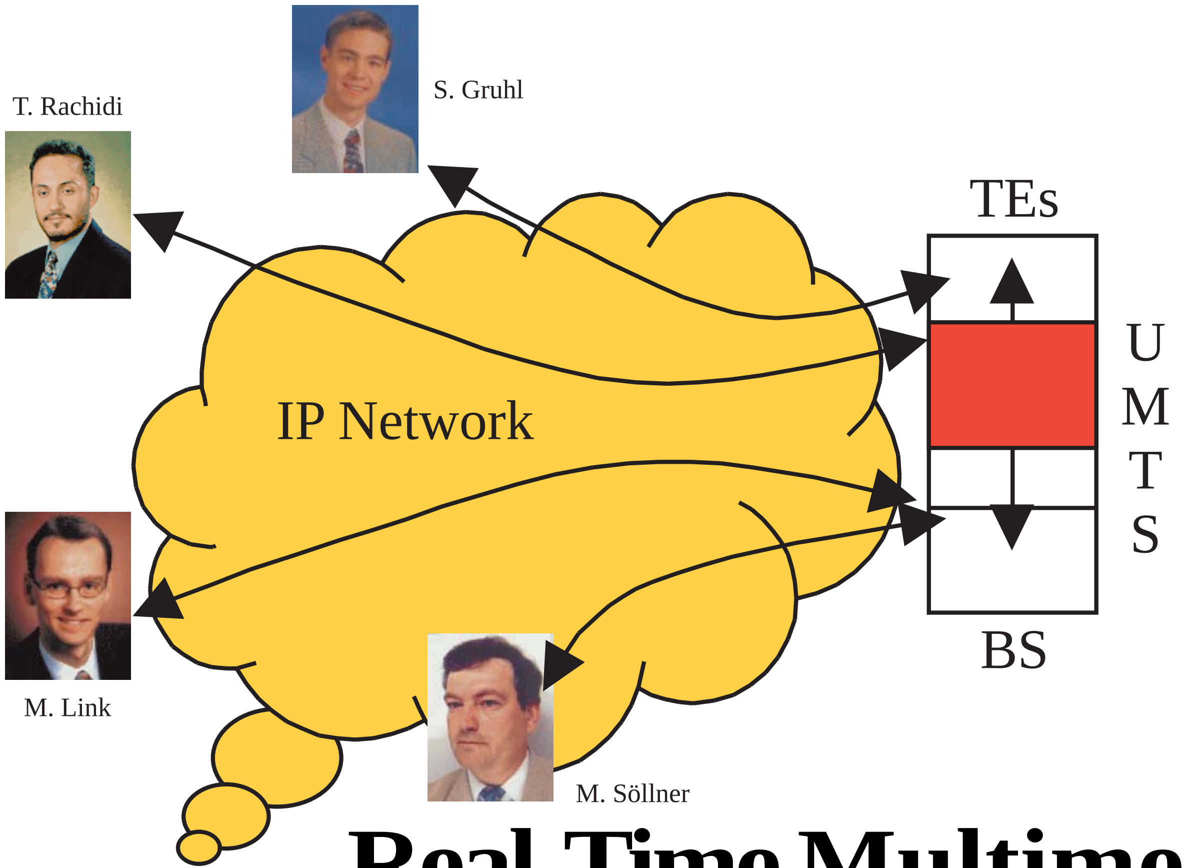

Real-Time Multimedia Applications over 3rd Generation Wireless Networks

by S. Gruhl, T. Rachidi, A. Echihabi, M. Link, and M. Söllner

Abstract—*It is expected that in 3rd generation wireless networks in general, and in the Universal Mobile Telecommunications System (UMTS) in particular, complex control and transport mechanisms will influence the data communication. The wireless link quality varies for a given application's data flow, while the application itself adapts to the system and thus influences the control mechanisms. In this paper, we study the effects of such a dynamic system on the user-perceived Quality of Service (QoS) for real-time multimedia applications over UMTS, by means of simulating the wireless link. The simulator comprises essential layer 2 and 3 protocol functionality of the UMTS Terrestrial Radio Access Network (UTRAN) for terminal equipment (TE) and base station (BS). It is used to demonstrate in real-time the effects of 3rd generation wireless network access on IP-based multimedia applications.*

UMTS

The Universal Mobile Telecommunication System (UMTS) will extend the services provided by current second-generation systems (GSM, PHS, IS-95, etc.) from simple circuit-switched voice telephony to complex data services ranging from e-mail and web-browsing to voice over packets, media on demand, and video conferencing [1]. Users will be able to interact totally with their Wireless Information Devices to retrieve, store, and process data anywhere, anytime while being on the move. To this end, UMTS will support packet-switched data services for up to:

- 144 kbps for high speed mobile users
- 384 kbps for low speed mobile users
- 2 Mbps for portable/fixed users

Packet switched services use the system capacity more efficiently, and allow for user idle time and volume charging policy. The statistical gain of packet switching results from increased link utilization due to non-continuous bandwidth requirements from applications. Such a mechanism is well investigated in wireline networks where medium capacity does not vary. Major differences exist, however, between wireline and wireless networks, because the radio link constitutes a massive bottleneck:

- Power limitation, interference and altering radio link conditions due to mobile terminal position cause the link capacity to change rapidly.
- Handoff calls lead to additional and unpredictable load in a cell.
- High bit error rates (BER) are encountered in wireless communications. Forward error correction (FEC) becomes important and retransmissions are more frequent than in fixed networks. Consequently, data rates are affected.

Once deployed, UMTS is expected to interface seamlessly with the wide variety of interactive, media-on-demand and multimedia IP-based applications developed originally for the Internet, like the ITU standard H.323 "Packet Based Multimedia Communications Systems" [12] as depicted in Fig. 1.

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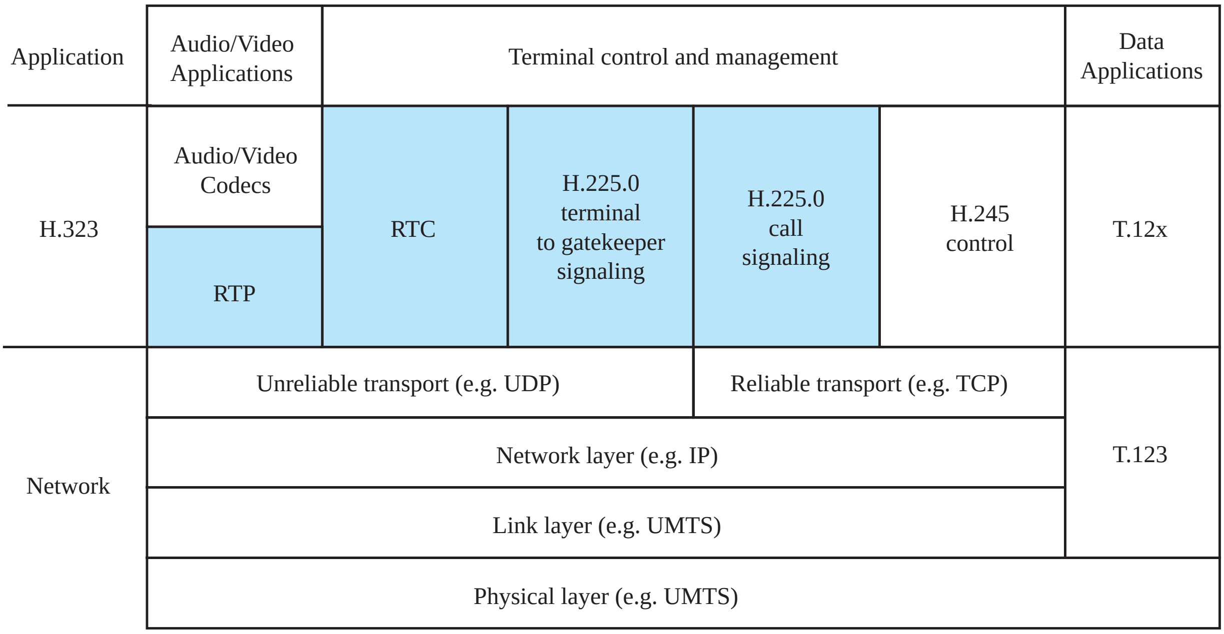

Figure 1. Principal scheme of an Internet multimedia protocol stack, as used in the ITU standard H.323 "Packet Based Multimedia Communications Systems" [12–17].

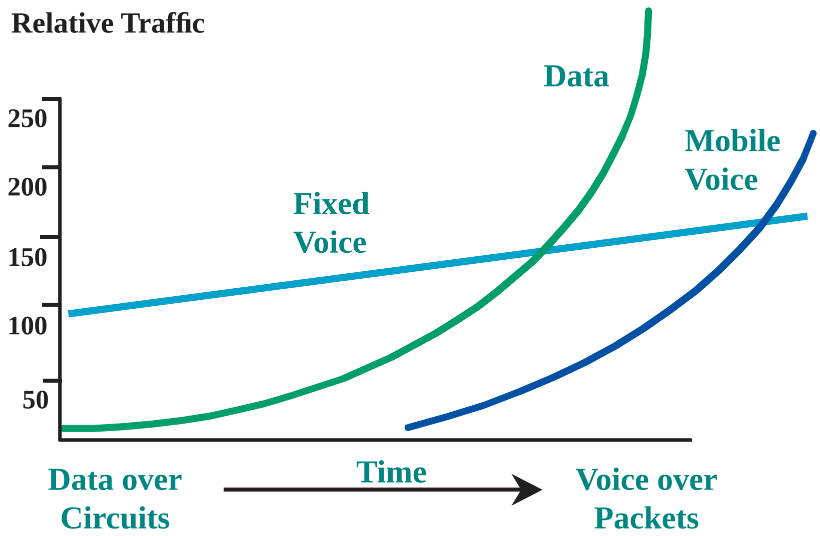

Figure 2. Estimated relative traffic mix.

Wireless Networks … continued from Page 17

The volume of data generated by these applications is expected to grow over-proportional in terms of bandwidth consumption as depicted in Fig. 2.

However, due to the above mentioned intrinsic characteristics of the wireless link, it is a challenge to deliver circuit-switched-like Quality of Service (QoS), such as bounded delay and jitter, which are essential for multimedia and interactive applications. The QoS needed for the broad variety of data applications that will be available over UMTS can be specified in terms of several QoS parameters and classes. Table 1 describes these classes.

It is critical to investigate UMTS system performance from the perspectives of system efficiency and QoS contract fulfillment. To this end, it is essential to model the wireless subsystem in terms of its capacity and transmission technique. Wideband Code Division Multiple Access (WCDMA) as the transmission technique for UMTS has the following characteristics:

- CDMA is a spread spectrum technique developed for military anti-jam applications.
- Wide bandwidth supports high bit

Table 1. The four UMTS traffic classes defined by ITU-R [2–3].

Class Number	Traffic Class	Class Description	Example	Relevant QoS Requirements
1	Conversational	– Preserves time relation between entities making up the stream – Conversational pattern based on human perception – Real-time	– Voice over IP – Video conferencing	– Low jitter – Low delay
2	Streaming	– Preserves time relation between entities making up the stream – Real-time	– Real-time video	– Low jitter
3	Interactive	– Bounded response time – Preserves the payload content	– Web browsing – Database retrieval	– Round trip delay time – Low BER
4	Background	– Preserves the payload content	– Email – File transfer	– Low BER

rates and helps to combat fading in multi-path radio channels.

- Many users share the same radio carrier.
- Each user is assigned a unique random code different from and approximately orthogonal to other codes.
- Quality degrades as the number of users on a channel/carrier increases (interference limited system).

WCDMA technology results in soft-capacity behavior. In classical schemes employing a combination of Time Division Multiple Access and Frequency Division Multiple Access (TDMA/FDMA schemes), the total capacity is static (see Fig. 3), and the

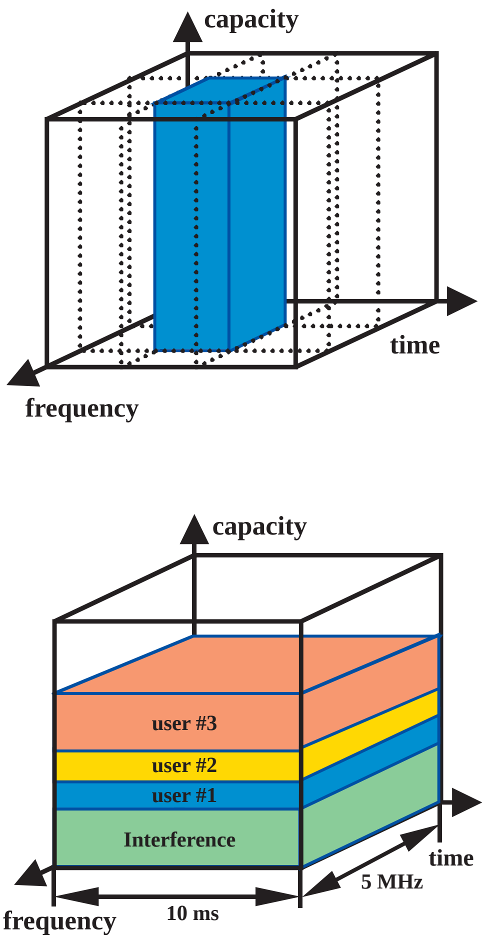

Figure 3. Capacity in FDMA & TDMA (top) versus W-CDMA (bottom).

Multiple-Access	DS-CDMA (TD-CDMA)
Duplex scheme	FDD (TDD)
Chip rate	3.84 MChip/s
Carrier spacing	Flexible in the range 4.6–5.0 MHz (200 kHz carrier raster)
Frequency bands	1920–1980 / 2110–2170 paired (1900–1920 and 2010–2025 unpaired)
Frame length	10 ms; 15 time slots
Inter-BS synchronization	No accurate synchronization needed (synchronization needed)
Multi-rate/ Variable-rate scheme	Variable-spreading factor + Multi-code Spreading factor: 4–256 (1–16)
Channel coding scheme	Convolutional coding, Turbo coding, rate 1/2–1/3
Packet	Dual mode on common and dedicated access channels

Table 2. UMTS Key Parameters (source 3GPP).

QoS of an individual link is hardly correlated to other carriers in the cell. In CDMA the whole capacity is limited by the relative signal to noise ratio of the individual links (interference limited system). Adaptive techniques, e.g. power control and admission control are key in providing QoS for each individual service.

This naturally imposes the need to model physical layer behavior with regard to individual services. Table 2 lists typical UMTS key parameters to be taken into consideration in the model.

To address the specific requirements of services while achieving a high spectral efficiency there are dif-

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Real-Time Multimedia Applications over 3rd Generation Wireless Networks

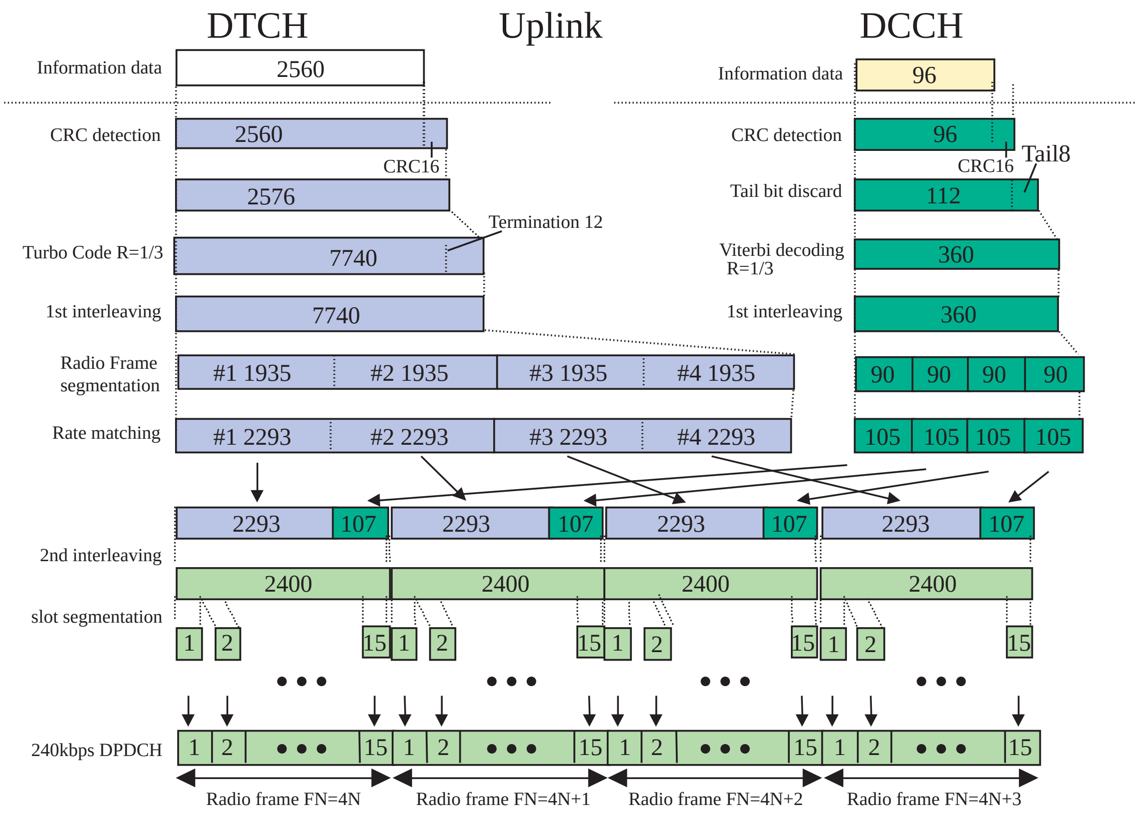

Figure 4. Example of how two different services data/voice receive different physical layer processing.

Wireless Networks … continued from Page 19

ferent coding schemes applied to user data as shown in Fig. 4.

For our investigations we concentrate on the UMTS terrestrial radio access network (UTRAN), see Fig. 5. Thus we explicitly model the UTRAN part with its protocols, see Fig. 6. As we work with the standard IP interface we are able to include possible transit networks in our investigation, by using existing networks as access networks to our demonstrator. These are not modeled but actually coupled with the real-time testbed. The detailed description of the testbed is out of the scope of this paper, and is presented in [11].

UMTS Protocol Stack

The UMTS Radio Interface architecture is layered into a physical layer, a data link layer and a network layer (IP in this work). The data link layer is divided into a Radio Link Control (RLC) sublayer and a Medium Access Control (MAC) sublayer. The Logical Link Control (LLC) sublayer present in many early UMTS proposals is not considered. It is expected that this sublayer will be reduced to a Null-sublayer to minimize protocol overhead and/or will be merged with the PDCP Layer, which is doing header compression. Figure 5 shows protocol termination for UMTS dedicated channel (DCH).

The following is a summary of the main services and functions of Layer 1 and Layer 2 [4–6] that have been considered in the system:

Physical Layer Services:
- Information transfer services to higher layers though Transport Channels

Physical Layer Functions:
- Error Detection and FEC on transport channels
- Multiplexing and de-multiplexing of transport-channels/coded-transport composite channels
- Synchronization
- Measurements and indication to higher layers

MAC Sublayer Services:
- Data transfer
- Radio resources and MAC parameters reallocation
- Measurements reporting

MAC Sublayer Functions:
- MAC-scheduling and selection of an appropriate transport format.
- Multiplexing/de-multiplexing of higher layer PDUs into/from transport frames delivered to/from the physical layer
- Traffic volume monitoring and reporting to RRM

RLC Sublayer Services:
- Transparent data transfer
- Acknowledged/unacknowledged data transfer
- QoS settings

RLC Sublayer Functions:
- User data transfer
- Segmentation and reassembly
- Padding
- In-sequence/out-of-sequence delivery of higher layer PDU
- ARQ—backward error correction and flow control

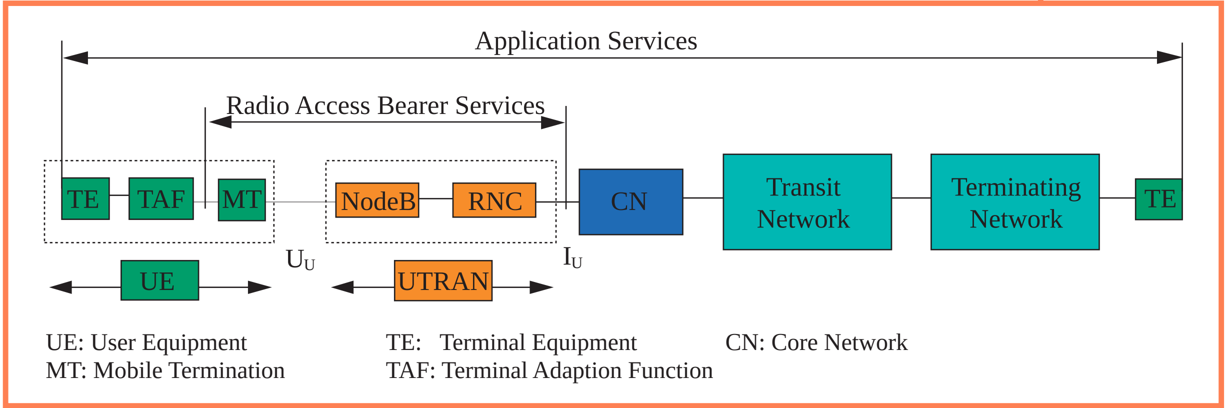

Figure 5. Logical network architecture.

UMTS provides besides other services a reliable RLC mode to be used by non-real-time or interactive applications, based on FEC using various channel coding schemes, radio blocks.

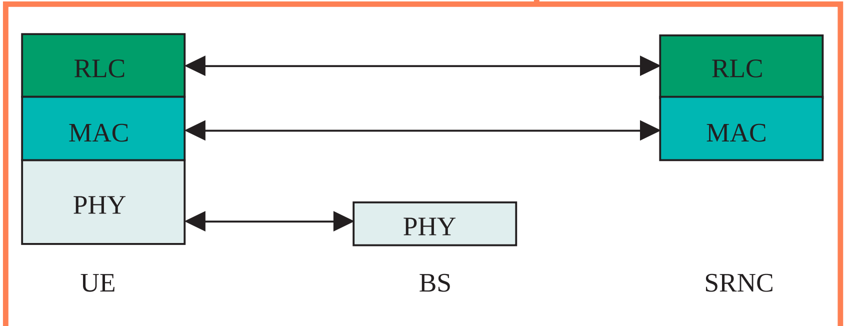

Figure 6. Our model for the protocol in the UTRAN.

As can be seen, most UMTS layer 1 and 2 connectionless services and functions have been considered. Functions for connection establishment on the other hand have been ignored. This assumption is valid for the scope of our

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Stefan Gruhl

Tajje-eddine Rachidi

Wireless Networks … continued from Page 21

project, as we are only interested in the dynamic system behavior of an established link.

Control Loops

We have identified a set of challenging questions pertaining to wireless network performance in general and to UMTS in particular, which we address through the real-time testbed. These questions are:

- How successful does a certain QoS enabling technology such as UMTS perform with a given application?
- What will user-perceived QoS be like over a future wireless link?
- What are the traffic arrival characteristics of a certain type of future service, e.g., a video real-time session?

To illustrate what we consider as protocol performance in this context, we reference some studies of dynamic TCP/IP behavior and its performance over erroneous/slow links. TCP has TCP-flow-control as an adaptation mechanism for network congestion situations. It acts on packet loss and adapts its sending rate. For lossy links, as experienced in wireless transmission, this mechanism can show unexpected behavior and perform poorly [7–10]. We view these investigations as one example of undesired control loop interaction in a multi-protocol-layer wireless environment. We are interested in control loops in general, as e.g. also found in bandwidth adaptation methods in a scalable video codec. Our approach also allows us to address TCP performance investigations, but is not restricted to this.

In the worst case we expect to find oscillation effects that arise due to the various dynamic controls that are applied simultaneously by higher and lower level protocols. These mechanisms are found in lower layers to overcome lossy and variable capacity links, and in higher layers specifically to adapt to changing network capacity due to congestion situations.

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Stefan Gruhl received the masters in computer science in 1998 from the Friedrich Alexander University of Erlangen-Nürnberg. He is currently participating in a joint research program between the Department of Computer Architecture and Performance Evaluation Institute of the Friedrich Alexander University and the Global Wireless Systems Research Group at Bell Labs, Lucent Technologies. His research interest is quality of service for packetized data for cellular wireless systems, particularly GPRS and UMTS.

Tajje-eddine Rachidi is associate professor of computer science at Alakhawayn University in Ifrane, Morocco. He received his first degree from the Ecole Nationale Supérieure d'Informatique et de Mathématiques Appliquées de Grenoble (ENSIMAG), France, in 1989 specializing in real-time systems, and earned the Ph.D. degree from Essex University, U.K., in 1995. His research interests include 3rd generation wireless networks and computer vision. He consults for various local companies in Internet/Intranet and database technologies. He is co-founder of CoreSoft: a software company developing intelligent components.

Michael Link received the diploma in electrical engineering from University Erlangen-Nürnberg, Germany, in 1994. From 1994 to 1999 he worked as research assistant at the Telecommunications Laboratory, University Erlangen-Nürnberg. In September 1999 he joined the Network Technology Group at Lucent Technologies, Nürnberg. His research interests are in channel coding for the transmission of multimedia data in packet switched networks, channel modeling, and wireless multimedia protocols, especially for UMTS.

Michael Söllner received the diploma in mathematics in 1979 and the Ph.D degree from the University of Bochum, Germany in 1982. Since 1984 he has held various positions in communication technology in industry, entering mobile communications with Philips in 1990, where he focused on traffic and protocol engineering for mobile radio systems, being also involved in ETSI GSM standardization. He is now responsible for a Bell Labs wireless research group with Lucent Technologies in Nürnberg, Germany. His current main research interests are radio system aspects of 3rd generation wireless services.

In particular, we have looked at adaptive video codecs on top of the Real-time Transport Protocol (RTP), which run their independent control loops. These interact with other adaptive control loops from a wireless network, in a way comparable to the TCP phenomenon. There can be many control loops, e.g. power control to adjust the transmit power on the radio link, which affects error rates or control performed by the Radio Resource Manager (RRM) on Resource Allocation. These interactions, if not investigated and understood, can hinder the many benefits expected from future wireless communications networks.

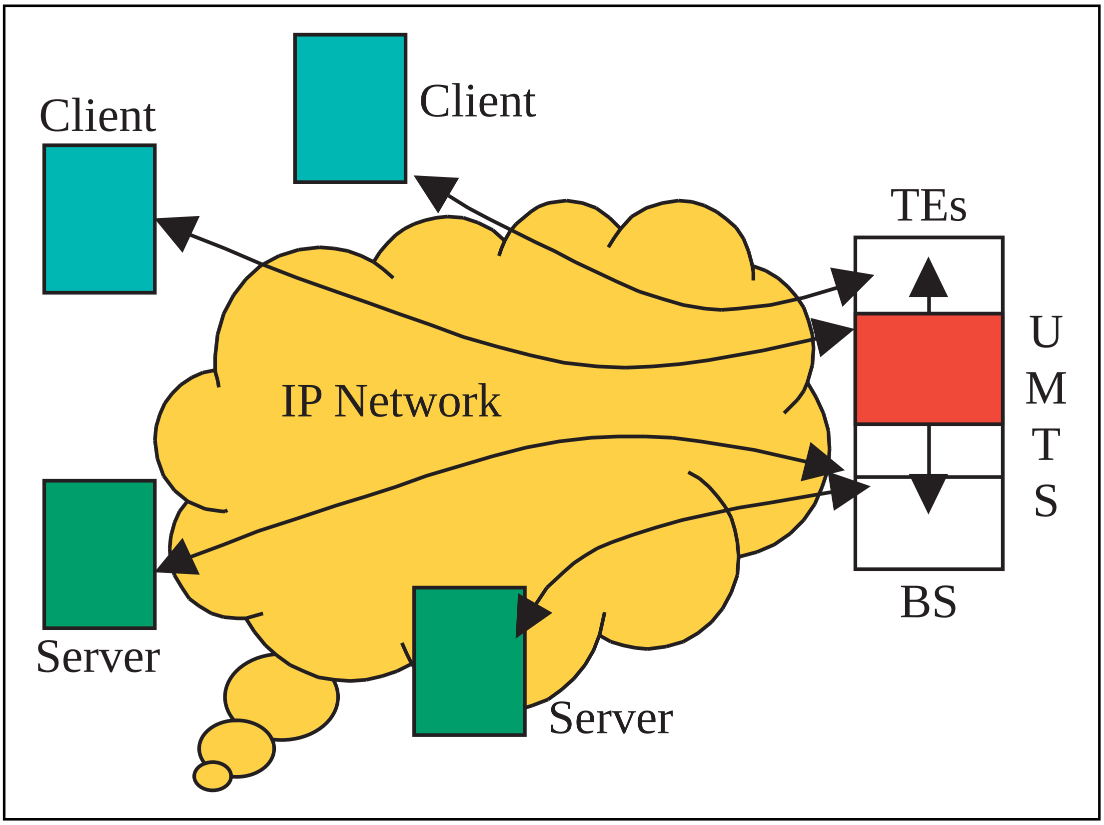

Figure 7: Standard Client/Server IP applications run through the UMTS protocol stack (red).

Control Loops and QoS

Future packet switched networks will have to incorporate QoS enabling techniques to address specific application requirements. Imagine a guaranteed bandwidth link with a low error rate. Then you will find most of the TCP-flow control problems to be softened to a large degree. How does this work for other control loops? What are the QoS requirements that should be satisfied? How sensitive is this approach to wireless intrinsic link variation or actions like a handover to another BS?

With our approach we will be able to investigate the benefits of a QoS enabled system. As we offer a real-time real-application interface, these effects will be accessible for measurements. Furthermore, our system will allow demonstrating user perceived Quality of Service over a simulated UMTS network.

For many higher layer protocols and applications we lack the full description of internal control mechanisms, but often we find them as implemented applications. Most applications today communicate via IP traffic. We use this transparent standard IP-interface to transport application data seamlessly in and out of our real time simulation of a third generation network. Thus standard IP-based applications such as FTP, web browsing, video applications or Microsoft Netmeeting™ can be simulated concurrently over the simulated UMTS protocol stack and radio link (see Fig. 7). Our approach links existing applications during run-time with our prototype implementation of a UMTS

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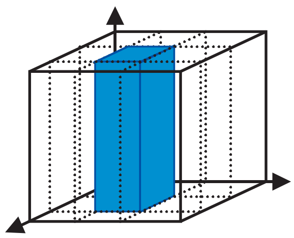
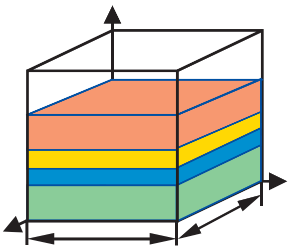
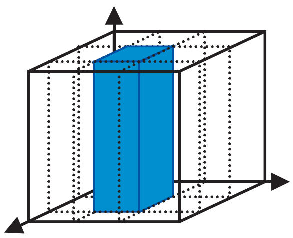
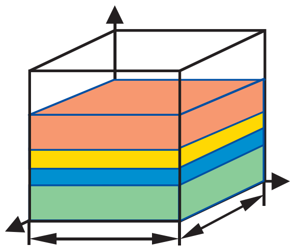
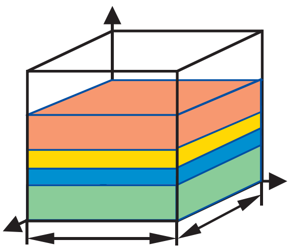
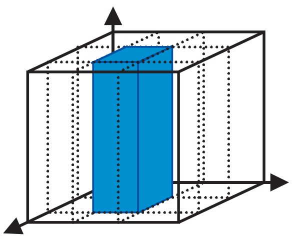
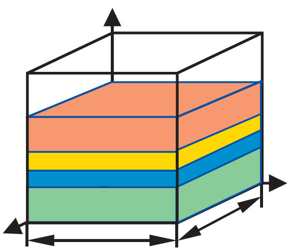
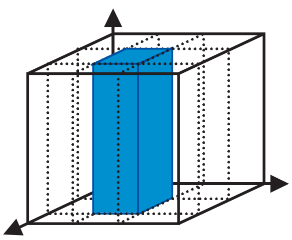
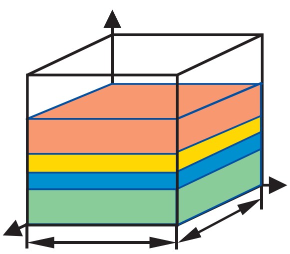

Wireless Networks … continued from Page 23

protocol stack and with our PHY-layer simulator. Thus we can run our investigations without the need to model the application/higher layer part. By using the real application we do not need any traffic models assumptions, but use the real dynamic behavior—including user interaction—for our investigations.

Although there are QoS protocols for IP networks, such as the Resource Reservation Protocol (RSVP), these QoS negotiation protocols are rarely found in present applications. Therefore we have developed the concept of individually assigned QoS for individual flows. A flow is distinguished by its sending/receiving port number. This allows us to do investigations with a simulation of QoS enabled networks. The QoS requirements are for example for reliability, delay, jitter, and guaranteed bandwidth.

The main advantage of an on-line simulation system is the possibility to demonstrate the user-perceived quality of service. We can investigate the effects of the various control mechanisms, altered radio conditions or a congestion situation in a cell.

Conclusions

Quality of Service will be highly important to enable new real-time multimedia services for 3rd generation mobile networks. Therefore it is important to investigate and understand the influence of all protocol layers from the physical link to the application. Therefore we have designed a real-time testbed for performance assessment of IP-based packet switched and emulated circuit switched multimedia applications over UMTS. Our proposed architecture allows concurrent access for an arbitrary number of application-flows, while we provide a QoS understanding on a per-flow basis even for non-QoS aware applications. This specifically allows investigating user-perceived QoS of today's multimedia applications over a simulated wireless link.

Starting from existing investigations on the performance of TCP over a wireless link we broaden the field of investigations to other higher layer control loops such as those found in adaptive RTP-based video applications, and combine them with simulated third generation wireless specific RRM functionality and wireless lower layer protocols.

Acknowledgements

We would like to thank Dr. Urs Bernhard and Dr. Jens Mueckenheim for their kind support on all areas of UMTS.

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