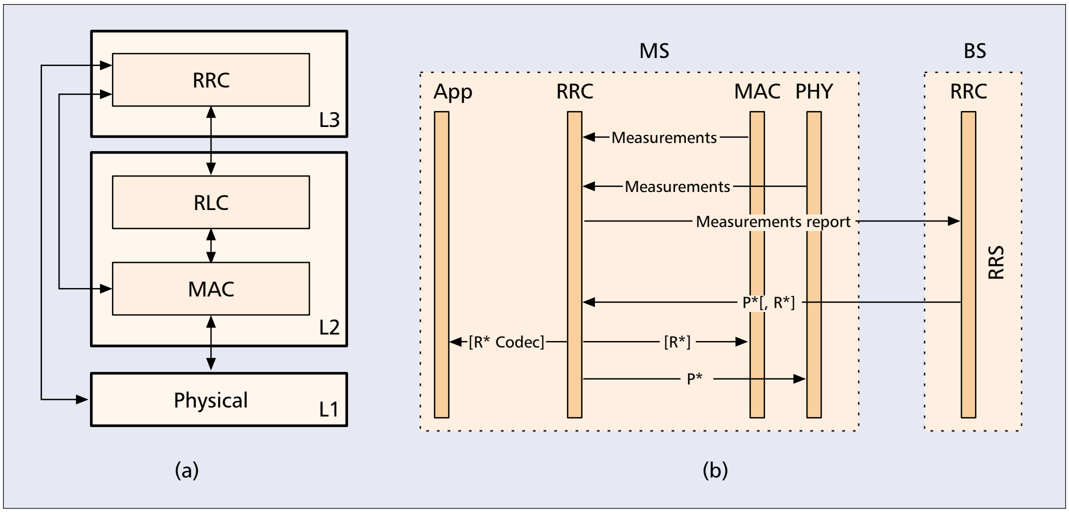


Figure 4. *a) Control plane layered radio interface protocol architecture; b) resource control signaling..*

received RLC PDUs to ensure that the resultant higher-layer PDUs are delivered only once to the upper layer and in the same order as submitted for transfer. RLC can also suspend/resume data transfer based on an RRC command.

The RRC layer handles the control plane signaling of layer 3 between MSs and RAN. It performs several functions:

- Broadcast of system information to all MSs
- Establishment, maintenance, and release of an RRC connection between MS and RAN including admission control
- Assignment of radio resources (e.g., codes, bandwidth) needed for the RRC connection while ensuring that the QoS requested for the radio bearers can be met
- Connection mobility functions such as handover based on measurements done by the MS
- MS measurement reporting (what to measure) and control of the reporting (when to measure and how to report)
- Outer loop power control by setting of the target Q of the CLPC

Radio Resource Scheduling Signaling

Naturally, RRC protocol implements most RRM techniques and specifically RRS functionalities at the BS since it controls and signals the allocation of radio resources to the MS. Figure 4b illustrates the signaling messages interchanged between the BS and MS on the C-plane to execute the RRS task.

Initially, the MS's L1 and MAC layers perform some measurements (e.g., transmitter power *P*, bit rate *R*, interference power) and report these values to the MS's RRC layer. Next, the MS composes a measurement report and forwards it to the BS RRC, which needs this information to determine which radio resources are available. Subsequently, an RRS algorithm residing in the BS, like GAME, determines the minimum power P^* and bit rate R^* necessary for each MS to satisfy its QoS requirements. Finally, the MS's RRC layer receives those values (P^*, R^*) and in turn passes them on to appropriate layers (L1 and MAC) to be in effect. Optionally, the MS's RRC can send a control message to the application coder to adjust the data encoding bit rate as well, provided that the coder has multirate capability. For example, for streaming video data, the coder can drop some intermediate frames or increase its quantization step when advised by the RRC to reduce outgoing bit rate.

QoS Architecture

Since next-generation mobile telecommunications systems are not devoted to either voice or best effort services, it is important to address QoS provisioning, which is essential for the success of packet data services, especially in a bandwidth-constrained and error-prone environment such as cellular networks.

Naturally, network services are considered end-to-end, that is, from terminal equipment (TE) to another TE. The user's TE, such as a personal digital assistant (PDA), notebook PC, or digital phone, is connected to the UMTS network through MS equipment such as a wireless modem. As illustrated in Fig. 5, the *local bearer service* handles QoS between TE and MS through a QoS-capable application programming interface (API). A *bearer service* defines all aspects between communicating endpoints to enable provision of a contracted QoS. These aspects include control signaling, user plane transport, and QoS management functionality. *UMTS bearer service*, offered by the UMTS operator, is in charge of QoS management inside a UMTS network, that is, from MS to CN. This is actually where the UMTS QoS model, described below, is in effect. Finally, QoS support outside the UMTS network is provided by the *external bearer service*, for example, IETF-defined integrated services (IntServ) and differentiated services (DiffServ), or simply best effort service.

UMTS QoS Model

Most of the services featured in UMTS can be divided into four main QoS classes primarily discriminated based on their ability to tolerate delay and BER. As defined in [1] and seen in Table 1, *conversational* and *streaming* classes preserve time relation between information entities of the stream and have a guaranteed bit rate R^G from the network. They are suitable to carry real-time traffic since they define an upper limit on transfer delay within their QoS profiles. The main difference between them is the maximum transfer delay value. Conversational traffic is subject to strict human perception in a VoIP talk or videoconferencing, while real-time streaming traffic has slightly flexible delay requirements.

On the other hand, *interactive* and *background* classes are mainly intended to represent conventional Internet applications (e.g., Web browsing, telnet, FTP, and email). They concentrate on preserving payload content by means of channel coding and retransmission, and are characterized as best effort traffic since neither RG nor maximum delay is specified. Interactive class traffic essentially follows a request-response pattern (e.g., interactive Web browsing or message chatting), while background class is chosen for background traffic (e.g., email or file downloading); the destination is not expecting data within a certain time. Interactive applications have higher priority than background ones in terms of resource assignment to ensure responsiveness.

The Outside-UMTS QoS Model

The UMTS QoS model, which includes QoS classes and attributes, is effective only within the UMTS bearer service shown in Fig. 5, and is designed to be independent of external QoS mechanisms. On the other hand, outside the UMTS network, say, Internet applications, use IntServ (signaled by Resource Reservation Protocol, RSVP) [9] or DiffServ (6-bit QoS attribute on each IP packet) [4] QoS models.

Practically, a TE application usually uses a suitable API for requesting QoS. As an example, the generic QoS (GQoS) Winsock 2 API, defined by Microsoft, provides a QoS interface that invokes RSVP to signal resource allocation to a network.

RSVP, an IETF layer 3 end-to-end signaling protocol standard, simply expresses application resource requirements to the network nodes on the path between a sender and its receiver(s). RSVP is receiver-driven, meaning that the receivers (not the senders) make the resource reservation. Specifically, a sender issues RSVP PATH messages to its receivers. Then the receivers reply by RSVP RESV messages along the reverse path to request for resource reservation. Although GQoS and RSVP are supposed to be protocol-independent, in reality they assume IP. As proposed in [10], the relevant GQoS parameters here are:

- *Level of service quality*: Guaranteed, controlled load, and best effort.

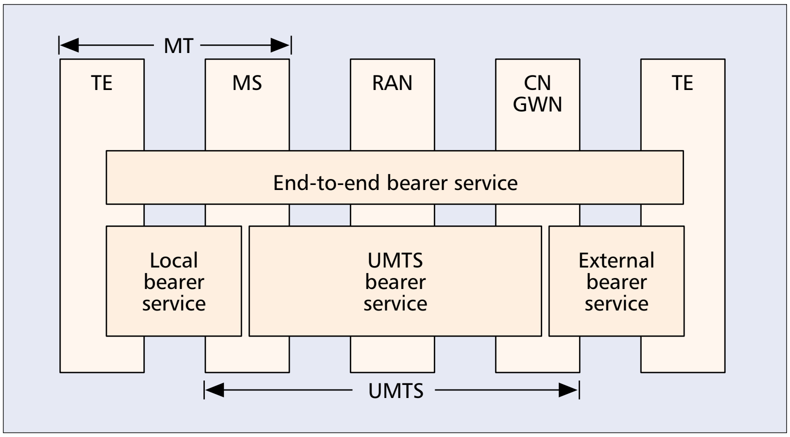


Figure 5. *End-to-end QoS architecture.*

- *Source traffic specifications*: The *token bucket model* parameters (token rate in bytes per second and token bucket size in bytes) used to specify the rate at which permission to send traffic (or credits) accrues. The *peak bandwidth* (bytes per second) is also specified to limit how fast packets may be sent back to back from the application.
- *Latency*: Upper limits on the amount of delay and delay variation (both in microseconds)

QoS Interworking and Parameter Mapping

Obviously, to have actual end-to-end QoS support, some mapping has to be done to ensure QoS interworking between existing defined schemes and the UMTS model. Fortunately, this mapping overhead is needed only on boundary elements (i.e., MS and GWN). Hence, the RAN and CN's other components do not have to understand external QoS mechanisms, and consequently do not have to be upgraded frequently.

Specifically, on the uplink (traffic sent by an MS toward UMTS), the MS should be able to represent the external QoS requirements in a form suitable to the UMTS QoS model, while GWN translates internal UMTS QoS parameters to the external network. MS and GWN mapping roles are reversed on the downlink (traffic sent to the MS).

As suggested in [10], GQoS *level of service quality* maps fairly well to the UMTS *QoS class* definition, and GQoS *peak bandwidth* maps to the UMTS *maximum bit rate* attribute. UMTS *guaranteed bit rate* integrated with traffic shaping at network boundaries can support the GQoS

		Conversational class	Streaming class	Interactive class	Background class
Features	Application	Voice over IP	Audio/video	Web browsing	Email
	Delay	Strict and low	Bounded	Tolerable	Unbounded
	BER	″ 10^{-3}	″ 10^{-5}	″ 10^{-8}	″ 10^{-8}
Defined attributes	Maximum bit rate	Yes	Yes	Yes	Yes
	Guaranteed bit rate (R^G)	Yes	Yes	No	No
	Maximum delay	Yes	Yes	No	No

Table 1. *UMTS QoS Classes: fundamental features and attributes [1].*

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token bucket model. GQoS *latency* maps to UMTS *maximum delay*. However, UMTS does not currently recognize delay variation at all due to rapid fluctuations in radio channel conditions.

Thus, the uplink flow starts when the TE application uses GQoS parameters along with a QoS signaling protocol, such as RSVP through the local bearer service. Next, the UMTS layer in the MS maps the specified parameter values in the signaling protocol to those understood within the UMTS bearer service. On the other end, GWN translates UMTS attributes, through the external bearer service, to GQoS native terminology. As a result, end-to-end QoS support between TE and external networks is maintained.

Sometimes QoS mapping is not needed between the RAN and the Internet. This has been investigated through the Broadband Radio Access Network for IP-Based Networks (BRAIN) project. The BRAIN architecture includes a RAN that is fully IP-based, thus eliminating the need for a mediator like CN in the UMTS to connect with the Internet. In this model, the RAN uses internally the same QoS techniques as the outside world. Hence, the QoS internal and external bearer services can be merged into one entity.

CONCLUSIONS

In this article we present an overview of radio resource management in 3G mobile telecommunication systems, especially the wideband CDMA mode of the UMTS. We review some key resource scheduling schemes covering power-only and combined power rate control methods. We also highlight the benefits provided by the combined algorithm in terms of lower outage probability and higher capacity while maintaining QoS. Moreover, we describe the signaling details to support the MS power and bit rate control over the UMTS radio interface protocol architecture. Finally, we elaborate on the end-to-end QoS design model including interworking and parameter mapping between MS application, UMTS, and external networks. We believe that all those pieces are necessary for the success of broadband wireless mobile access to established wireline networks like the Internet.

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BIOGRAPHIES

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