Investigation of Radio Resource Scheduling in WLANs Coupled with 3G Cellular Network

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ABSTRACT

In this article, based on the overview of network coupling structure between radio access technologies, the concept of joint radio resource management built onto the reference structure is briefly introduced. In order to optimize usage of radio resource and jointly designed from the user perspective, a joint scheduling mechanism allowing traffic to be split over a tightly coupled radio network supported by an adaptive radio multihoming approach is deliberately discussed in this article. With respect to the time-division access scheme in HIPERLAN/2, which is selected as one example of WLAN, algorithms and performance of traffic scheduling in such a radio access technology are given. The required synchronization scheme supporting traffic splitting is also introduced at the end.

INTRODUCTION

Future telecommunication systems are supposed to provide high-bit-rate services that enable high-quality multimedia services. They will consist of a number of coexisting subnetworks with different radio access technologies (RATs). The interworking between radio subnetworks, especially tight cooperation between them, is of great interest in operating RATs for system capacity optimization. The sharing of the radio resource, with the support of policy agreements among RATs in terms of low-coverage wireless local area networking (WLAN) becoming a complementary RAT to the cellular network is being investigated in order to give significant trunking gain and capacity enhancement in operating radio networks.

It is not easy for a single-layered third-generation (3G) system such as the Universal Mobile Telecommunications System (UMTS) to provide the required QoS and all the required bandwidth for high data rates with good connectivity. A WLAN system (e.g., HIPERLAN/2) cannot provide high connectivity due to its presence only in hot spots. Hence, proper interworking between radio subnetworks must be discussed.

However, the ongoing research activity in software-defined radio (SDR) introduces flexible terminal reconfiguration by replacing radios completely implemented in hardware by ones configurable or even programmable in software to a large extent [1]. These concepts include reconfiguration of not only the transceiver but also applications/services, as well as network-based reconfiguration support provided by a dedicated network infrastructure. The reconfiguration technologies target adaptation of the radio interface to varying RATs, provision of possible applications and services, update of software, and enabling full exploitation of flexible resources and services of heterogeneous networks. A certain depth of reconfiguration could update terminals with radio link layer functioning embedded according to network architecture in order to enable cooperation among multiple RATs.

In this article a proposal for Joint Radio Resource Management (JRRM) over heterogeneous networks is given. The requirement of using more than one system for the traffic scheduling demands and hence upgrading the concept of JRRM is studied. The article is organized as follows. We introduce the two subsystems needed in order to allow system capacity enhancement thanks to JRRM functions. We introduce two different types of traffic, HTTP traffic and a simplified scalable video traffic model. Joint scheduling of traffic through multiple systems will also be introduced. We then conclude the article.

SYSTEM MODELS

ARCHITECTURE OF COEXISTING NETWORKS

People have started thinking of possible convergence of available air interfaces in order to optimally utilize radio resources in future networks; for examples, there are common radio resource management proposals in the 3G Partnership Project (3GPP), coupling scenarios in the European Telecommunications Standards Institute (ETSI), and SDR technologies [1]. Service continuity and seamless access are required by some service types for some interworking between 3G
systems and WLAN, which are classified by 3GPP in [2], where scenario 5 requires fast interworking between WLAN and cellular networks.

Wideband code-division multiple access (WCDMA) technology has been adopted as the best candidate for a 3G radio system. Many countries have already started deploying and operating UMTS. The advantage of UMTS is higher user bit rates than 2G systems: a data channel up to 2 Mb/s can be established in a single radio cell [3]. UMTS has high connectivity; statistically speaking, at a low data rate the connectivity of UMTS is above 90 percent.

The complexity of network deployment and reconfiguration requires methods guaranteeing remote RNC even if they are not associated. The central processing units, such as mobile terminal (MT) can be processed by a transceiver station (BTS) hotels, will be connected to the terminal in order to allow the terminal to maintain simultaneous links with the radio network. The adaptive radio multihoming concept provides multiple radio access for a single terminal in order to allow the terminal to maintain simultaneous links with the radio network. Besides the capacity gain from a network operation point of view, mentioned earlier, the advantage of having concurrent parallel streams is manifold. If one bearer service has high availability in the network (low-data-rate bearer services offer high coverage, e.g., a 16 kb/s service is available in 99 percent of the cases), this link would be used to transfer important information to the terminal. On the other hand, a low-data-rate service cannot fulfill the requirements of multimedia traffic with high-data-rate demands. If traffic is intelligently split into rudimentary and optional information streams, higher QoS for the user is provided. Whenever possible the user combines both streams to yield higher QoS; due to the higher availability of a lower-data-rate service in UMTS, a minimal QoS can be fulfilled to the user, as shown in Fig. 1.

As radio technology evolves, more advanced radio resource management features should be added to a radio network supported by reconfigurable technologies. Possible future network architecture equipped with distributed radio controlling mechanisms over multiple air interfaces could emerge thanks to the reconfigurable technologies [1]. For reliability and efficiency reasons, the central processing units, such as radio network control (RNC) centers and base transceiver station (BTS) hotels, will be connected via optical fiber, over which the computational power is shared. For instance, calls from a mobile terminal (MT) can be processed by a remote RNC even if they are not associated. The complexity of network deployment and reconfiguration requires methods guaranteeing the reliability of network management.

**OVERVIEW OF RADIO RESOURCE MANAGEMENT**

Conventional radio resource management (RRM) aims to offer networks controlled mechanisms that support intelligent admission of calls and sessions, and distribution of traffic, power, and variances in them, thereby targeting optimized usage of radio resources and maximized system capacity.

The JRRM aims to support intelligent interworking between different access technologies mainly controlled by the central controller, effectively manipulating coupled subnetworks, and overall system capacity gain thanks to interworking between subnetworks. Joint radio resource scheduling and joint admission control are required to optimize spectral efficiency for coupled systems; handle various bearer types (e.g., voice, video) and different QoS constraints of users and services; and schedule traffic adaptively for mixed traffic types. With JRRM deployed in coupled RATs (Fig. 1) in terms of alternatively admitting incoming calls or offering access to substrings of incoming calls, significant capacity gain can be obtained [4]. Optimal QoS can be reached thanks to traffic splitting supported by adaptive radio multihoming, which provides multiple radio access for a single terminal in order to allow the terminal to maintain simultaneous links over RATs.

**THE ADAPTIVE RADIO MULTIHOMING CONCEPT**

An extension of the conventional multihoming concept is to run simultaneous connections on the radio frame level, which we call w.r.t. reconfigurable terminals, is the adaptive radio multihoming approach.

The adaptive radio multihoming concept provides multiple radio access for a single terminal in order to allow the terminal to maintain simultaneous links with the radio network. Besides the capacity gain from a network operation point of view, mentioned earlier, the advantage of having concurrent parallel streams is manifold. If one bearer service has high availability in the network (low-data-rate bearer services offer high coverage, e.g., a 16 kb/s service is available in 99 percent of the cases), this link would be used to transfer important information to the terminal. On the other hand, a low-data-rate service cannot fulfill the requirements of multimedia traffic with high-data-rate demands. If traffic is intelligently split into rudimentary and optional information streams, higher QoS for the user is provided. Whenever possible the user combines both streams to yield higher QoS; due to the higher availability of a lower-data-rate service in UMTS, a minimal QoS can be fulfilled to the user, as shown in Fig. 1.

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**TRAFFIC SPLITTING OVER HETEROGENEOUS NETWORK**

As investigated, Joint Session Admission Control (JOSAC) takes the neighboring RAT system load into account. The traffic stream is routed through the cooperating systems according to the restrictions and advantages of each system. From the service point of view, different levels of service calibration can be identified to meet the user’s satisfaction; for example, for video transmission, the video traffic can be split into base and enhancement layers, where the base layer consists of the most important lower-frequency information. The user will not be satisfied receiving only the enhancement layer information. On the other hand, from the user mobility point of view, the connection of UMTS is not restricted by the location of a user, but the WLAN is.

If the MT has simultaneous connections supported by reconfigurable terminals that should be equivalent to 3GPP terminal type three and type four [5], the data flows and control commands can be routed via different air interfaces, which have different delay characteristics (average delay and delay variance, i.e., jitter).

Generally, three key points are covered by radio multihoming:
• Traffic prioritization and splitting: As Figs. 1 and 2 depict, the incoming traffic is split over two (more) substreams. The important information goes through a reliable RAT, the rest through other RATs.
• Synchronization: Packets belonging to a substream are multiplexed back to original traffic stream in the receiver based on proposed synchronization schemes (in the MT and RNC).
• Buffer management: Jitter and average delay parameter are controlled by buffer size and synchronization approaches. The static terminal and user profiles stored on the network side will be retrieved by the RNC to determine the calculation power and buffer size of the terminal, and to evaluate the user preference and cost. The synchronization methods are used mainly to compensate average delay, whereas buffers are used to compensate for jitter.

As Fig. 2 shows, suppose a terminal is demanding a scalable video service from a remote server through tightly coupled subnetworks being controlled by one RNC. In order to establish simultaneous sub-streams belonging to the same video context, the following procedure should be fulfilled:

Step 1: Signaling and initialization. The RNC receives an application from an MT with multiple radio accesses/addresses. After estimating the available radio resource in a controlled subnetwork, the RNC will apply to the remote server for traffic splitting indicating the average rate in each sublink.

Step 2: Traffic is split according to the RNC’s application (step 1) and sent to the RNC. Substreams are labeled differently.

Step 3: The RNC receives split traffic with labeled packets to further map to tightly coupled subnetworks (e.g., traffic with label Vi to WLAN, label Ai to UMTS, i informs the RNC of the timing relationship between substreams). The possible services to be applied are:
• Video and audio substreams
• HTTP with separation of main objects and inline objects
• Scalable video traffic (base and enhancement layers)
• Real-time traffic and its control signaling

Figure 1. Functional architecture of joint radio resource management and delay factors.
Step 4: The synchronization mechanism in the RNC remedies delays generated by radio subnetworks due to:

- Different TTI values for bearer services
- Automatic repeat request (ARQ) actions due to different connection qualities
- Different processing power of different BTSs (especially in the shared processing component case, which is valid for the functional partitioning scheme mentioned earlier)

**TRAFFIC MODELING**

Mobile users expect services that will not only depend on a traditional single traffic type but multimedia traffic types. In order to test the goodness of RRM algorithms, HTTP traffic representing data service and scalable video traffic types is investigated.

**DATA SERVICES AND HTTP TRAFFIC**

Data services refer to applications such as the Web, email, and ftp. As HTTP is the dominant data service, it is taken as the data traffic model in this article.

A Web page typically consists of a hypertext document. The hypertext document is HTML coded text with links to other objects that make up the whole page (pictures, music, etc.). We consider a widely accepted on-off model composed of main objects and inline objects [6]. An object is an entity stored on a server as a file. The file containing the HTML document is the main object, and the objects linked from the HTML document are the inline objects. We assume that a session starts to be delivered when the user enters a Web request. The objects are bound by maximum transfer unit (MTU) in the subsystem. The MTU is mainly given by the physical limitations of the system. A TCP IP connection is composed of a number of sessions; each session is composed of one main object and a number of inline objects. All objects are quantized into packets according to MTU size, which is modeled as 1500 bytes in this article. The model simulates the on-off source, where the on state represents activity when the browser is working and the off state represents when the browser is not working. Detailed parameters modeling different instances are specified in [6].

**VIDEO TRAFFIC MODELING**

Video traffic consists mainly of two important parts: constant data rate traffic due to the scalable video ability of H.263 and MPEG-4 standards [7], and variable bit rate traffic, where the video codec is assumed to be able to identify layered substreams.

A scalable video stream is coded at different rates, according to the available bandwidth for transmission to the decoder. This enables users to access a higher-bandwidth channel for decod-
ing high-quality video, while the low-bandwidth user can still view the basic video stream at rather low quality. Therefore, depending on user QoS requirement the video can be streamed to the specific user. It consists of an intracoded picture (known as the I frame) and multiple inter-coded pictures (known as predicted or P frames). The enhancement layers, which require additional bandwidth for encoding, will improve according to the spatial or temporal quality of the reconstructed base layer sequence. Four scalability approaches can be classified: temporal, signal to noise ratio (SNR), spatial, and hybrid scalability [7].

Taking SNR scalability as an example, it refers to the process of improving the SNR of the base layer picture by including additional information in the enhancement layer. The additional information is encoded as either enhancement I (EI) or enhancement P (EP) frames. An EI picture is only upward predicted from a decoded picture in the reference layer and does not use motion vectors, whereas an EP picture can be either upward predicted from the reference layer, forward predicted from the previous frame in the enhancement layer, or by some combination of both prediction modes.

A simple video model is used to investigate RRM for scalable video traffic. In order to have the same rate of information, we consider incoming traffic at rate $R_0$. The split traffic types are base layer at rate $R_E$ and enhancement layer at rate $R_B$. $R_B$ is modeled as bell like distribution, which maintains a tight relationship with the base and enhancement layers; whereas, base layer substream $R_E$ is modeled as constant, identical to data channel capacity in UMTS. The enhancement layer is left as $R_E$, which is the result of rate control.

To model the overall data rate, we put Gaussian noise with unit variance and average value $0.572$ in a FIR filter with coefficient vector $[0.8781, 0.1108]$. With the assumption of $2.5 \times 10^4$ pixels/frame and 30 frames/s, a stream of video traffic with average rate of 390 kb/s can be generated [8].

**InVESTIGATION OF JRRM IN COUPLED RATs**

With the information of the estimated load in all subnetworks, the joint load control entity (JOLDC) located together with the joint admission control entity will distribute the traffic based on the characteristics of the coexisting RATs. Based on the QoS requirement of the user traffic, the amount of resources to be offered is assigned to the contributing subnetworks. The network-specific part is introduced in order to clarify the difference between CDMA- and time-division multiple access (TDMA)-based scheduling algorithm.

**DEFINITION OF QoS**

Since the average rate of HTTP traffic has a long tail distribution phenomenon, a self-similarity concept based QoS parameter is defined that is used to evaluate the performance of a scheduling algorithm over HTTP traffic. The QoS parameter is defined as the ratio between scheduled traffic and demanded traffic, which is inside the range $[0, 1]$ where the best user QoS cannot be higher than 1. For video traffic, the conventional delay time is used to evaluate the performance of scheduling time.

**RESTRICTIONS IN BOTH SUBNETWORKs**

In a UMTS network, traffic is scheduled based on the transmission time interval (TTI) and transmission format of the physical channel. While the data stream is transferred between the medium access control (MAC) and physical layers, a transport block (TrBk) corresponding to the MAC packet data unit (PDU) is conveyed. The TTI equals the periodicity at which a TrBk is transferred by the physical layer on the radio interface. It is always a multiple of the minimum interleaving period (e.g., 10 ms, the length of one radio frame). TrBks may have different numbers of blocks and are not all necessarily active [9].

In H/2 data link control (DLC) PDU trains, only logical channels (LCHs) are assigned to
carry user data [10] (i.e., a basic delivery unit per user per session through the H/2 air interface is 54 bytes). According to the structure of the MAC frame, the payload of the traffic is combined with the control channels to be scheduled in the 2 ms frame. The maximum time for the scheduler to utilize depends on the number of links, which makes the length of the forward channel (FCH) vary, as well as other broadcasting control channels, feedback channels, and associated short channels.

The scheduling performance is investigated based on a fixed spreading factor in a UMTS channel; the numbers of LCH channels are calculated to maintain fairly stable QoS over heterogeneous networks.

**Joint Scheduling of Data Traffic**

As depicted in Fig. 3, traffic arrives at the joint scheduler where the data is classified into main objects and inline objects.

Right before the end of a scheduling chain, delay estimation is done for each channel and used to combine substreams together according to user QoS constraints. In order to obtain the same QoS distribution in substreams, an average number of LCH channels assignment is obtained corresponding to available capacity in the UMTS subsystem, as shown in Table 1.

In HTTP traffic, the QoS obtained from both the UMTS and H/2 channels should be the same from the user point of view. The user needs constant QoS irrespective of the number of networks used. Table 1 is obtained so as to get the same QoS from both RATs.

If the delay difference exceeds the delay threshold, only the main objects are shown to the user, so minimum QoS is always maintained. Otherwise, the data from the subnetworks of both UMTS and H/2 are recombined for the user and obtained on the terminal. Although the delay threshold is also kept as a measure for HTTP traffic, it is not as critical as for video traffic.

**Video Traffic Scheduling Algorithm over H/2**

Scheduling video traffic for a single user through two networks is very similar to the HTTP traffic case. The delay threshold is of major importance for video traffic and much more critical than in the HTTP case. Moreover, opposite to HTTP traffic, the base layer of the video traffic is constant bit rate traffic; the enhancement layer is variable bit rate traffic. Thus, the scheduling algorithm for video traffic is a little different from that for HTTP traffic.

As seen in Fig. 3, the joint RRC entity (JOSAC) receives the video traffic and, based on the available UMTS capacity, informs the video codec of the scaling scheme. The joint scheduler in the RAN separates this base and enhancement layer data, and then sends the constant base layer data through the dedicated channel of the UMTS-based cellular network; the enhancement layer data is channeled to the H/2-based system.

Due to the real-time nature of video traffic, certain synchronization of the data from the two subnetworks must be implemented in order to support QoS. The traffic must be combined within the delay threshold given, or the amount of resource must be varied to obtain the same, as shown in Table 1.

<table>
<thead>
<tr>
<th>Equivalent average numbers of LCH channels per radio frame for various UMTS channel capacities.</th>
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<tbody>
<tr>
<td>UMTS channel capacity (kb/s)</td>
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<tr>
<td>-------------------------------</td>
</tr>
<tr>
<td>7.5</td>
</tr>
<tr>
<td>15</td>
</tr>
<tr>
<td>30</td>
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<tr>
<td>60</td>
</tr>
<tr>
<td>120</td>
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<tr>
<td>240</td>
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<td>480</td>
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</tbody>
</table>

**Table 1.**

For multiple accesses, different scheduling algorithms in an individual radio network should be compared. Since the UMTS assigns a constant bit rate based on the base layer, H/2 is of great interest for us to investigate. The round-robin algorithm is a good scheme for fairness scheduling, but it does not keep track of the amount of traffic left for the user and schedules user data based on the constant quantum size, which cannot guarantee optimal resource allocation. The Last Finished First Served (LFFS) algorithm solves the disadvantage of the round-robin algorithm. In the LFFS algorithm, the queue of each user is compared, and the users are prioritized based on queue length. The user with the largest queue is scheduled first and so on, until either the end of radio frame is reached or the data of the user is over. At the end of the radio frame, the user priority is recalculated.

For highly loaded systems, more users can be admitted in the system with LFSS, as the delay occurred is less than with the round-robin algorithm.

**The Synchronization Approach and Buffer Management**

Due to the heterogeneity of coexisting networks, the performance of a joint scheduling algorithm is highly dependent on synchronization, as depicted in Fig. 4. In unsynchronized traffic from different layers, if the delay difference overrides a certain threshold, the user/terminal will not wait until all of the information from the different sublayers is received. Also, the target of traffic scheduling is to reduce the individual system load in subsystems, and if an unsynchronized case occurs, radio resource will be wasted; unnecessary delivery through one or more air interfaces occurs. As Fig. 4 shows, the delay time caused by individual subnetworks ($T_{n1}$ and $T_{n2}$) (e.g., the delay time to the Inb interface) is highly dependent on transport system architecture. For instance, a six-hop synchronous digital hierarchy (SDH) µ-wave link brings extra delay to user
traffic. In a distributed WLAN environment, the delay time is more difficult to estimate. Due to unexpected delay and jitter, a robust synchronization approach is required for the joint scheduling algorithm. As shown in Fig. 4, from different subsystems traffic streams are accumulated in the individual queue with different arrival times. The target is to reduce difference between the arrival times from transceivers chains in subsystems. The queuing size therefore also shrinks. The synchronization can be applied in an interactive, a passive (assigning a delay bound), or a predictive way with synchronization avoidance.

Take the interactive synchronization approach as an example: a training sequence is transmitted to the terminal periodically in order to allow estimation of the delay caused (timestamp) by the independent paths. With the timestamp, the terminal can autonomously decide to wait for the coming layered information or not. On the other hand, an uplink channel can be established to transmit the timestamp back to the network in order to tell the joint scheduler to upgrade the scheduling parameters (e.g., the quantum size organized by the round-robin algorithm in the subsystem).

The joint scheduler obtained delay differences from the training period periodically. Assume that a training sequence is sent each $T_{\text{s}}$; in each training period, the RNC sends the sequence over UMTS and H/2 air interfaces with the same channel and multiplexing algorithm as the data to be scheduled. After the training process, the terminal obtains the estimated delay difference, $\Delta(t)$, which is upgraded to $\Delta(nT)$ in a discrete timescale. The target of the training process is to offer the knowledge of delay difference between scheduled data in UMTS and in H/2. An interactive process takes place directly after the training process, as depicted in Fig. 5.

The synchronization process aims to minimize the difference between concatenating $\Delta(nT)$ and the absolute value of $\Delta(nT)$. The task of minimizing both values is, on one hand, to enhance the stability of the video decoder to give the best-quality video sequence. The size of the buffer, which is used to remedy jitters, can be kept rather constant, and bring convenience and efficiency to the decoder; the last traffic units of different streams belonging to the same packet, session, or decoding interval arrive at the same time.

There are two basic advantages of such a method: estimated delay values can easily be taken into account and sent on both networks as a timestamp in the radio frame. This helps the joint scheduler to schedule more intelligently, say, by artificially delaying one stream or assigning a longer transmit time value to a faster delivered stream. The second advantage is valid for a video decoder, wherein the joint scheduler informs the decoder of the estimated delay. For instance, if the estimated delay time for an H/2 stream is too high, the decoder does not wait for it and only decodes the base layer. As Fig. 5 shows, if the terminal enters the H/2 coverage area and the traffic split is commonly agreed between the multimode terminal and radio network after the initialization phase and before traffic splitting starts, the training phase of the synchronization procedure is carried out. The terminal must report to the joint scheduler its maximum buffer size with the measurement of delay difference, as shown in Fig. 5. Based on the delay difference, the network assigns the terminal the waiting time for it to adjust the buffer in both links. If the available buffer size does not fulfill the requirement to remedy the different difference and jitters, the traffic split should be cancelled (i.e., only a single link is maintained).

The synchronization procedure can be based not only on periodicity but also on demand, as shown in the optional signaling directly after the initialization phase. The reason to keep the possible on-demand synchronization application signal from the terminal is to guarantee the reliability of synchronization. If the terminal loses synchronization after unexpected situations (e.g., handover phases or loose connections in sporadic coverage of H/2), the terminal can initiate synchronization.

**CONCLUSIONS AND OUTLOOK**

For future systems with coexistence and interworking of heterogeneous networks, network radio resource management modules must support an integrated RATs structure with simultaneous connections over multiple radio links. Incoming traffic could be prioritized and split into different radio networks. A joint scheduler can allocate radio resources from associated sub-networks to substreams toward target QoS. An RNC must have a mapping table to give the mapping of the number of LCH channels required in H/2 for similar QoS in UMTS for split traffic streams. Although the mapping will be upgraded according to different user densities, QoS requirements, and network loads, it gives a fast estimation of resource allocation to corresponding traffic types. The Last Finished First Serve algorithm is shown to be a suitable scheduling algorithm with lower delay and higher throughput than other scheduling algorithms.
for high-data-rate traffic. Synchronization guarantees the reliability and QoS of traffic splitting in an adaptive radio multihoming approach.

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**REFERENCES**


**BIOGRAPHIES**

JUIN LUO (jesse.luo@siemens.com) received his M. Eng. degree from Shandong University, China, 1999, and his M. Sc. degree from Technical University Munich, Germany, in 2000. He joined Siemens in 2000, heading for his Ph.D. from Aachen University, Germany. He has published numerous technical papers and holds many patents, mainly in information theory, mobile communication systems, software-defined radio, and radio resource management.

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MARKUS DILLINGER received his diploma in telecommunication in 1990 from the University of Kaiserslautern, Germany. In 1991 he joined the Mobile Network Division at Siemens and developed call processing software for GSM base stations. Later, he joined the system engineering division responsible for evaluation software for channel sounder measurement equipment and GSM base station line interfaces. Since 1995 he has worked on the definition of the third mobile radio generation in the European research project FRAMES, and in 1999 was appointed technical manager of FRAMES. Since January 2000 he has been the project leader of the European research projects TRUST and SCOUT for software radio.

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EGON SCHULZ received his diploma in physics from the University of Siegen, Germany, in 1982, and his Ph.D. from the Department of Electrical Engineering, Technical University of Darmstadt, Germany. In 1988 he joined the Mobile Network Division at Siemens AG, Munich, and investigated and developed radio link control strategies for GSM and other standards. In addition to system modeling, he was a member of the ETSI standardization group for the GSM half rate speech codec. After one year as a professor at Fachhochschule, Darmstadt, he returned to Siemens as director of system engineering in 1993. Since 1998 he has served as director of radio access network simulation for 3G mobile systems.

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![Figure 5. Illustration of interactive synchronization.](image-url)