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Optimized Network Access in Heterogeneous Wireless Networks

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Zusammenfassung

Die vorliegende Dissertation ist am Fachgebiet Kommunikationstechnik (ComTec) im Fachbereich Elektrotechnik / Informatik der Universität Kassel verfasst worden. Ein Teil der vorgestellten Ergebnisse entstand in einer durch Cisco Systems (USA) geförderten Zusammenarbeit mit dem Institut für Nachrichtenverarbeitung der Universität Siegen. Die meisten der hier vorgestellten Ergebnisse wurden im Rahmen einer Zusammenarbeit mit Alcatel Research & Innovation (Stuttgart) innerhalb des durch das Bundesministerium für Bildung und Forschung (BMBF) geförderten Forschungsprojektes *IPonAir* (Laufzeit 01.07.2001 – 30.06.2004) erzielt. Der Autor ist als wissenschaftlicher Mitarbeiter am Fachgebiet Kommunikationstechnik tätig und war als Projektleiter bei ComTec für den erfolgreichen Verlauf des Forschungsprojektes verantwortlich. Teile der hier vorgestellten Ergebnisse wurden bereits in Veröffentlichungen und Patentanträgen vorgestellt (siehe hierzu Preface ab Seite 9).

Es wurden drei Möglichkeiten untersucht, durch die Nutzung neuer Ansätze die Leistungsfähigkeit drahtloser Netzwerkzugänge zu verbessern. Die erste Untersuchung stellt die Möglichkeiten eines drahtlosen Zugangs basierend auf diffuser Infrarotübertragung am Beispiel einer neuen Detektortechnologie vor. Die zweite Untersuchung betrachtet die Kapazitätssteigerungen, die sich aus einer Ergänzung zellularer CDMA-basierter Mobilfunksysteme wie UMTS durch Ad-Hoc Relais ergeben können, und zeigt schematisch einen möglichen Lösungsansatz für die notwendigen Modifikationen der Signalisierungsabläufe. Die dritte Untersuchung bildet den Schwerpunkt dieser Dissertation und beschäftigt sich mit dem so genannten *Multi Standard Radio Resource Management* (MxRRM). Hierbei wird versucht, durch eine intelligente Kopplung unterschiedlicher Mobilfunksysteme die Leistungsfähigkeit des heterogenen Gesamtsystems zu vergrößern. Zentraler Punkt ist die Betrachtung von Algorithmen und deren Leistungsfähigkeit auf Basis umfangreicher Simulationen.

Infrarotbasierte Datenübertragung eignet sich für Innenraumanwendungen. Licht als Übertragungsmedium erlaubt hohe Datenraten, die Übertragung ist weitgehend auf einen Raum begrenzt, so dass sich das Spektrum gut wieder verwenden lässt und das unbe-

rechtigte Abhören einer solchen Übertragung erschwert wird. Auch entstehen durch Infrarot keine Probleme bezüglich Elektromagnetischer Verträglichkeit (EMV), wie sie bei Einsatz von Radiowellen leicht entstehen können. Darüber hinaus steht Infrarot im Gegensatz zu Radiowellen nicht in dem Verdacht, Gesundheitsschäden zu verursachen. Diesen Vorteilen stehen bislang Einschränkungen bei der technischen Realisierung gegenüber. Üblicherweise werden gerichtete Verbindungen über kurze Distanz eingesetzt, um die notwendige Signalqualität zu erhalten. Dieser Ansatz ist jedoch für den Anwender unkomfortabel. Diffuse Verbindungen bieten bessere Einsatzmöglichkeiten, scheitern aber in der Regel an der erzielbaren Signalqualität. Mittels eines neuen Detektorelements, dem Photonic Mixer Device (PMD), scheinen nun diffuse Infrarotverbindungen mit hoher Datenrate möglich. Die Betrachtungen hierzu finden sich in Kapitel 2 ab Seite 25 dieser Arbeit.

In Kapitel 3 ab Seite 37 wird der Ansatz untersucht, zellulare Mobilfunknetze mittels Ad-Hoc Relais zu erweitern. Der Grundgedanke hierbei ist, dass die Kommunikation zwischen Basisstation und Teilnehmer nicht mehr direkt, sondern über ein mobiles Relais stattfindet, wobei die Kommunikation zwischen mobilem Relais und Anwender über ein Ad-Hoc basiertes Funksystem wie zum Beispiel Bluetooth oder WLAN geführt wird. Dadurch wird die Distanz, die durch das zellulare System überbrückt werden muss, verringert. Auf diese Weise kann die Reichweite und Abdeckung einer Basisstation vergrößert werden. Eine Verkürzung der Distanz kann aber auch zu einer Reduzierung der Sendeleistung genutzt werden, was durch eine Verringerung der erzeugten Interferenz insbesondere bei CDMA-basierten Systemen wie UMTS unmittelbar zu einer Vergrößerung der Kapazität führt.

Das Hauptthema der Dissertation wird ab Seite 69 in den Kapiteln 4, 5 und 6 behandelt. Die Entwicklungen im Bereich der Kommunikationstechnik haben zu einer Pluralität der Mobilfunksysteme geführt. Ein mobiler Anwender hat beispielsweise schon heute potentiell GSM, UMTS und WLAN zur Auswahl, wenn er eine Verbindung herstellen möchte. In dieser heterogenen Umgebung existieren die unterschiedlichen Systeme weitgehend unabhängig voneinander. Durch Multi Standard Radio Resource Management sollen diese Systeme sich gegenseitig so ergänzen, dass ein Gesamtsystem mit höherer Leistung entsteht und die Dienstequalität wie auch die Diensteverfügbarkeit erhöht werden. Durch die Kombination der Systeme können Bündelungsgewinne erzielt und die charakteristischen Vorteile einzelner Systeme in bestimmten Situationen oder für bestimmte Dienste ausgenutzt werden. Durch ausführliche Simulationen wurde mit dieser Arbeit gezeigt, dass auch unter Berücksichtigung der Fähigkeiten der realen Systeme die mit MxRRM erzielbaren Gewinne erheblich sind.

Preface

A major part of the research work presented in this dissertation has been performed in the context of the national research project IPonAir, which was funded by the German Ministry of Research (Bundesministerium für Bildung und Forschung, BMBF) and managed by the DLR (Deutsches Zentrum für Luft- und Raumfahrt). The Chair for Communication Technology ComTec, Universität Kassel, participated as a partner of Alcatel Research & Innovation, Stuttgart, Germany.

The research work presented in chapter 2 was performed as joint work with the "Institut für Nachrichtenverarbeitung" at University of Siegen, Germany, and was supported by Cisco Systems, Inc., USA.

Most of the results constituted in this doctoral thesis have been published recently in journals and on international conferences. Additionally, two patent applications evolved from this research work.

In chronological order, the publications are:

[Hil_1] Matthias Hildebrand, Klaus David, Bernd Buxbaum, Markus Grothof, Rudolf Schwarte: *"A new Approach for Indoor Wireless Internet based on Diffuse Infrared"*, IASTED Wireless and Optical Communications Conference, Banff / Calgary, Canada, June 27 - 29, 2001

[Hil_2] Bernd Buxbaum, Rudolf Schwarte, Thorsten Ringbeck, Matthias Hildebrand, Klaus David: *"Wireless LAN based on Optical CDMA using a new High Speed Correlation Receiver (MSM-PMD)",* SPIE's International Symposium ITCom 2001, Denver, Colorado, USA, August 20 - 24, 2001

[Hil_3] Matthias Hildebrand, Gabriel Cristache, Klaus David, Frank Fechter: *"Location-based Radio Resource Management in Multi Standard Wireless Network Environment"*, IST Mobile & Wireless Communications Summit, Thessaloniki, Greece, June 17 - 19, 2002

[Hil_4] Matthias Hildebrand, Guihua Piao, Klaus David, Gabriel Cristache, Frank Fechter: *"Efficient Estimation of Transmission Power applied to The Simulation of the Cell Breathing Effect in CDMA-based Wireless Systems"*, IASTED International Conference Applied Modeling and Simulation (AMS 2002), Cambridge, USA, November 4 - 6, 2002

[Hil_5] Gabriel Cristache, Klaus David, Matthias Hildebrand, José Diaz, Rolf Sigle: *"Aspects for the Integration of Ad-hoc and Cellular Networks"*, 3rd Scandinavian Workshop on Wireless Ad-hoc Networks (ADHOC 2003), Stockholm, Sweden, May 6 - 7, 2003

[Hil_6] Matthias Hildebrand, Klaus David, Guihua Piao, Gabriel Cristache, Frank Fechter, Rolf Sigle, Anja Warich: *"Investigation of Cell Breathing Effect in CDMA-based Cellular Systems"*, IST Mobile & Wireless Communications Summit, Aveiro, Portugal, June 15 – 18, 2003

[Hil_7] Gabriel Cristache, Rolf Sigle, Matthias Hildebrand, Klaus David, *"Power Efficient Signaling Protocols for the Cellular with Ad-hoc Integration"*, World Wireless Congress (WWC 2004), San Francisco, USA, May 25 – 28, 2004

[Hil_8] Matthias Hildebrand, Klaus David, Guihua Piao, Rolf Sigle, Dietrich Zeller, Ingo Karla, *"Performance Investigation on Multi Standard Radio Resource Management for Circuit Switched Services"*, IST Mobile & Wireless Communications Summit, Lyon, France, June 27 - 30, 2004

[Hil_9] Matthias Hildebrand, Guihua Piao, Klaus David, Rolf Sigle, Dietrich Zeller, Ingo Karla, *"Performance Investigation on Multi Standard Radio Resource Management for Packet Switched Services"*, IEEE Vehicular Technology Conference (VTC 2004-Fall), Los Angeles, USA, September 26 - 29, 2004

[Hil_10] Matthias Hildebrand, Klaus David, *"Estimation of User Density required for Ad-Hoc Extension of Cellular Systems"*, submitted to IEEE Transactions on Wireless Communication

[Hil_11] Matthias Hildebrand, Klaus David, Rolf Sigle, *"Resource Management in Heterogeneous Wireless Networks"*, submitted to IEEE Transactions on Mobile Computing

[Hil_12] Matthias Hildebrand, Guihua Piao, Klaus David, Rolf Sigle, *"Performance Investigation on Multi Standard Radio Resource Management in Mixed Service Scenarios"*, submitted to IEEE Communication Letters

In chronological order, the patent applications are:

[Hil_13] Frank Fechter, Rolf Sigle, Ulrich Barth, Matthias Hildebrand, Klaus David, *"A location map for a radio access network"*, Alcatel patent application to the European Patent Office, No. 02360331.9, March 03, 2003 (Status: withdrawn because of a similar application made by a competitor)

[Hil_14] Klaus David, Rolf Sigle, Matthias Hildebrand, José Diaz Cervera, Gabriel Cristache, *"Method for controlling the sleep mode on a mobile terminal corresponding mobile terminal and corresponding radio access node"*, Alcatel patent application to the European Patent Office, No. 04290730.3, May 26, 2004 (Status: under review)

Content - Overview

Content

1 Introduction

In recent years the use of wireless devices has become common practice particularly in the field of communication technology. The range of wireless communications spans from simple devices realizing a mere cable replacement to more complex approaches offering direct connectivity to the internet or to the public telephone system by realizing a complete system solution or even by constituting a network of its own.

In Germany the public mobile communication was started by a telephone mounted in a train in 1926. The first actual mobile telecommunication system that was publicly available and commercially operated was the “A-Netz”, which started in 1958. In 1972 the “B-Netz” was established, introducing automatic switching to mobile communications. With the “C-Netz” starting in 1986 nearly 100 % coverage was achieved in Germany (West). Other European countries installed their own cellular telecommunication systems, which where similar but not identical to the “C-Netz”. Consequently, international roaming was impossible and the vendors could provide only a small variety of infrastructure as well as end user terminals for each system. Eventually a joint European development created the Global System for Mobile Communication (GSM), which was used to start the “D-Netz” in 1992 [1].

With this event by launching the first completely digital cellular phone system, i.e. GSM, the second generation (2G) of mobile telecommunication systems has been introduced. Today GSM is supposed to be the most successful system with more than one billion subscribers twelve years after its start [2].

The original GSM was designed primarily for one particular narrow band real time circuit switched service: for voice calls. Over time the request for data services, which do not require real time capabilities nor circuit switched connections but would greatly benefit from a higher bandwidth, was increasing so the so called 2.5G technologies GPRS and EDGE have been developed to enhance the existing GSM networks.

Despite the overwhelming success of GSM, the third generation (3G) of mobile telecommunication systems, i.e. IMT2000, has been developed and is currently being introduced globally in either of its subspecies CDMA2000 or UMTS. The European countries as well as others implement UMTS, which is standardized by the 3rd Generation Partnership Project (3GPP) [3]. UMTS is supposed to provide better spectrum efficiency as well as better support of several types of services, particularly high bit rate data services, than the previous systems.

Additionally, the wireless local area networks (WLANs) experience a strong growth. These systems have been derived from computer networks and could be seen as kind of a wireless Ethernet. Consequently, their original purpose was the transmission of data, meaning non real time packet switched services, using higher bandwidths. Reacting to the strong demand for data services many operators complement their cellular system by offering WLAN access at special locations ("Hot Spots") like congress centres, shopping centres, train stations or airports. The most common type of public WLAN access currently is based on the IEEE 802.11b standard [4].

Hence, the migration from 2nd generation mobile communication systems to 3rd generation systems is creating a heterogeneous environment where different radio access technologies (RATs) offer their services independently to the user. These RATs will consist primarily of cellular systems like GSM/GPRS and UMTS, but may also include other systems like WLAN.

Since frequency spectrum, being a scarce resource, is shared by an increasing number of systems, operators and users a common issue of all the systems mentioned is the limited capacity of the air interface, which evolves into being the bottleneck of the overall system.

There are numerous possible options to increase the capacity of the overall system. This research work focuses on methods that do not change the fundamental concepts of the individual communication systems involved. So for example optimizing modulation techniques or the medium access protocols is not considered. Here basically three different approaches are investigated:

- Introduction of diffuse infrared as a new wireless access technology
- Extension of cellular systems by means of ad-hoc relaying

- Multi Standard Radio Resource Management (MxRRM)

The investigation of MxRRM performance constitutes the main part of this research work.

In the following subsections these approaches are shortly introduced. In chapter 2 the concept of wireless network access based on infrared is exemplified through the discussion of a promising new detector technology and in chapter 3 extension of cellular systems by means of ad-hoc relaying is discussed in more detail. In chapter 4 the idea of Multi Standard Radio Resource Management is described extensively. The powerful simulation environment created during this research work is introduced in chapter 5 and the simulation results are presented in chapter 6. Finally, in chapter 7 a summary is given.

1.1 Wireless Access by Diffuse Infrared

Radio spectrum does not constitute the only way to enable wireless network access; also the optical spectrum could be utilized to provide wireless connectivity.

Contrary to radio waves the infrared spectrum is license free. Additionally, the costs for transmitters and receivers are potentially less than the costs for radio transceivers. Hence, using infrared as wireless access medium can be interesting economically.

From a technical point of view infrared also has attractive capabilities: It offers the possibility for very high bandwidths and due to the optical propagation the coverage of an access point can be determined relatively accurate thus allowing for more efficient spectrum re-use than radio waves.

A shortcoming of infrared is the sensitivity to interference caused by other light sources like the sun or incandescent lamps. As a result, infrared is considered suitable for indoor communications only. Even then usually a directed beam is required to enable sufficient bit rates, but only non-directed – diffuse – infrared connections would allow for true wireless access similar to radio waves.

In this thesis a new approach for diffuse infrared communication is discussed.

1.2 Ad-Hoc Relaying

For increasing capacity the mobile terminal could establish the connection to an intermediate node, a relay, instead of connecting directly to the base station. As a result the range of the base station could be extended. Additionally the capacity could be increased because the radio path is split up into two segments. Since the propagation loss is increasing over proportional with the distance this splitting reduces the transmission power required thus reducing the interference created by this connection, which leads to an increase of system capacity.

The use of fixed relay stations for extending the capabilities of infrastructure based wireless systems was already extensively discussed, see for example [5].

The use of mobile relay stations that potentially act at the same time as mobile relay and mobile client is typically considered by research that is related to pure ad-hoc networks. In [6] an approach is described that uses the cellular interface for relaying. This research work provides a discussion on the extension of cellular systems by ad-hoc relays, which recently experiences a growing interest in the scientific community, see for example [7].

The basic idea is that the mobile terminal performing a call – the mobile client (MC) – does not physically connect to the base station but to a second terminal – the mobile relay (MR) – which forwards the connection to the base station. Since the connection between MC and MR uses an ad-hoc type of air interface, e.g. WLAN or Bluetooth, only the relatively short connection between MR and base station has to be provided by the cellular system. Particularly for CDMA based systems this approach would reduce the cellular resources consumed by a call significantly and therefore increase the system capacity by a considerable amount.

In this thesis an estimation of the possible capacity gain is provided as well as a discussion of the feasibility by deriving the probability of finding a MR as well as by presenting possible approaches for protocols to support this feature.

1.3 Multi Standard Radio Resource Management

By applying a common Multi Standard Radio Resource Management (MxRRM) the heterogeneous environment could be utilized in a way that the different radio access technologies complement each other.

The first benefit to be expected would be an increased capacity due to the trunking gain created when combining the resources of different systems. A second benefit would be an increased capacity due to improved efficiency by assigning service requests of different types to the most appropriate radio access technology.

For this work the coupling of GSM/EDGE and UMTS is considered, because these systems already co-exist in the real world. In the current version these systems act independently, offering only a kind of “blind” handover between GSM and UMTS in case a call would be lost. A mature overall radio resource management is not included yet; future implementations based on UMTS Release 5 or later will provide an interface for the exchange of load information messages without specifying how to exactly utilize this information [8].

Thus, it can be supposed that the application of MxRRM to current cellular communication systems is technically feasible. Consequently, this research work focuses on the estimation of possible performance gains achieved by MxRRM. For this purpose, several MxRRM algorithms have been analyzed by means of extensive simulations.

2 Wireless Access by Diffuse Infrared

Indoor wireless network access is one important part of today's research in communication technology. Wireless access enables the user to place or even to move his communication device with no consideration for the existence of connecting wires or wall sockets.

For seamless wireless extension of the fixed network a further requirement are bit rates far above 10 Mb/s. Possible solutions could be based either on radio frequency or optical infrared signals as different research projects are showing.

In this section an approach based on a new infrared detection technology is presented, which was invented at the University of Siegen by Prof. Dr.-Ing. Rudolf Schwarte. The new Photonic Mixer Device (PMD) is an infrared detecting sensor element that physically correlates the signals within the detection process in real-time. Due to this unique capability it has been successfully utilized in Infrared Laser Radar systems. Transferring this technology into the field of communications is promising to enable the realization of high performance multi-user optical wireless indoor communication systems.

The experimental results proved the principle and showed possible device bandwidths in the GHz-range. In this section, the fundamental operational principle of the device and the application vision is discussed, thus expounding the capabilities of the new approach.

In modern communication technology there is on the one hand a strong growth of bandwidth request and on the other hand an increasing demand for wireless applications. Today's solutions for indoor wireless network access are still limited in performance, especially concerning the bit rate as discussed below.

So an important task could be considered the development of a new high performance as well as low cost technology. The approach presented here allows for a high bit rate, multiple user access optical wireless indoor solution. The leading standard for indoor wireless network systems being installed today is the IEEE 802.11b standard. These systems operate at 2.4 GHz in the unlicensed ISM band providing a bandwidth of 11 Mb/s shared between the clients on air in the cell. The range (the radius of a cell) is several 10 meters in indoor environment [4].

Recent developments like IEEE 802.11a or HiperLAN use the 5 GHz band and provide gross data rates of 54 Mb/s [9], [10]. Further work to extend the families of IEEE 802.11 and 802.15 standards are ongoing. For even higher bit rates, like 155 Mb/s, extensive work has been undertaken for wireless ATM. These systems operate in the 40 or even 60 GHz frequency range. Besides the expense of components for these frequencies, an additional problem could be potential health hazards [11].

Another approach is based on infrared (IR) systems. There are a small number of wireless network systems based on diffuse infrared currently available but they provide low data rates (up to 4 Mb/s) [12]. There are also research projects, which under experimental conditions reach data rates in the order of 50 Mb/s using diffuse infrared links [13].

Despite the dominance of solutions based on radio frequency (RF) a system based on infrared (IR) communication links would offer certain advantages, which cannot be realized otherwise [14]:

IR transmissions are confined to a room, because the walls block them. Compared to RF this means an exactly defined shape and size of one cell and results in smaller cells. So the same spectrum can be re-used in the room next door for another communication link without interfering each other. This increases the overall system throughput significantly.

The IR spectrum is license-free worldwide and offers a large amount of bandwidth.

Infrared communication is invulnerable to matters of electromagnetic interferences (EMI). IR transmission does not disturb any other electronic device and the reception of IR signals is not interfered by electromagnetic fields. Since modern offices are crowded by several kinds of electronic devices this is certainly an advantage. Assuming the power used by communication devices does not exceed the limits given for eye safety there is no biomedical impact to the human organism contrary to RF, which is discussed to cause health-hazards.

A further non-technical but also important advantage of using IR is the relatively low costs of transmitters and receivers.

To achieve high data rates in IR systems usually directed line-of-sight links are preferred because they offer a relatively easy to handle communication channel since they minimize path loss and reception of ambient light [14].

Directed links need the communicating devices to be aiming at each user thus giving discomfort to the user. Furthermore they usually establish only point-to-point connections to one certain device so that multiple access is not possible. Therefore directed links do not allow for a powerful networking environment. With directed links the devices must not move during communication so only quasi-stationary operation is allowed, even more inconvenient: devices must be positioned which eliminates one important advantage of wireless communication.

With diffuse infrared we can allow the communicating devices to be used without positioning and even to move during communication and we can establish point-to-point and point-to-multipoint connections.

So the conclusion has to be that a high performance diffuse infrared network system would be the ideal solution for indoor wireless internet access.

2.1 Photonic Mixer Device: Basics and Fundamentals

The Photonic Mixer Device (PMD) is a new infrared detecting device that was developed by the University of Siegen's Institute for Signal & Data Processing (INV) and the Centre for Sensor Systems (ZESS). Its original purpose was the measurement of distances, i.e. to serve as infrared radar, but in the meantime the application to communication systems is being investigated also.

To explain the operation principle of a Photonic Mixer Device (PMD) a simplified CMOS-PMD structure is depicted in Figure 2.1 (a). Although there are other promising technology alternatives for the PMD realization (which have already been proven in theory as well as experimentally), they are not discussed in detail here. The PMD itself is a so-called smart pixel device because of signal processing features directly being integrated within its operation principle.

For its realization more detailed considerations, other than shown in Figure 2.1 (a), have to be taken into account, such as gate overlaps, geometrical dimensions, doping concentrations, doping profile, and many more. The charge transfer mechanism of PMD is, in principle, comparable to that of surface channel Charge Coupled Devices (CCD) - but there are some important differences that have to be considered. For example the charge movement in the PMD channel differs from the CCD approach, because the charge is moved back and forth inside light sensitive area in contrast to a monotonous transfer direction in CCDs [15].

Figure 2.1 (a) schematically illustrates the cross-section of a PMD pixel. It is a five terminal device with an optical input window, i.e., the two semitransparent modulation electrodes in the middle of the illustration. The photo gates are conductive, and they reside upon an insulation layer. This insulation layer is directly arranged over the p-type substrate for a surface channel realization

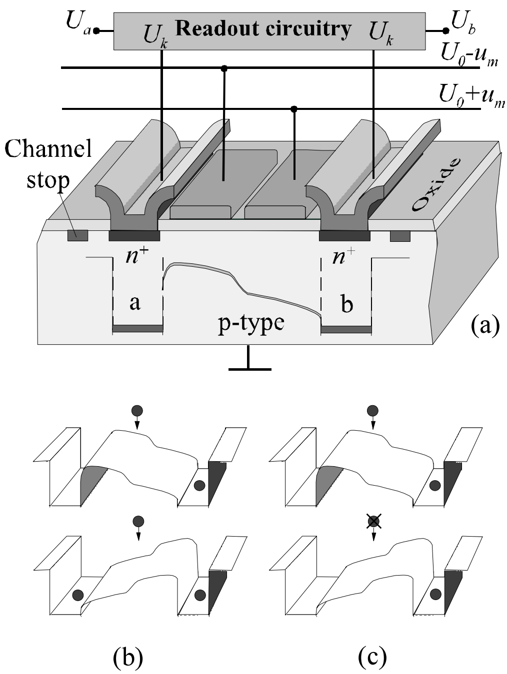


Figure 2.1: (a) Simplified PMD structure, (b, c) push-pull surface potential distribution, constant (b), and modulated light illumination (c).

of the PMD. Also buried channel realizations are imaginable, but they will not be discussed further in this contribution. The two photo gates, as well as the two outer pn-junction readout diodes, are surrounded by a channel stop diffusion to prevent inversion. The signals applied to the modulation gates are push-pull signals of arbitrary waveforms. Due to these applied modulation signals, the potential distribution inside is influenced in a way leading to a "dynamic seesaw" for the generated charges - as already indicated Figure 2.1 (b) and (c).

If the incoming light intensity is not modulated (e.g. any kind of background light, see Figure 2.1 (b)) the accumulated charge on both sides of the device is, on the average, equal. Only if there is a more or less timely dependence between the push-pull-modulation of the gates and the modulation of the light intensity, the resulting charge packets on both sides become unequal (refer to Figure 2.1 (c)). The difference of both output ports is directly dependant on the phase delay between the light and the pixel modulation and represents the correlation function which might be used for the synchronization as well as the bit decision as will be shown in the

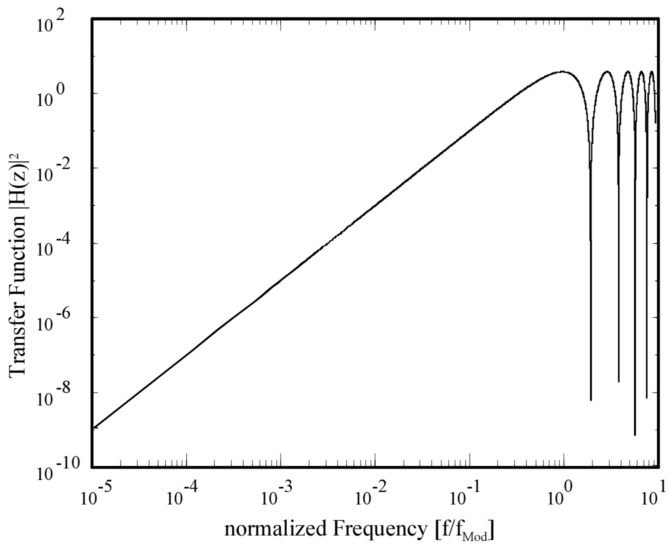


Figure 2.2: Simulated transfer function of the CBS-functionality inherently delivered by PMD-devices. As can be seen, a PMD is a re-configurable smart filter, which is only sensitive to correlated light signals.

latter parts [16].

Using an on-chip readout circuit already realized offers the possibility to subtract the uncorrelated light-portion – which results in equal signals on both readout ports as discussed above – directly during light reception. Therefore the device is, in a first estimation, “blind” to light that differs from the kind of intensity modulation that should be detected. Due to the push-pull sampling of the incoming signal even noise might effectively be reduced. This so called correlated balanced sampling (CBS) has also already been proven experimentally, thus enabling nearly independent detection of signals, even in heavy background scenarios.

Using the Z-transformation the idealized push-pull sampling, which is responsible for the CBS-functionality, can be written as:

$$\begin{aligned} U_{out} &= \Delta U_{a,b} = U_a(n \cdot T_{Mod}) - U_b((n-\frac{1}{2}) \cdot T_{Mod}) \\ \Leftrightarrow \quad H(z) &= \frac{U_{out}(z)}{U_{in}(z)} = 1 - z^{-\frac{1}{2}} \end{aligned} \tag{1}$$

With U_a, U_b *being the signal at the readout diodes,* T_{Mod} *being the period of the modulation signal and* n *counting these periods.* U_{in} *represents the optical input signal.*

Transformation in the frequency range delivers:

$$\begin{aligned} |H(\omega)|^2 &= 2 \cdot \left[1 - \cos\left(\omega \frac{T_{\mathrm{mod}}}{2}\right)\right] \\ &= 4 \cdot \sin^2\left(\omega \frac{T_{\mathrm{mod}}}{4}\right) \end{aligned} \tag{2}$$

With $\omega = 2\pi f$ *being the frequency of the optical input signal.*

Figure 2.2 shows an idealized transfer function of this push-pull sampling. As can be seen, lower frequency parts (in comparison with the modulation frequency) are suppressed. Raising the modulation frequency obviously improves this suppression.

2.2 Use in Communication Systems

The outstanding feature of PMD is the ability of physically correlating the received optical signal with an electrical control signal on-chip in real-time. So using PMD gives us improved signal quality and will probably enable us to use diffuse infrared connections with a high performance formerly impossible to realize.

Since PMD supports implicitly multiple access methods like CDMA due to its correlation capabilities this new device offers a very efficient way for a high performance multi user parallel optical wireless indoor communication system.

A PMD element is a kind of a small square pixel having momentarily a width of about 200 µm. Due to this geometrical shape a large number of PMD pixels can be easily arranged as an array enabling the use of advanced detection methods like angle-diversity receiving.

With angle-diversity the incoming signals are distinguished by their direction. Applying an appropriate algorithm to combine the several inputs a significant reduction of the effects of noise and multi path distortion can be achieved [14].

Since PMD can be produced in a cost-effective process similar to standard ASICs, additional circuits for signal pre-processing (for example a PLL for signal clock recovery or a pseudo noise code generator for direct sequence spread spectrum) can be easily realized on-chip. So PMD enables the design of powerful but low-cost communication devices.

To prove the applicability of PMD to communication systems a demonstrator was realized that uses a single LED with 10 mW average power for each channel. So far up to 10 channels with data-rates of 10 kb/s per channel were realized. As they can work totally independent from each other, an overall data-rate of 100 kb/s is reached. The system is able to work at distances of a few meters between transmitter and receiver (the range is mainly defined by the optics used for transmitting and receiving). So the feasibility of the new approach could be demonstrated.

Due to the fact that the values mentioned result from the limited performance of used system components (i.e. RS232 interface and CPLD design), the real experimental limits of the actual Photonic Mixer Device could not be found out so far [Hil_1], [Hil_2].

As already demonstrated in optical radar applications, the charge separation process of PMD-devices – and therefore also the correlation capability – works excellent up to several GHz at 1 pA of photocurrent and even clearly below. Consequently, PMD is supposed to enable high bit rate data transmission [17], [18].

2.3 Application Vision

A modern working environment usually consists of a number of different devices (desktop computer, notebook computer, PDA, phone, mobile phone, fax machine, printer, scanner, …), which have to communicate with each other or with a network (computer local area network, telecommunication network, …). A result of this situation is the existence of countless interconnecting cables, which are running on the desk, under the desk or behind the desk.

If you want to visit a neighbouring office or a meeting room and you do not want to miss the services provided by the devices mentioned you have first to unplug all the cables and then to plug them in the new room.

This state of affairs is not only very uncomfortable but often leads to situations in which the proper working of the devices can not be guaranteed because the proper wiring may be impossible.

To avoid this new solutions were developed and are still being developed. These wireless technologies are considered either as simple cable replacement or as a complete wireless local area network. All the technologies currently available have in common a small performance compared with the usual cable links like for example Fast-Ethernet. Additionally there are further limitations specific to each technology as discussed above.

Diffuse infrared indoor wireless network systems based on PMD technology promise to offer high performance and high usability so one can imagine different application scenarios, which would be very desirable to realize.

2.3.1 PMD as Cable Replacement

If PMD is the built-in communication interface of devices then these devices can communicate using the full data rate provided by PMD. Since PMD uses a diffuse Infrared link the devices do not need to point at each other. A user can simply enter a room, place his mobile device (for example a notebook computer) on a desk and control the printer located somewhere in the room.

The diffuse IR link enables the user to freely move the device inside the room during the ongoing data communication. Even the communication to several devices at the same time is possible.

With these features PMD offers not only a significantly higher performance but also more comfort and a higher usability than currently available wireless cable replacements.

2.3.2 PMD as Ad Hoc Network

Several users gathering in a room for let's say a meeting can interconnect their electronic devices, their PDAs and notebook computers, for the exchange of information. This interconnection will be done using the high performance diffuse infrared links provided by a PMD based system.

The high performance of PMD communication will enable even the use of sophisticated applications like video conferencing on each user's notebook computer. So every person in the meeting is able to join the videoconference from his/her seat, using the own built-in camera of the notebook computer. The conference could be recorded to the own personal hard disk, too.

Offering such a high performance and ease of use PMD will make ad hoc networking to become an everyday occurrence.

2.3.3 PMD as Wireless Local Area Network

To provide for example a building with a PMD based wireless LAN there has to be a PMD transceiver in each room. Every user who enters a room with a PMD-equipped mobile device is able to access the LAN by a diffuse Infrared link from every position within the room.

The bandwidth provided by PMD is shared among the users of one room using a multiple access method like for example CDMA. Since walls block infrared the complete PMD bandwidth is available again in every single room so not only the performance per link but also above all the system throughput is significantly higher than for currently available wireless systems.

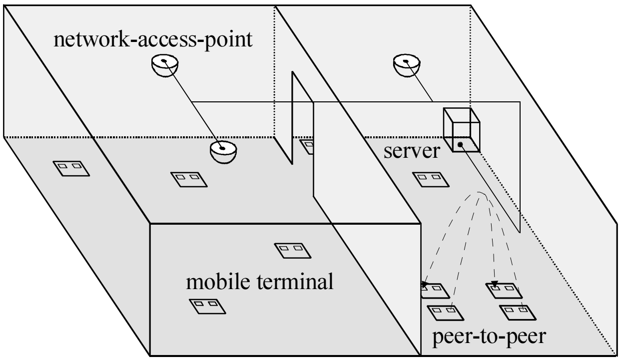


Figure 2.3 Possible application scenarios for a PMD-based LAN and as cable replacement for direct connections [14].

2.4 Conclusion

Communication systems based on diffuse infrared links offer a wide range of possible applications. They could be simple cable replacements, enable ad hoc networks or they could act as complete indoor wireless local area network solutions. The system approach detailed in this thesis is based on a new detection technology.

The new infrared detecting sensor element PMD (Photonic Mixer Device) possesses potential advantages for communication systems:

- High bandwidth of the sensor element of up to several GHz.
- Remarkable high suppression of uncorrelated signals like noise, neighbour channel interference, jamming, daylight and artificial light sources.
- Powerful signal detection even in difficult environments (heavy background noise, multi path propagation).
- Possibility to integrate additional circuits with the sensor elements on one chip, thus realizing small and cost-effective devices.
- Implicit support of multiple access schemes like CDMA.

Infrared Laser Radar applications already successfully exploit these capabilities for distance measurement so the high capacity of this technology is already confirmed. Experimental results also proved the basic function principle for communication applications.

Applying this detection technology to communication systems promises to result in optical wireless indoor communication systems that would offer a high usability in modern networking environments. PMD is produced in a very cost-effective way like any other integrated circuit. Since additional signal pre-processing circuits can easily be added onto the sensor element communication devices based on this technology are expected to be feasible at a low price.

PMD-based optical wireless communication systems obviously offer the potential of moving optical wireless communication technology forward by a quantum-step. To finally evaluate this potential and to identify the limits further experimental and theoretical work has to be done.

3 Ad-Hoc Relaying

Ad–hoc communication is becoming more and more successful in the marketplace as indicated by the increasing usage of Bluetooth and WLAN. This success is due to several reasons, including their lack of infrastructure requirements as well as their suitability for several application scenarios like the communication between laptops or handheld devices.

Despite all the successes so far and the promising future of ad–hoc networks, Internet connectivity will still be very important, and could be supported by cellular networks like GSM/GPRS or UMTS.

Another possibility is to use ad-hoc networks as an extension of the cellular air interface, leading to improvements in cellular capacity and coverage. Due to the reduced coverage range of Bluetooth, WLAN seems to be better suited for this purpose, and hence, different interworking solutions between UMTS system and WLAN are presented in this thesis. In section 3.1 the extension of cellular systems by means of ad-hoc relaying is introduced and an estimation of the potential capacity gain is provided in section 3.2. In section 3.3 an approach for the possible enhancement of the cellular signalling procedures is presented and the user density required for successful relaying is estimated in section 3.4

3.1 Ad-Hoc Extension of Cellular Systems

In this section, a possible ad-hoc extension of the UMTS cellular system is presented considering the one-hop case as shown in Figure 3.1. The basic idea is that it is possible for the mobile terminal (termed mobile client) to communicate with the base station not only directly but also using another mobile terminal in between that performs relaying (and hence is termed mobile relay).

There are several possibilities to realize the co-existence of cellular and ad-hoc links based on the use of either different or identical air interfaces and either different or identical frequency spectra. In case of sharing the same spectrum the mutual interference has to be minimized, e.g. by modifying the multiple access schemes appropriately.

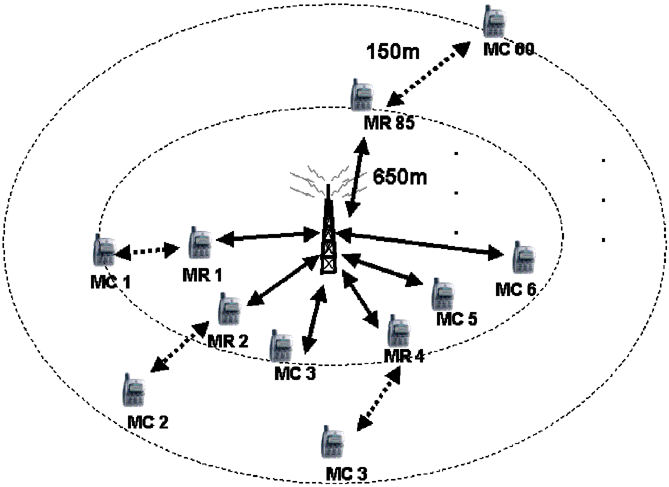


Figure 3.1: An example of the ad-hoc extension

Here the use of different air interfaces in different parts of the spectrum is considered. Since for the ad-hoc link no cellular radio access technology is required, the use of a less complex air interface is sufficient. Additionally this approach promises to achieve the highest capacity gain, because by utilizing additional spectrum more resources are made available to the system.

The coupling of cellular and ad-hoc systems offers benefits to both. In ad-hoc networks new services like connectivity to the internet could be made available. This aspect is not within the focus of this research work, though.

For the cellular system this approach offers an increase of both coverage and capacity [19], [20]. In a cellular communication system, terminals that are far away from the base station need to use an over-proportional large amount of transmission power (due to the non linear decrease of the received power with the distance) [21]. Using high transmission power means creating a high level of interference to users in the same as well as in adjacent cells. UMTS, being a CDMA based system, is interference limited, i.e. increased interference results in decreased capacity.

By using mobile relays, the distance to be covered by the cellular link and consequently the transmission power required is decreased. Thus the interference is reduced and the capacity of the cellular system is increased. An estimation of the possible capacity gain is presented in the next subsection.

A one-hop relay extension is also proposed in [5] by integrating an additional ad-hoc interface into the GSM protocol stack to enable relaying through this newly created interface. The integration of cellular networks and ad-hoc relaying technologies was also studied in the project “integrated Cellular and Ad hoc Relaying System (iCAR)” [22], [23]. The basic idea of the iCAR is to place a number of Ad-hoc Relaying Stations (ARSs) at strategic locations, which can be used to relay signals between mobile hosts and base station. Communication between a mobile host and a base station is done via the C-interface (C for cellular) while those among mobile hosts are via the R-interface (R for relaying). The R-interface (as well as the medium access control protocol used) is similar to that used in ad-hoc networks.

Additionally by using a mobile relay the connection could be routed to an adjacent cell. This would allow for the application of load balancing mechanisms, which potentially would further increase the overall system capacity [7]. Using load-balancing strategies, in [22] the authors show that the call-blocking probability of the iCAR system is significantly lower than for a corresponding cellular system.

3.2 Estimation of Capacity Gain

An estimation of the capacity improvement of the UMTS system using ad-hoc extensions was made using the simulation environment that was developed for the investigation of MxRRM, which is presented in section 4. Details on the simulator can be found in section 5.

In a first step the relation between UMTS capacity and coverage was investigated. Coverage of a UMTS cell is depending on the current load, and in turn the capacity of a cell is depending on the coverage. In a real system during operation this leads to fluctuations of both cell capacity and cell coverage, which is known as cell breathing effect. An investigation of this effect is presented in the next section.

In a second step, using these results the possible capacity improvements caused by the extension of the cellular system by means of ad-hoc mobile relays were derived and are presented in section 3.2.2.

3.2.1 UMTS "Cell Breathing"

In the context of the IPonAir project a simulator was created, which here was used for an investigation of the mutual dependency of cell coverage and cell capacity in a CDMA-based cellular system [24], the so-called cell breathing effect. In the following it has to be distinguished between the (nominal) cell radius and the cell coverage. The first term means the size of the cell as intended by the operator and realized by placing the base stations, whereas the latter one describes to which area the cell could provide service. Depending on the actual conditions the coverage of a cell could be larger or even smaller than the cell radius.

The desirable situation would be to have coverage larger than the nominal cell radius to get overlapping coverage regions, thus enabling smooth handover processes [25].

The cell breathing effect could be exploited to establish a kind of load balancing between adjacent cells [26], but this kind of radio resource management strategies is not in the focus of this investigation.

The simulation environment used for this investigation is described in detail in section 5; in the next section the configuration details relevant for this investigation are given.

3.2.1.1 Simulated Scenario

For this investigation a scenario with 42 cells was considered. The cells are arranged in seven columns and six rows according to the usual pattern based on hexagonal cell shapes. Each cell has a base station using an omni directional antenna at its centre.

The mobile stations are supposed to be quasi stationary, i.e. they do not move during an ongoing call but for each new call a mobile station is placed at a new position.

A typical set of values was chosen as reference configuration. For a particular investigation one parameter was varied. The reference configuration is:

- Nominal cell radius: 800 m.
- Maximum transmission power per cell: 20 W.

- Only centre cell is under investigation.
- All other cells are fully loaded, i.e. they are continuously transmitting with maximum TX power to create the worst case extra cell interference level.
- Soft Handover is disabled implicitly because all adjacent cells are full by default.
- Traffic is circuit switched with 8 kb/s user data rate, mean call duration is 120 s.

The position of the most distant user is continuously tracked to monitor the current coverage of the cell but this value is recorded only if the condition given in Equation (3) is true, i.e. the value is only recorded if the base station is operating close to the maximum transmission power and thus providing maximum coverage for the current load (vice versa: is providing maximum capacity for the current coverage).

$$P_{\max} \geq P \geq P_{\max} - 1dB \qquad (3)$$

With P_{max} *being the maximum transmission power of the cell and* P *being the current transmission power of the cell.*

If soft handover was enabled the users at the cell perimeter would tend to handover into the adjacent cell thus resulting in values for the maximum distance between user and base station that never would exceed the nominal cell radius significantly. Hence, the adjacent cells are configured as “being full”, i.e. they are transmitting continuously with maximum transmission power so that they are not able to accept handovering users but even so they are creating maximum extra cell interference.

3.2.1.2 Simulation Results

Twenty different random seed values are used to increase the statistical confidence, which results in 20 simulation runs for each configuration, i.e. for each curve. A short explanation of the concept of confidence intervals is provided in Appendix: Statistical Confidence, starting at page 129.

To cover low user numbers as well as high user numbers for each investigation a relatively high value for call arrival rate was chosen. This value is the same for all investigations except for the configuration investigating an increased cell radius of 1600 m. Using a high call arrival rate will result in a high number of users being active most of the time, but due to the randomness of the actual call arrivals the number of active users fluctuates, thus resulting occasionally also in situations with a low number of active users. Hence, the confidence intervals for results at small user numbers are slightly larger than for results at large user numbers, because there are more "samples" available at higher user numbers.

The confidence interval for 99 % confidence computes to about ± 4.5 % on average for all results, but for better readability it is not shown in the graphs.

3.2.1.2.1 Impact of Transmission Power

For investigating the impact of the maximum transmission power on the cell breathing effect three different maximum output powers of the base stations were configured: 10 W, 20 W and 40 W.

As to be expected the coverage of the cell (and the capacity, respectively) increases with increasing maximum transmission power

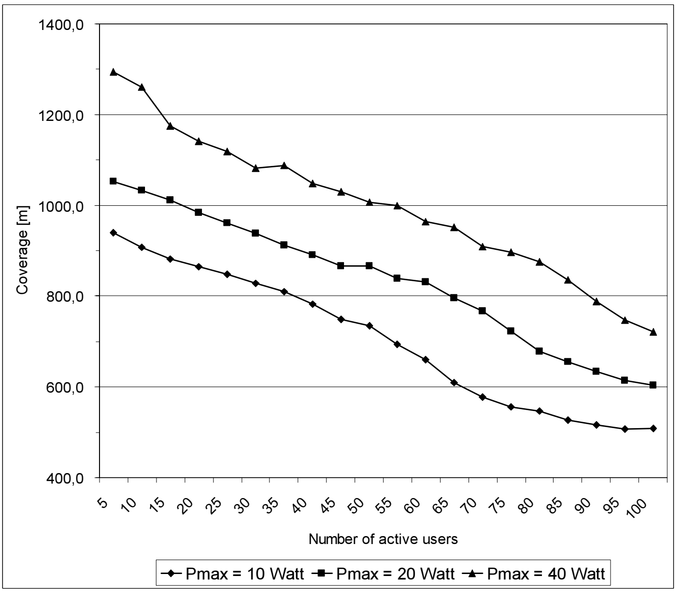


Figure 3.2: Impact of transmission power

(see Figure 3.2).

For coverage of 800 m (corresponding to the nominal cell radius) about 35 users could be supported at 10 W, ca. 60 users could be supported at 20 W and about 85 users could be supported at 40 W. The relative gain in coverage or capacity becomes smaller with increasing values for maximum transmission power.

3.2.1.2.2 Impact of Extra Cell Interference

For investigating the impact of the extra cell interference on the cell breathing effect three different interference levels were configured: maximum, maximum - 6 dB and maximum - 12 dB, where maximum defines the extra cell interference level where all extra cell base stations are transmitting at the same maximum transmission power of 20 W as the investigated center cell base station.

As to be expected the coverage of the cell (and the capacity, respectively) increases with decreasing extra cell interference (see Figure 3.3).

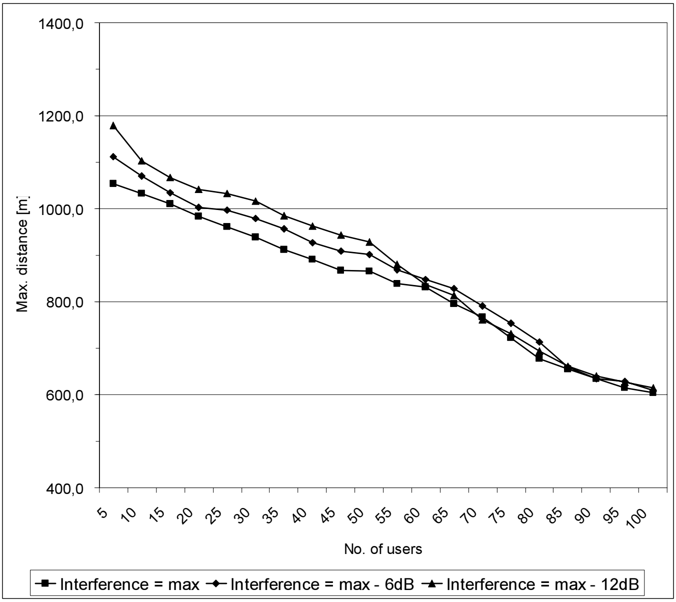


Figure 3.3: Impact of extra cell interference

3.2.1.2.3 Impact of Cell Radius

For investigating the impact of the nominal cell radius on the cell breathing effect three different nominal cell radii were configured: 400 m, 800 m and 1600 m.

For coverage of 800 m about 10 users could be supported at a nominal cell radius of 400 m, about 60 users could be supported at 800 m nominal cell radius and about 65 users could be supported at 1600 m nominal cell radius.

It is obvious that for this configuration (omni directional antennas, 20 W maximum transmission power) a nominal cell radius of 1600 m would be too large, because the coverage never achieves values exceeding even 1200 m (see Figure 3.4).

3.2.1.2.4 Impact of Data Rate

For investigating the impact of the user data rate on the cell breathing effect three different user data rates were configured: 8 kb/s, 16 kb/s and 32 kb/s. Also two different scenarios were used, one with 800 m and one with 400 m nominal cell radius.

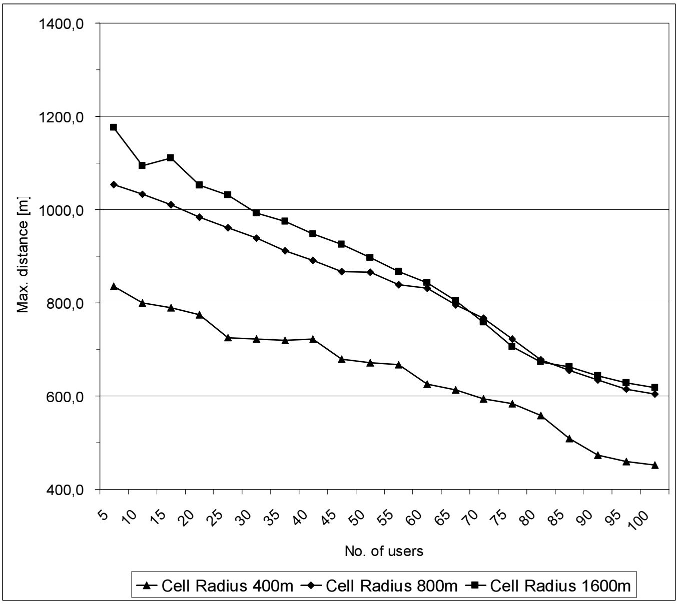


Figure 3.4: Impact of cell radius

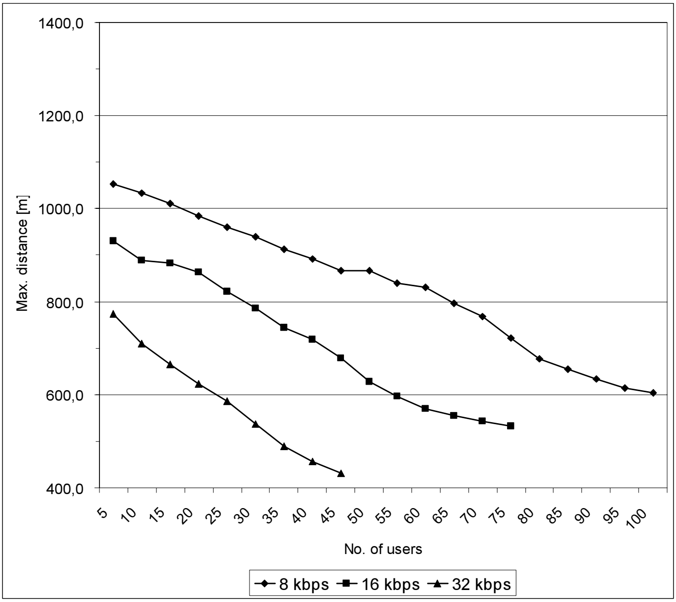


Figure 3.5: Impact of user data rate (r = 800 m)

As to be expected the coverage of the cell decreases with increasing user data rate (see Figure 3.6 and Figure 3.5).

For a cell radius of 800 m coverage of at least 800 m has to be provided (see Figure 3.5). Then about 60 users could be supported at a user data rate of 8 kb/s (= 480 kb/s total throughput), about 25 users could be supported at a user data rate of 16 kb/s (= 400 kb/s total throughput) and less than five users could be supported at a user data rate of 32 kb/s (< 160 kb/s total throughput).

For a cell radius of 400 m coverage of at least 400 m has to be provided (see Figure 3.6). Then more than 100 users could be supported at a user data rate of 8 kb/s (> 800 kb/s total throughput), about 75 users could be supported at a user data rate of 16 kb/s (= 1200 kb/s total throughput) and about 30 users could be supported at a user data rate of 32 kb/s (= 920 kb/s total throughput).

Increasing user data rate decreases the number of users that can be supported by the cell, since for each user more resources have to be allocated when the user data rate increases. Furthermore the user data rate has an impact on the total throughput of the cell that depends also on the CDMA processing gain.

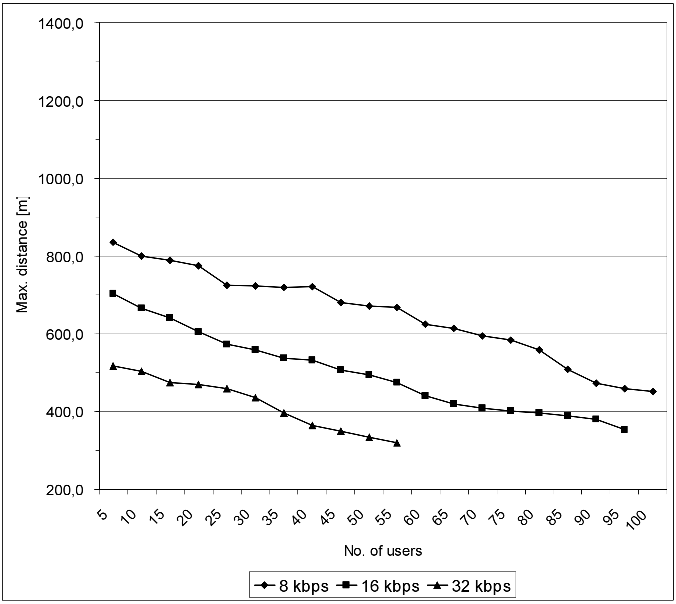


Figure 3.6: Impact of user data rate (r = 400 m)

If the cell is too large for the load desired then the impact of high data rates becomes even worse.

3.2.1.2.5 Impact of Soft Handover

For investigating the impact of soft handover on the cell breathing effect the two scenarios used have a nominal cell radius of 400 m and of 800 m. For soft handover a sufficient overlapping of adjacent cells is required, which is provided by the smaller cells but not by the larger ones (see section 3.2.1.2.3). Two different configurations were used: One with soft handover disabled as before, one with soft handover enabled. For the latter case the other cells must not be configured as “fully loaded by default” because otherwise soft handover actually would be impossible.

Hence the comparison of the maximum coverage is meaningless here: The coverage with soft handover enabled will never exceed the nominal cell radius significantly, because around the cell perimeter and beyond the adjacent cells become more attractive

r = 400 m	SHO: No	SHO: Yes
Average number of active users	98.83 (± 1.43)	129.02 (± 0.54)
Average TX power [W]	19.48 (± 0.01)	18.88 (± 0.03)

Table 3.1: Impact of soft handover (in brackets the 99 % confidence interval is given)

and distant users either will be accepted in an adjacent cell from the beginning or they tend to handover.

To get an idea of the potential impact of soft handover we compare the average number of active users and the average total transmission power required to support these users.

In Table 3.1 the results for this investigation are given for the smaller cells. It can be seen that the use of soft handover decreases the average transmission power slightly by about 3 % but increases the average number of active users by about 30 %. Hence the average transmission power per user is decreased by about 26 % from 197.1 mW (22.9 dBm) to 146.3 mW (21.7 dBm).

For high loads a nominal cell radius of 800 m does not provide a sufficient overlapping of adjacent cells (see Figure 3.2), hence soft handover does not improve capacity here (see Table 3.2).

Of course the improvement achievable depends on the actual algorithm used to implement soft handover. Here a simple algorithm based on a 3GPP proposal [27] with 2.5 dB hysteresis was used. For other algorithms the results may vary but the basic

r = 800 m	SHO: No	SHO: Yes
Average number of active users	84.92 (± 1.49)	81.41 (± 1.04)
Average TX power [W]	19.55 (± 0.01)	19.48 (± 0.01)

Table 3.2: Impact of soft handover in larger cells (in brackets the 99 % confidence interval is given)

behavior is expected to be the same because a sufficient cell overlap is required for any kind of soft handover.

3.2.2 Interpretation of "Cell Breathing" Results

The mutual dependency of cell coverage and cell capacity in fixed spectrum CDMA cellular systems has been investigated. Here, cell capacity C is defined as the number of active users per cell that can be supported at the same time for a given Quality of Service. The results show that for a given cell layout (i.e. base station placement) the maximum capacity is primarily determined by the nominal cell radius (coverage required) and by the maximum transmission power. Since legal and economical restrictions may define an upper limit for the latter, choosing the appropriate cell radius according to the expected load has been manifested to be crucial when doing the cellular network design, whereby the expected load has to be defined in terms of user number and user data rate.

The application of soft handover increases the capacity of a CDMA-based cellular system, thus mitigating the effect of cell breathing, but even then overlapping cells are required also. Hence, the interdependency of capacity and coverage still has to be considered.

More advanced systems, e.g. using directional or even smart antenna systems, will increase the absolute values for coverage and capacity respectively, but the mutual dependency of cell coverage and cell capacity will continue to exist. So the cell breathing effect has a major impact on the network design.

The starting point for the estimation of the capacity gain caused by relaying is based on the observation that the maximum cell capacity decreases as the cell radius increases.

Supposing a suitable distribution of mobile relays, the ad-hoc extension of cellular coverage can be used to reduce the radius that has to be covered by the cellular system thus increasing the capacity. The idea is illustrated in Figure 3.1. As shown in the figure, the cellular base station covers only the inner part of the cell with a radius r smaller than the nominal cell radius R. The area of the ring between this inner circle and the cell border is covered by mobile relays using the ad-hoc air interface.

Since the capacity C of a cell depends on the area to be covered we can define C as a function of the radius:

$$C(r) > C(R) \mid r < R \tag{4}$$

Hence, the relative capacity gain g can be defined as:

$$g = \frac{C(r) - C(R)}{C(R)} \tag{5}$$

Using Figure 3.5, capacity values (in terms of number of users) can be achieved for different values of the cell radius. For example, when 8 kb/s connections are considered, 60 users can be served in a radius of 800 m, whereas 85 users can be served within 650 m of the base station, i.e. $C(800\ m) = 60$ users and $C(650\ m) = 85$ users.

By supporting all mobile terminals at a distance between 650 m and 800 m using mobile relays based on e.g. WLAN, the UMTS coverage required the whole area is reduced to 650 m, thus increasing the number of users that can be supported at a distance of 800 m from the UMTS cell by about 41 % (see Table 3.3). For this example a WLAN range of at least 150 m has to be assumed. The principle is valid also for other ranges or other types of ad-hoc air interfaces, though.

It is assumed that a sufficient number of mobile terminals are located within the inner circle and can act as mobile relays, providing ad-hoc coverage to the mobile terminals within the outer ring. This assumption can be justified because in a real cellular system, idle terminals – which could potentially act as relays – are always much more numerous than active terminals. Mobile terminals in the outer ring can only act as mobile clients via WLAN, whereas terminals inside the inner circle can act as mobile relays but are not allowed to be served by other mobile relays.

For the actual assignment of mobile relays and mobile clients a selection algorithm has to be applied, which is out of the scope of this investigation.

The increase in capacity can be explained by the fact that ad-hoc relaying allows for a reduction of the downlink transmission power on the UMTS air interface. Additionally, by using "extra spectrum" for the ad-hoc links auxiliary resources have been added to the system.

Coverage	8 kb/s	16 kb/s	32 kb/s
450 m – 600 m	-	-	100 %
500 m – 650 m	-	-	100 %
550 m – 700 m	-	62 %	150 %
600 m – 750 m	42 %	66 %	300 %
650 m – 800 m	41 %	80 %	-

Table 3.3: Capacity improvement using ad-hoc extension (range of 150 m)

The observation has been extended to other distances and the results are summarized in Table 3.3. The evaluation is limited to 800 m, because this is the nominal cell radius of the system considered in the simulation.

To study the impact of different user data rates, the same procedure described before was applied. The results are also summarized in Table 3.3. For the same number of users the cell coverage decreases with increasing user data rate. This has also an impact on the capacity improvement. As it can be observed in the table, the relative capacity improvement is increasing with the data rate. For 32 kb/s up to 300 % capacity improvement could be achieved.

The investigation has been extended by assuming values of 100 m and 50 m respectively for the range of the ad-hoc air interface. The results are summarized in Table 3.4 and Table 3.5. As

Coverage	8 kb/s	16 kb/s	32 kb/s
450 m – 550 m	-	-	60 %
500 m – 600 m	-	-	50 %
550 m – 650 m	-	44 %	66 %
600 m – 700 m	33 %	25 %	100 %
650 m – 750 m	21 %	50 %	200 %
700 m – 800 m	25 %	60 %	-

Table 3.4: Capacity improvement using ad-hoc extension (range of 100 m)

expected reducing the distance covered by the ad-hoc extension reduces the capacity improvement.

From the investigation above it can be concluded that the capacity improvement achievable by ad-hoc extension of cellular is dependent on the average range the ad-hoc extension and on the user data rate.

One explanation for the latter factor is that the absolute values for coverage as well as for user number decrease with increasing data rates (since they require a higher transmission power to maintain the energy per bit ratio, hence causing a higher interference). Since the curves in Figure 3.5 are roughly parallel the absolute capacity improvement in terms of number of users is similar for all data rates. Therefore, the higher the considered data rates, the better the relative improvement in capacity.

The achievable improvement depends also on the configuration considered. For other configurations the results will vary but the basic behaviour is expected to be the same.

Coverage	8 kb/s	16 kb/s	32 kb/s
450 m – 500 m	-	-	33 %
500 m – 550 m	-	-	20 %
550 m – 600 m	-	30 %	25 %
600 m – 650 m	17 %	11 %	33 %
650 m – 700 m	13 %	12.5 %	50 %
700 m – 750 m	7 %	33 %	100 %
750 m – 800 m	16 %	20 %	-

Table 3.5: Capacity improvement using ad-hoc extension (range of 50 m)

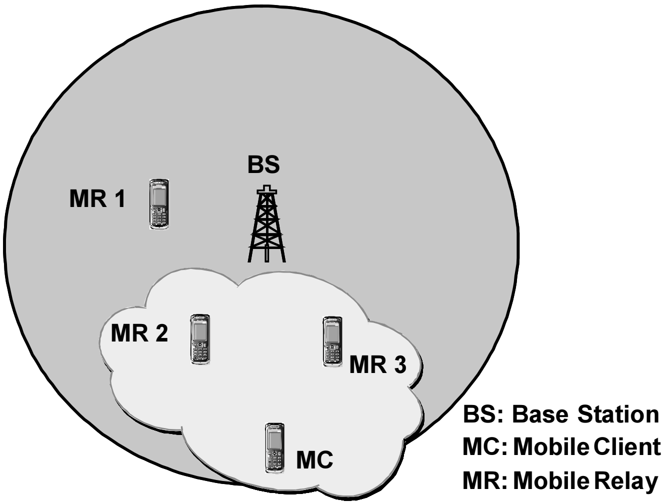


Figure 3.7: Scenario for the definition of signalling protocols

3.3 Signalling Protocols for Ad-Hoc Extension

In the previous sections the potential advantages of extending the cellular network by means of ad-hoc relaying have been discussed. However, current signalling mechanisms are not designed for the integration of ad–hoc and cellular networks, so for applying this extension to the UMTS system the existing signalling protocols have to be enhanced.

In this section one exemplary general signalling mechanism for finding the appropriate mobile relay during call setup is introduced. Several possible approaches have been developed and analyzed during this work (see e.g. [Hil_7]); here only an approach requiring cellular coverage for the mobile client but providing good power efficiency is presented. Power consumption – always being a critical issue for mobile terminals – is assumed to be even more important for potential mobile relays that have to spend part of their battery power for supporting the communication of somebody else. For the sake of simplicity the discussion is kept to an abstract level not considering UMTS-specific details.

3.3.1 Design Criteria and Scenario

A typical scenario as shown in Figure 3.7 is assumed where the mobile client (MC) wants to initiate a connection e.g. for a data service. Hence, a decision has to be taken whether to take the direct cellular link or to use an ad-hoc relay. In this investigation the network is supposed to take the decision, although other approaches would be possible, too. For taking this decision knowledge of the quality of all the potential links is required in order to estimate which way of connecting the mobile client to the cellular network would be the best, whereby the actual definition of "best" may include - depending on the operator's policy - overall system performance as well as the quality of service provided to the particular user. Also, there are several ways for determining the quality of a link; here all values being necessary (e.g. received signal power, signal to interference ratio, bit error rate, etc.) are supposed to be contained in an abstract data set named link performance parameters (lpp).

Basically there are three steps: During call setup the network has to (1) collect the link performance parameters of all potential links, and then it has to (2) select the best link or in case of relaying the best combination of one cellular and one ad-hoc link, and finally it has to (3) inform the mobile client and the mobile relay accordingly.

The second step is not a matter of protocol mechanisms, but rather a question of designing an appropriate decision algorithm, so it is not considered here. For the first and the third step in the following sections there are some possible approaches proposed.

3.3.2 Collection of Link Performance Parameters

Basically there are three types of links to be analyzed during the call setup process:

- The cellular link between mobile client and base station (MC-BS) and vice versa
- The cellular links between potential mobile relays and base station (MR-BS) and vice versa
- The ad-hoc links between mobile client and potential mobile relays (MC-MR)

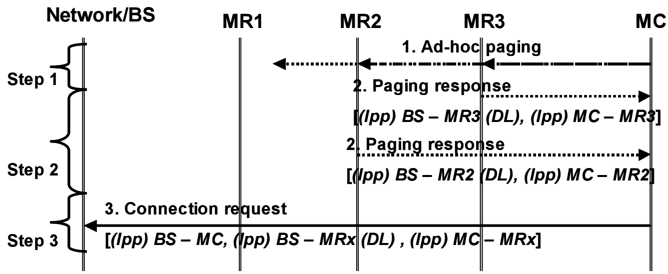


Figure 3.8: Collection of information at the network side

Due to the use of frequency division duplex (FDD) in the cellular system the uplink and downlink have different characteristics, whereas ad-hoc systems typically operate in simplex mode, i.e. ad-hoc systems typically use the same frequency for transmitting and receiving, so for this investigation the radio link between two ad-hoc terminals is assumed to have similar – if not same – characteristics in both directions. Consequently, for the potential cellular links the link performance parameters have to be collected for uplink as well as for downlink while for the potential ad-hoc links the link performance parameters have to be determined only for one direction.

For the collection of information at the network side, a mechanism is necessary to identify potential mobile relays within the ad-hoc range of the MC (e.g. MR2 and MR3 in Figure 3.7). This mechanism could be based on the use of cellular paging, ad-hoc paging from the mobile client or ad-hoc beaconing from the mobile relays.

One criterion to be considered for the design is whether the approaches are working if the MC is within cellular coverage or if the MC has no cellular coverage. Here cellular coverage is supposed to exist.

When all the necessary information has become available to the cellular network, the selection algorithm is applied in order to make a sound decision between the different possible links, including the direct cellular link.

If a direct cellular link is selected for the connection, the call set-up will continue using standard cellular protocols. Otherwise, if the best option found is the use of a relay connection, the MC and the MR have to be configured accordingly.

In Figure 3.8 a possible approach for collecting the information on link qualities is depicted with relevant parameters carried by each message being specified in square brackets, cellular signalling being represented by continuous lines, and ad-hoc signalling being represented by dotted lines.

The first step of this signalling approach is paging performed by the mobile client on its ad-hoc air interface. The potential mobile relays reply to the paging by sending the link performance parameters of their cellular and ad-hoc links to the mobile client. Finally the mobile client sends a connection request to the base station, i.e. to the cellular network, that contains the information collected on the link qualities. Now the network would be able to take a sound decision whether to use a mobile relay and - if necessary - which mobile relay to choose.

The initial paging is received only by the mobile terminals being within the ad-hoc range of the mobile client. Hence, only those mobile terminals are affected that actually would be able to serve as mobile relay. This is a major advantage of this approach compared to other approaches that make use of e.g. cellular paging, by which all mobile terminals in a cell would be affected, because simply based on cellular paging the mobile terminals being in the ad-hoc range of the mobile client could not be determined. Hence, in this case a significantly higher number of mobile terminals would have to react to the paging, thus wasting their own battery power for transmitting as well as wasting resources of the mobile client or the network for receiving and evaluating numerous – and mostly needless - messages. For further details refer to [Hil_7].

3.4 Estimation of User Density Required

Capacity as well as coverage of a cellular system could be increased by utilizing multimode terminals as mobile relays to provide connectivity to mobile clients, as it has been discussed above and also by several publications (see e.g. [7], [21], [28]). Basically each user should potentially serve as a mobile relay for other users and in turn should be able to act as a mobile client using another terminal as mobile relay. Although this mechanism could be based also on multi-hop relaying we limit our considerations to single-hop relaying (see Figure 3.1).

This approach is assumed to work sufficiently provided that there are a high number of mobile terminals that could act as mobile relays.

If the number of mobile relays is too small, a MC may not be able to find a MR that can relay the connection so the MC either would have to switch back to a direct cellular connection or – if this was impossible for some reason – the connection could not be sustained. In the first case the relaying would not increase the system performance as desired whereas the second case would result in the service request being rejected ("blocking") or in the ongoing communication being interrupted ("dropping"), depending on whether it happens at session setup or when a connection already had been established. Hence, the user density is an important parameter when investigating solutions for the extension of cellular systems by use of mobile relays. Consequently this section introduces an approach to roughly estimate the user density required based on the range of the ad-hoc air interface and the service availability desired.

A further enhancement to this approach could be to consider the MR to be a gateway into a pure ad-hoc network, i.e. to serve clients that do not have a cellular air interface. As a result the pure ad-hoc network would achieve a – potentially more direct and reliable – connection to the global internet through the cellular system. Although the investigation of this enhancement is not within the scope of this thesis, the results presented here could be applied analogously.

This section is organized as follows. After this introduction the probability of finding a mobile relay is calculated in section 3.4.1. In section 3.4.2 the frequency of handovers is estimated. In section 3.4.3 the user density required for this approach to work sufficiently is derived and finally a conclusion is provided in section 3.4.4.

For the following considerations the mobile terminals are supposed to have an identical and constant velocity as well as individual but constant directions.

3.4.1 Probability of Finding a Mobile Relay

Based on user density and ad-hoc range we will calculate the probability of having at least one potential MR available within the

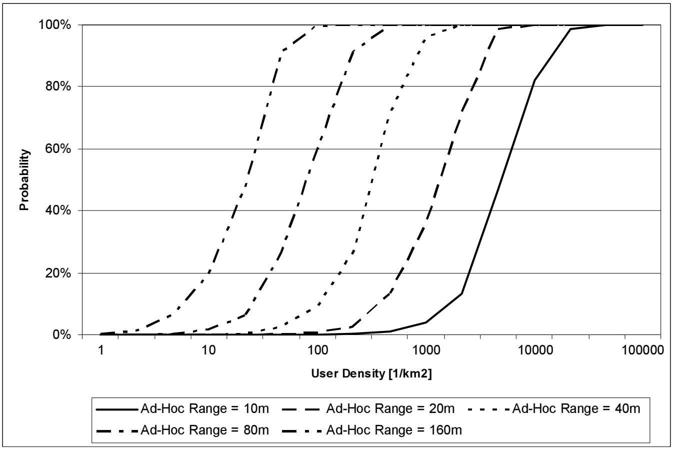


Figure 3.9: Probability of finding a MR according to Equation (7)

range of the ad-hoc air interface. For this we assume that at any time instant the users' positions are randomly but uniformly distributed over the area and that each user could serve as a mobile relay.

A user density expressed in terms of users/m^2 is a mean value, hence expressing a probability to have a mobile terminal being placed inside a particular square meter.

Based on ad-hoc range the coverage of the MC is a certain number of square meters. This can be considered as a random experiment performed for each square meter with probability of success being equal to the user density. An exact model for this would be the Hypergeometric Distribution, which describes "sampling without replacement", but since the number of "samples" is small compared to the total "lot size" the Binomial Distribution was chosen as a more convenient but still valid approximation [29].

We use the Binomial Distribution to model the probability *P(n)* of exactly *n* mobile terminals being in a certain area as given by Equation (6):

$$P(n) = \binom{N}{n} \cdot p^n \cdot (1-p)^{N-n} \tag{6}$$

With N *being the "number of samples", which corresponds to the area [m^2] covered by ad-hoc,* n *being the "number of successful samples", i.e. the* *number of mobile terminals being inside this area and* p *being the "probability of success", which corresponds to the user density [users/m²].*

A MC being connected to the cellular network by using a MR obviously needs this MR being located within the range of its ad-hoc air interface. Consequently, in the area covered by the MC's ad-hoc interface there have to be always at least two mobile terminals, i.e. the potential MR(s) as well as the MC itself.

Hence, we need to calculate the probability of at least two mobile terminals being inside the area covered by ad-hoc (see Equation (7)).

$$p_{MR} = P(n \geq 2) = 1 - P(n=1) - P(n=0) \tag{7}$$

With p_{MR} *being the probability of having at least one MR available.*

The results based on varying values for the ad-hoc range are depicted in Figure 3.9.

Here the term user density describes how many users per area are potentially available, i.e. are either active or in a stand-by mode of operation and could act as MR or MC at any time.

According to [30], urban UMTS macro cells are expected to support user densities of e.g. 68 users/km². Assuming several cells covering any given place due to the presence of several operators then a user density of e.g. 300 users/km² could be assumed for urban areas. In this case even with a relatively short range of the ad-hoc air interface a high probability of finding a MR could be achieved (see Figure 3.9).

3.4.2 Estimation of Handover Frequency

For this investigation we are not interested in the handovers happening in the cellular part of the system but only in handovers that are generated by switching between different MRs. If a MC is supposed to use only relayed connections via MRs and if there are MRs sufficiently available then the number and frequency of handovers is determined by the duration of the ongoing communication, by the range of the ad-hoc air interface and by the movement patterns of the MC as well as of the MRs.

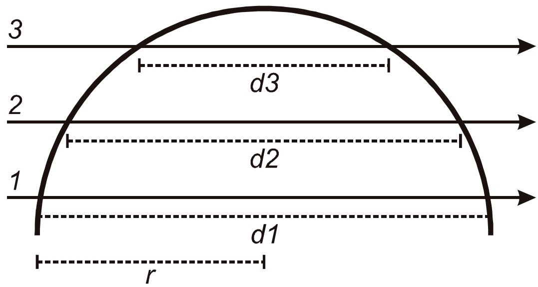


Figure 3.10: Distances when crossing a circular area

For this estimation it is assumed that there is always free capacity on the ad-hoc air interface to enable relaying and that the handover algorithm is reactive, i.e. that the switching from the current MR to the next MR would be triggered when the current link between MC and MR is broken.

3.4.2.1 Mobility

Simple user mobility is assumed where the users move with the same constant speed and individual but constant directions. As a special case, a stationary MC is also investigated.

The time a particular MR can offer relaying services to a particular MC depends on the time that the MR is within the ad-hoc range of the MC.

Assuming a stationary MC, this time would depend on the time needed by the MR to cross the area that is covered by the MC's ad-hoc air interface. If the MR is moving along the fringe of this area then the resulting time would be small or even close to zero, whereas if the MR is moving full centre through the area then the time depends on the MR's velocity and on the ad-hoc range, as given by Equation (8).

When a mobile terminal is crossing an area with circular shape then the distance covered inside the area depends on the radius of the circle and on the distance of the trajectory to the centre point of the circle (see Figure 3.10).

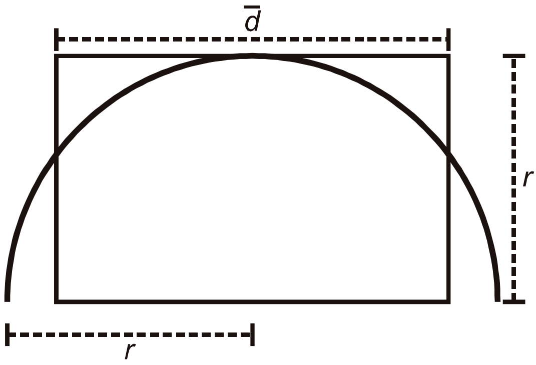


Figure 3.11: Mean distance based on area

$$t_{avail} = \frac{d}{v} \begin{cases} = \frac{2r}{v} & at\,center \\ \cong 0 & at\ fringe \end{cases} \tag{8}$$

With t_{avail} *being the time that a MR is available to the MC,* d *being the distance that the MR moves within the coverage of the MC,* r *being the range of the ad-hoc air interface and* v *being the velocity of the MR.*

For determining the mean distance that is covered inside the circle by using random trajectories we use an oblong with the same area as the according half circle (see Figure 3.11), which gives us Equation (9):

$$\frac{1}{2}\pi r^2 = \bar{d}\, r \Leftrightarrow \bar{d} = \frac{1}{2}\pi r \tag{9}$$

Since the area covered by the ad-hoc interface is assumed to have circular shape and since it can be crossed by the MR at a random distance to the MC the mean value is considered for d, which results in Equation (10):

$$t_{avail} = \frac{r}{v}\frac{\pi}{2} \tag{10}$$

When the trajectories of two mobile terminals cross each other they enclose the angle α (see Figure 3.12), which is defined by Equation (11):

$$\alpha = \varphi_1 - \varphi_2 \tag{11}$$

With φ_1 being the direction of mobile terminal 1 and φ_2 being the direction of mobile terminal 2.

For this investigation we are interested in the point where the two converging trajectories get close enough to allow ad-hoc communication between the two mobile terminals and in the point where the two diverging trajectories exceed this distance after crossing. The time needed to move from the first point to the second point is the time t_{avail} when ad-hoc communication can happen, i.e. when a potential MR could provide service for a potential MC, because the distance between the two mobile terminals does not exceed the ad-hoc range.

For any two trajectories with $0<\alpha<\pi$ we can place a coordinate system as depicted in Figure 3.12. Then we can calculate the time of availability t_{avail} of the potential MR as given by Equation (12):

$$t_{avail} = \frac{d}{v} = \frac{\sqrt{dx^2 + dy^2}}{v} = \frac{\sqrt{dx^2 + r^2}}{v} \tag{12}$$

The distance dx covered in x-direction is given by Equation (13):

$$dx = \frac{r}{\tan\left(\frac{\alpha}{2}\right)} \tag{13}$$

From Equations (12) and (13) the final result is obtained as given by Equation (14).

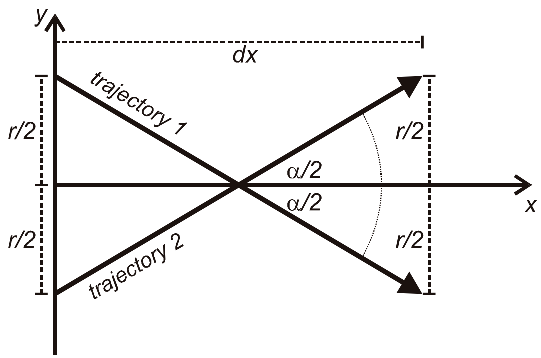


Figure 3.12: Trajectories of two mobile terminals

$$t_{avail} = \frac{r}{v}\sqrt{1+\frac{1}{\tan^2\left(\frac{\alpha}{2}\right)}} \qquad with\ 0<\alpha<\pi \tag{14}$$

If both mobile terminals, MR and MC, were moving with the same constant velocity but individual directions then the time for which the MR could provide relaying services to the MC is given by Equation (14).

Assuming the directions of movement being uniformly distributed among the mobile terminals a mean value for α of $\pi/2$ is considered, which results in Equation (15):

$$t_{avail} = \frac{r}{v}\sqrt{1+\frac{1}{\tan^2\left(\frac{\frac{\pi}{2}}{2}\right)}} = \frac{r}{v}\sqrt{2} \tag{15}$$

The results of Equation (15) are depicted in Figure 3.13.

With velocities of 50 km/h and above the average serving time of a MR is approximately 10 s or less, depending on the ad-hoc range.

3.4.2.2 Traffic

The traffic models proposed by [31] have been considered, which result in a call mean duration of 120 s for circuit switched voice services and in a mean session duration of e.g. 330 s for packet switched data services, depending on the user data rate.

3.4.2.3 Conclusion

Based on the estimations expressed by Equations (10) and (15) it can be concluded that the average time that a MR may be able to provide relaying services to a MC is increased if the MC is stationary compared to the case where the MC is moving with same speed as the MR. Nevertheless, for this investigation mobile relays and mobile clients, respectively, are supposed to be moving.

The average handover frequency per call is the inverse of the average time that a MR actually provides relaying service to a MC. This time is less then the time for which the MR potentially could provide the service, because a mobile terminal does not start acting as MR as soon as it enters the ad-hoc coverage of the MC since the current MR is used by the client as long as possible (reactive handover). Only if the current MR cannot be used any more then the next MR can be selected. Then this MR may have moved a distance since it entered the coverage of the MC, thus the remaining time in which the MR can provide service is reduced.

Since the selection of a MR can happen any time the MR is within the coverage of the MC, it can be assumed to happen on average when the MR has passed halfway through the area covered by the ad-hoc air interface. This assumptions also conforms to the fact that MRs that are close to the MC probably will be preferred when selecting a new MR because of the short distance and the resulting good radio conditions.

So the mean time of service $t_{service}$ for one MR is given by Equation (16):

$$t_{service} = \frac{t_{avail}}{2} = \frac{r}{v}\frac{\sqrt{2}}{2} \tag{16}$$

Then the mean number of handovers (switches between different MRs) h that a user experiences during a call is given by Equation (17).

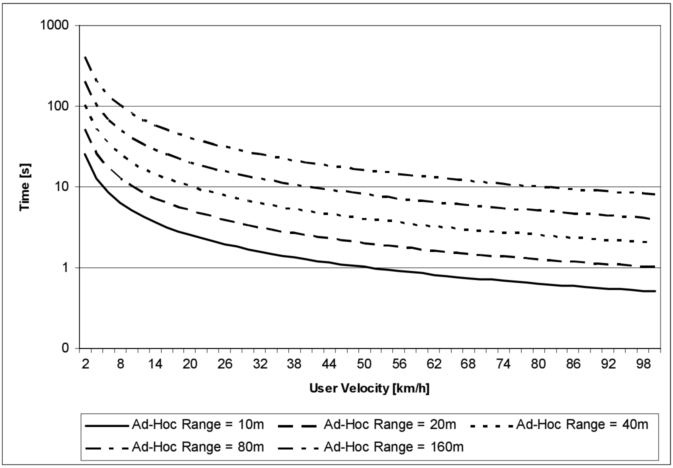


Figure 3.13: Average time that a moving MR could provide relaying service to a moving MC according to Equation (15)

$$h = \frac{t_{call}}{t_{service}} = t_{call} \frac{v}{r} \frac{2}{\sqrt{2}} \tag{17}$$

With t_{call} *being the mean duration of a call (voice) or session (data). The results of Equation (17) are depicted in Figure 3.14.*

With pedestrian velocity the average number of handovers per call is below 10. With increasing velocity the number of handovers increases as well, which may render this approach not feasible at high user speeds, depending on the actual systems and the actual applications.

3.4.3 Estimation of User Density Required

The user density required depends on the desired service availability υ, which is expressed by Equation (18):

$$\upsilon = p_{MR}{}^{h} \tag{18}$$

With p_{MR} *being the probability of having at least one MR available and* h *being the mean number of handovers per call.*

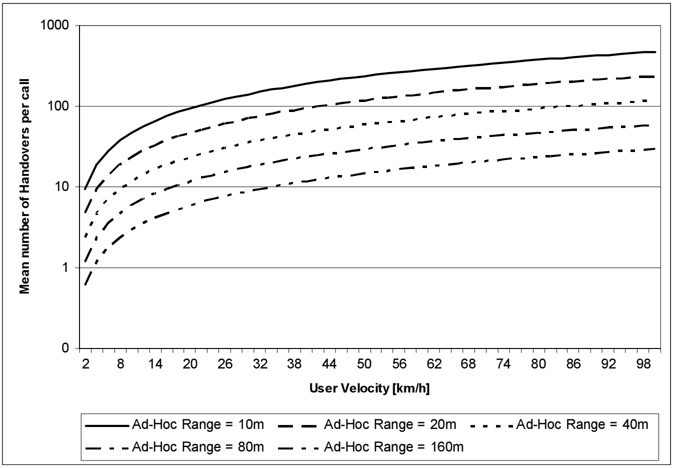


Figure 3.14: Mean number of handovers for a voice call (mean duration 120 s)

For a service availability the user density required can be calculated based on Equations (7), (17) and (18). Example results are depicted in Figure 3.15 for call mean duration of 120 s and a user speed of 3 km/h.

To achieve 95 % service availability with user speed of 3 km/h, ad-hoc range of 80 m and call mean duration of 120 s a user density

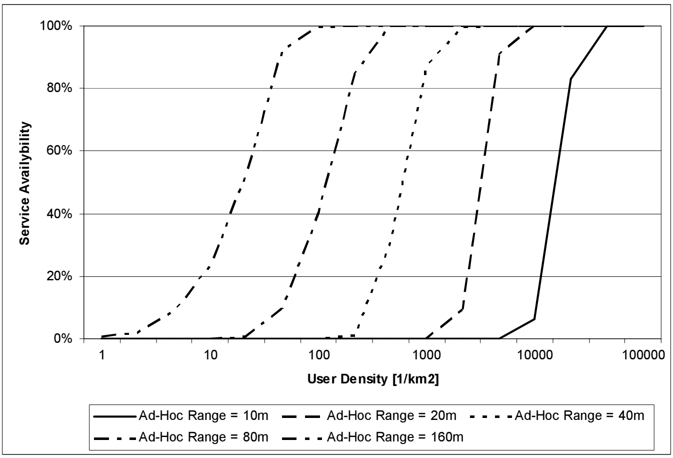


Figure 3.15: Service availability resulting from different user densities

of 270 users/km^2 would be required. With a session mean duration of 330 s a user density of 328 users/km^2 would be required.

3.4.4 Summary

The extension of cellular systems by mobile relays utilizing an ad-hoc type of air interface is a possible approach to enhance system performance or to increase connectivity. One prerequisite for this is the sufficient availability of mobile relays to achieve the dropping probability desired by the operator.

In this section a rough estimation of the user density required to realize this goal has been given. For this estimation, simplifying assumptions have been presumed, but the results are considered to give a dependable indication on the user density required.

The results show that user densities of ca. 300 users/km^2 - as they may be found in urban areas or may be even exceeded in city centres - suffice to meet this requirement.

Since in those areas the users' positions are not uniformly distributed over the whole area but are concentrated along streets, places or shopping malls even lower mean user densities may lead to satisfying service availability. Additionally in a realistic scenario the cellular system itself could provide a direct link to a mobile client in case there was no mobile relay available, thus the user density required could be further decreased.

This estimation has shown that it is reasonable to assume that mobile relays are sufficiently available if mobility of pedestrian type can be supposed.

3.5 Conclusion

In this chapter the extension of the cellular system by means of ad-hoc relaying has been discussed. An estimation based on simulation results has shown that significant improvements of the cellular system's capacity as well as coverage could be achieved. Load balancing between adjacent cells as a further possible benefit has been mentioned but was not in the focus of this discussion.

Since today's cellular systems are not designed to support this extension, a possible approach to endorse the existing cellular signalling mechanisms was introduced. Furthermore, based on an analytical estimation the user density required to enable this extension has been determined.

Thus not only the possible gains but also the feasibility of the extension of the cellular system by means of ad-hoc relaying has been shown.

4 Multi Standard Radio Resource Management

In Europe, 2nd Generation mobile communication networks including GSM as well as 2.5G systems like GPRS have achieved coverage close to 100 %. At the same time commercial wireless LAN installations based on 802.11 standards show up at hot spot locations in growing numbers while UMTS, the European 3G system, is being deployed.

As a result a heterogeneous environment is created where potentially several radio access technologies could serve a user's request. To benefit from this plurality the existing radio resource management has to be complemented by multi standard radio resource management (MxRRM) mechanisms that exploit the existence of alternative radio access technologies to improve service availability and quality of service.

This thesis describes fundamentals for successful MxRRM considering circuit switched as well as packet switched services and discusses some possible algorithms including performance results.

In section 4.1 the general concept of MxRRM is introduced. For enabling MxRRM the individual RATs involved have to be coupled for exchanging information on the current system state as well as for supporting seamless inter system handovers. Possible ways of coupling with respect to the current capabilities and architectures of UMTS, GSM and WLAN are presented in section 4.2.

One challenge for applying MxRRM is that knowledge is required on the RATs that are currently available to a mobile terminal. The availability depends on two issues, on coverage and load. A cell could already be full and thus not be able to support a mobile terminal, although the mobile terminal is within the coverage of the cell. Hence, despite coverage being a necessary prerequisite for establishing a communication link, coverage alone cannot answer the question of availability. In section 4.3 one possible approach to determine the RATs that are currently within the range of the mobile terminal is introduced. Algorithms for evaluating the current load are discussed in section 4.4.

In section 4.5 an analytical estimation of the possible performance improvements gained by MxRRM for circuit switched services is provided.

4.1 General Concept

The near future of mobile wireless communication will maintain or even increase the diversity of systems that a potential user may choose to avail. From today's point of view the systems most likely to dominate the European market are cellular systems based on 2G and 2.5G technology, namely GSM / GPRS / EDGE, and increasingly also systems based on 3G technology, namely UMTS, as well as more Ethernet-like systems based on wireless LAN technology, namely the IEEE 802.11 family.

In the current version these systems act independently, offering only a kind of "blind" handover between GSM and UMTS in case a call would be lost. A mature overall radio resource management is not included yet; future implementations based on UMTS Release 5 or later will probably provide an interface for the exchange of load information messages without specifying how to exactly utilize this information [8], [32], [33].

Depending on coverage, user preferences, application availability and last but not least business models and pricing the utilization of the individual systems may be very different. As a consequence the operators may experience decreased revenue while the customers do not get the best service they could possibly get. A solution to mitigate this impairment could be to couple these largely independent systems with a mechanism to commonly administrate their utilization. The fundamental idea is to keep the existing radio resource management (RRM) of each system to control the actual allocation of channels, time slots, spreading codes, etc., but to provide assistance in special situations like call setup, handover or overload (potential blocking or dropping). This assistance basically would mean to determine whether it would contribute to the overall system performance to redirect a call to an alternative system (to an alternative radio access technology, RAT). This mechanism has been named as Multi Standard Radio Resource Management (MxRRM). Some publications also use names like Common ~ (CRRM) [34] or Joint Radio Resource Management (JRRM) [35], [36].

To give a simplified example: Assume a call currently located in UMTS cell A is moving to cell B. The call has to perform a handover to cell B but for the sake of this example cell B is supposed to be fully loaded and therefore cannot accept the call. As soon as the current cell A cannot continue to support the call any longer the call would have to be dropped. Applying MxRRM it may be possible to determine a GSM cell that still has capacity available and will accept the call. So an inter system handover will be performed thus enabling the system to continue to satisfy the user demands as well as to generate further revenue for the operator.

The idea of MxRRM can also be extended to more advanced features like finding the system that is currently best suited to satisfy a particular service request. For circuit switched (CS) calls that have a guaranteed quality of service (QoS) this may result in reduced power consumption or less interference being created by the call, whereas for packet switched (PS) calls this may enable higher user data rates and may lead to an actual improvement of QoS.

Additionally, for different services requiring different QoS, systems based on TDMA and CDMA yield different performance results [37], [38]. From this it can be concluded that, having a set of different communication systems with different RATs, for a given service request there is a most appropriate system. Hence, assigning the service request to this appropriate system instead of assigning it randomly will increase overall system performance. Also assignment of micro/macro cell based on velocity in hierarchical cell architectures shows that assignment of resources, based on the individual characteristics of the call and on the current system state or on the capabilities of particular sub system parts, can improve performance [39].

In [40] the migration to 3G communication systems and beyond is discussed particularly considering QoS. The authors expect different radio access networks based on different radio access technologies - like UMTS, GSM, GPRS, EGPRS, WLAN - being connected to the same core network, thus sharing the core network's resources. This paper does not provide an approach for a common management of the different air interface resources, though.

Also with focus on QoS issues in a heterogeneous environment, the authors of [41] propose a "terminal interworking unit" to be involved in all inter system handover procedures. For an IP level QoS signalling protocol they propose "wireless hints" for describing the QoS required by a particular application. In their approach the inter system handover would be triggered by the mobile terminal mainly based on radio link quality and possibly also on user-perceived QoS.

Although these papers do not discuss MxRRM or related algorithms, they confirm the whole purpose of MxRRM in the way that they describe future communication systems and architectures, which not only would benefit from MxRRM, but which would also be capable of realizing it.

An approach complementary to MxRRM is the idea of dynamic spectrum allocation (DSA) as it is proposed e.g. in [42]. With this approach, in a heterogeneous environment the different RATs basically share one common pool of spectra and the spectrum being assigned to a particular RAT at a particular time and location is adapted according to the current load situation. Although this approach can improve the overall system performance, it is not part of the investigation presented here.

In [34] the idea of MxRRM is discussed using the term "Common Radio Resource Management (CRRM)". Simulation results are presented for circuit switched services indicating no significant capacity gain obtained by CRRM, particularly in the case of only two systems. The evaluation considers only blocking and traffic load, but neglects other measures like e.g. dropping. For packet switched traffic a typical capacity gain of less than 10 % was obtained. The simulator being used is based on a simplified system model and does not model e.g. the radio path or the interference situation.

In the context of the IST TRUST project the idea of MxRRM is termed "Joint Radio Resource Management (JRRM)" [35]. The scenario being considered here consists of the two systems UMTS in FDD mode and UMTS in TDD mode, where the TDD part is operated in congestion. The results presented show a capacity gain of up to 17.4 % for speech service only. Further investigations with respect to software downloading for SDR terminals lead to the result that JRRM has a positive impact on system performance. A continuation of this work is performed in the IST SCOUT project, with results being presented in [43] showing the same tendency for speech services with 12.2 kb/s. Additionally, a general framework for intersystem handovers was proposed in [44]. In 2004 the new EVEREST (Evolutionary Strategies for Radio Resource Management in Cellular Heterogeneous Networks) project has been started, which purposes to investigate strategies and algorithms for access and core networks for optimized utilization of radio resources within heterogeneous networks beyond 3G [45], [46]. Until now, no results from EVEREST are publicly available, though.

This thesis is intended to discuss some basic considerations on MxRRM and to present some possible MxRRM strategies as well as performance results. For this investigation two different radio access technologies are considered, i.e. UMTS using wide band code division multiple access (WCDMA) and GSM / GPRS (including EDGE features for packet switched services) using time division multiple access (TDMA), but the concepts and ideas presented here may also apply to a broader field covering also other radio access technologies.

Multi Standard Radio Resource Management for circuit switched services is expected to increase the overall performance of the combined systems, i.e. to increase capacity and service availability. Here, capacity is measured by the maximum amount of traffic (measured in Erlang) that can be offered to the system per cell area while maintaining a given minimum service availability, and service availability is measured by the proportion of calls that have been accepted and completed successfully. The term “cell area” is defined by the authors as the area covered nominally by one cell according to the network topology (i.e. base station distance). Since base stations of different radio access technologies may be co-located at the same position a particular cell area could actually be covered by several cells. The use of the measure “capacity / cell area” instead of “capacity / cell” is according to the fact that the idea of MxRRM is to improve the overall system performance, not only the performance of an individual radio access technology.

In the following text the service availability is expressed by blocking rate and dropping rate, which leads to a more detailed picture. “Blocking” describes calls that have been rejected immediately at the call setup whereas “dropping” describes calls that originally have been accepted by the system but cannot be supported any more and thus have been terminated by the network before the calls were completed successfully. Since dropping is considered to be more annoying for a user than blocking it is reasonable to distinguish between both.

When combining two pools of resources the resulting capacity is greater than simply the sum of the capacities of the individual resource pools. This effect named trunking gain is well known and can be proved analytically e.g. by the Erlang formulas. Faced with the constraints of real technical systems there may be problems in actually achieving the theoretically possible trunking gain. Hence, to develop MxRRM strategies that utilize most of the potential trunking gain is a desirable goal.

To further increase the gain of MxRRM differences between the RATs involved could be exploited. As mentioned above different RATs may have a different efficiency for different services. Thus, in a mixed service scenario the overall performance could be improved by selecting the most appropriate RAT for a particular service request.

Another approach could be to evaluate the radio path. Due to the current radio conditions one RAT could be preferable to the others. Even with identical base station positions this might happen because of different air interface features. For example, in UMTS a voice user close to the base station would consume a small part of the cell's resources, i.e. some milliwatts of transmission power, whereas the same user in GSM would always allocate a complete time slot independently of current radio conditions.

For packet switched services similar considerations apply. Since PS here means a "best effort" policy, these calls always can be accepted, thus a service availability of 100 % can be achieved, i.e. blocking or dropping does not occur for PS calls. As a result service availability would not serve well as a measure for system performance when investigating packet switched services. Contrary to CS calls, that have a guaranteed QoS with constant data rate, the PS calls experience varying data rates. Hence, the mean data rate per user or per completed packet call is considered as a good measure of system performance. Multi Standard Radio Resource Management for packet switched services is expected to increase the mean data rate as well as the overall system capacity.

An additional point of interest in this study is the impact MxRRM might have on the system by e.g. increased signalling load, which here is derived from the number of inter system handovers or directed retries.

4.2 Coupling of Radio Access Technologies

For enabling MxRRM different radio access technologies have to be coupled to allow for the exchange of information as well as for inter system handovers. In the following sections the features relevant for MxRRM of the possible UMTS-GSM coupling provided by Release 99 and Release 5 are discussed and possible levels of UMTS-WLAN coupling are compared.

4.2.1 Coupling of UMTS and GSM

Within the 3rd Generation Partnership Project (3GPP) [3] Multi Standard RRM has been investigated within a study item titled *"Improvement of RRM across RNS and RNS/BSS"*. The investigation of a radio resource management across RNS (Radio Network Subsystem) and BSS (Base Station Subsystem) carried out for this study item had proven that MxRRM improves the throughput of the network and reduces the blocking rate [34].

Since an MxRRM is only feasible if the RRM module is provided with the needed information from the adjacent cells, interfaces are required to exchange the data. Within 3GPP two solutions for Multi Standard RRM had been discussed, one solution proposes a centralized server for MxRRM and the other solution uses an MxRRM that is contained within every RNC/BSC (integrated solution). A decision has been taken to specify the required interfaces for the integrated solution within Release 5 and to postpone the centralized solution to Release 6. Consequently, there are no particular interfaces or mechanisms available within Release 99 to support MxRRM.

The solution found for the integrated approach contained some new information elements (IE) within existing signaling procedures of the RANAP- (Radio Access Network Application Part) and RNSAP-protocol (Radio Network Subsystem Application Part protocol) [8]. Due to these IEs the information concerning the current load of the cells can be exchanged between RNCs as well as between RNC and BSC, which enables the MxRRM-module within the RNC/BSC to take the load of adjacent cells into account while taking decisions regarding call admission or handover.

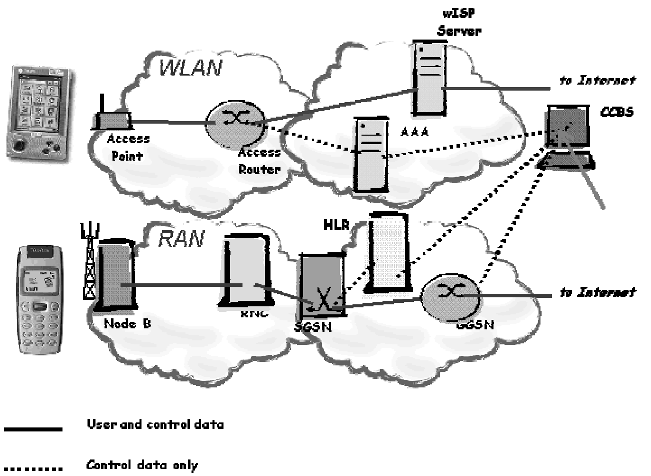


Figure 4.1: Open Coupling UMTS-WLAN (from [49])

4.2.2 Coupling of UMTS and WLAN

Cellular systems and WLANs should be considered as complementary systems: cellular systems could provide universal coverage and high mobility support while WLANs will be applied in hot spot areas offering high data rates.

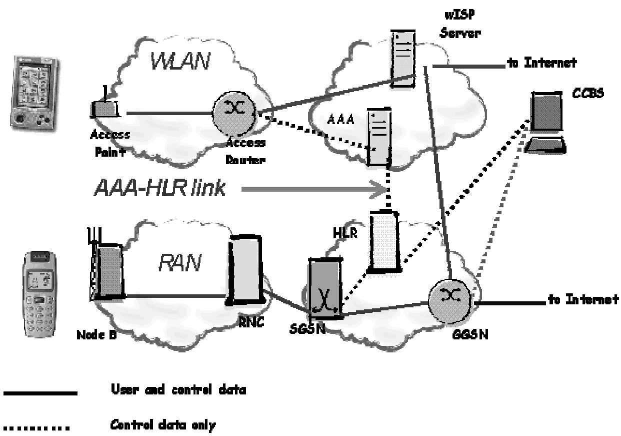


Figure 4.2: Loose coupling UMTS-WLAN (from [49])

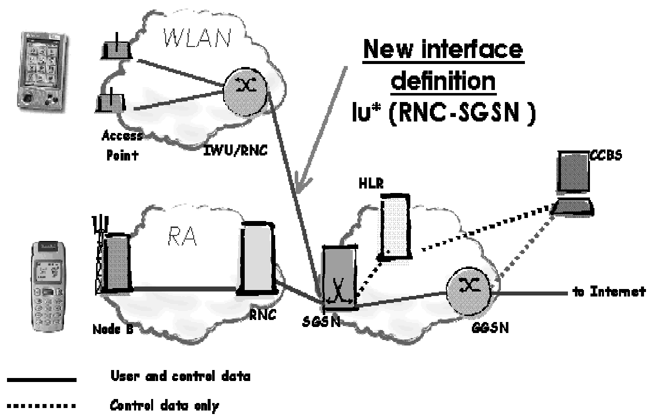


Figure 4.3: Tight coupling UMTS-WLAN (from [49])

WLANs according to IEEE 802.11a/b standard ([9], [4]) can be operated either in infrastructure-based or ad-hoc mode. The latter one could be exploited for a cellular extension by means of ad-hoc relaying, as discussed in chapter 3. For the considerations in this section a WLAN is assumed operating in infrastructure-based mode, providing internet access and having equipment required for commercial operation, like e.g. for authentication, authorisation and accounting (AAA).

The coupling between WLAN and UMTS can be implemented

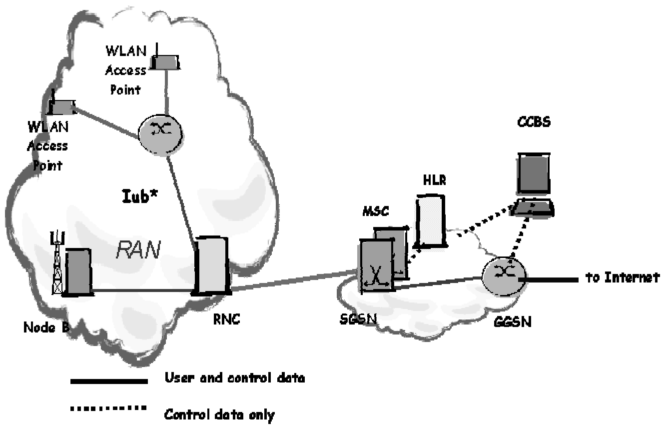


Figure 4.4: Very tight coupling UMTS-WLAN (from [49])

in different ways. 3GPP has defined six scenarios for the coupling, each of them enabling a particular feature and requiring an increasing level of integration for the interworking [47], [48].

The goal of the coupling is on the one hand to combine the wide-area coverage of UMTS, with its associated roaming and mobility properties and, on the other hand, to gain additional throughput and capacity by using WLAN in hotspots.

A general design criterion is the reuse of existing mechanisms and functions, e.g. UMTS subscriber management mechanism, billing functions, UMTS authentication, and security functions.

The six scenarios defined by 3GPP are focused on the type and quality of the service offered to the user. However, from an architectural point of view, these scenarios can be reduced to four levels of coupling, which are shortly discussed in this section. The first level is open coupling, presented in Figure 4.1. In this case UMTS and WLAN make use of two separated access and transport networks and billing is common – although still using different authentication mechanisms.

The next level is the so-called loose coupling, shown in Figure 4.2, which enables the use of common authentication mechanisms by providing a link between the AAA server in the WLAN subsystem and the Home Location Register (HLR) in the UMTS subsystem, which are still kept separate.

With tight coupling, the WLAN Access Point (AP) is connected like a Radio Network Controller (RNC) to the Serving GPRS Support Node (SGSN) in the Core Network (CN). This principle is shown in Figure 4.3. An Interworking Unit might be added, which is discussed in [47]. With tight coupling the handover between WLAN and cellular subsystems could be supported.

Finally, with very tight coupling (Figure 4.4) the WLAN access point is connected to the RNC and, thus, it becomes an integral part of the UMTS Terrestrial Radio Access Network (UTRAN). Very tight coupling provides the possibility to offer a homogenous service to the subscriber and to perform inter system Radio Resource Management (RRM).

The complexity of implementation increases with the level of interworking from the open coupling solution (the simplest one) up to very tight interworking. A higher degree of interworking involves a higher impact on UMTS network elements, which have to provide the necessary capacity with respect to processing power and interface capabilities.

4.3 Location Based MxRRM

For possibly switching between different RATs the system has to know, which RATs are actually providing coverage to the mobile terminal. Having no other information available the mobile terminal would be required to periodically perform extensive scans of the complete spectrum of all RATs that are supported by the mobile terminal. Based on the information gathered by the scans a list could be maintained – either at the mobile terminal or at the network – that contains all the base stations and access points being currently available to the mobile terminal.

However, this approach is expensive in terms of power consumption and hardware demands. For these regular measurements additional battery power would be consumed, which is undesirable for a mobile terminal. To enable the measurements while at the same time having the actual communication on the current RAT still being active, sophisticated transceiver technology has to be built into the terminal to either allow for the measurements to happen in parallel to the transmissions and receptions on the current RAT or either to enable very fast switching between different RATs for utilizing small gaps in the actual communication. In both cases there are high demands to the radio hardware, which may result in higher costs.

Furthermore, if special precautions have to be taken by the mode of operation of a RAT to enable these measurements, then also some of the scarce air interface capacity may be consumed by this approach for maintaining the list of available base stations and access points. An example for such a special mode of operation is the UMTS Compressed Mode [32].

Another approach would be to evaluate the current position of the mobile terminal. Based on this location information the network can provide information on which RATs are potentially available to the terminal. Consequently, only few – or maybe even no – measurements have to be performed by the mobile terminal only when it actually would have to switch from the current RAT to another one.

To be able to provide this information the network has to maintain a radio resource map. This map contains information on which RAT is available at any given position and possibly also on the signal quality to be expected there.

For a real application the area managed by the network probably would be subdivided into small fields and then the information would be maintained per field. Provided that the resolution of this quantization is sufficiently high, i.e. that the size of the individual fields is sufficiently small, this still would allow for adequate accuracy.

The position of the mobile terminal is supposed to be well-known. A first estimation could simply be based on the ID of the cell the terminal is currently registered to in its current RAT. An accurate determination of the position could be based on the assistance of a navigation system like GPS or on measurements performed by the RAT, like e.g. time of flight, timing advance, angle of arrival and many others [50], [51].

The information contained in the radio resource map could be gathered and updated during normal mode of operation. Each time a mobile terminal has a connection to a base station or access point typically several measurements concerning link quality are performed, e.g. for optimizing power control, error correction, etc. The results of these measurements can be added to the map, so over time the map will increase in accuracy and at the same time it is kept up to date continuously. Consequently, such a radio resource map could be created and maintained with virtually no overhead as far as the mobile terminal and the air interface would be concerned.

So the utilization of a radio resource map could be promising when implementing MxRRM [Hil_13].

An elaborated approach would not only consider the position but also speed and direction of the mobile station to further optimize the radio resource management. Fast moving mobile stations could be dispatched to systems with larger cells to reduce handover rate. A prediction of the future trajectory would enable to reserve the resources needed along the path of the mobile station. If within one system on the predicted path the required QoS can not be provided by one or several cells, the connection could be shifted to a system that can provide the QoS on the whole path in order to reduce inter-system handovers.

Currently in cellular systems typically a certain percentage of capacity is reserved for handovers from adjacent cells. This percentage could be increased in cells that seem to welcome more new mobile stations in near future and could be decreased in cells that are targeted by only a small number of moving mobile stations. As a result the dropping rate would be reduced while the capacity would be increased.

4.4 Algorithms

For this investigation four algorithms have been defined for handling circuit switched services and four algorithms have been defined for handling packet switched services, respectively. In mixed service scenarios one particular algorithm of the first group and one particular algorithm of the latter group will be employed concurrently.

4.4.1 MxRRM for CS Services

For circuit switched services four different MxRRM strategies and related algorithms have been defined and investigated. The different strategies represent different grades of integration. These grades range from no integration to a close integration by means of exchange of load information between the RATs as supported by Release 5 UMTS/GERAN enhanced signalling means.

For each call a default RAT (either GSM or UMTS) is defined. If sufficient capacity was available a call would be assigned to its default RAT and would remain there until successful call completion. If the capacity available was insufficient or was becoming insufficient, i.e. if blocking or dropping would occur otherwise, a call may be assigned to the alternative RAT (directed retry or inter system handover) according to the MxRRM strategy being used. If in both RATs the call could not be supported then finally it would be blocked or dropped.

4.4.1.1 Separate Systems

This strategy does not allow inter system handovers of mobiles. Mobiles always stay in their RAT, i.e. both systems are separated completely. This strategy represents a reference case allowing the comparison of gains achievable by different grades of RAT integration by different MxRRM strategies.

4.4.1.2 Intersystem Handover (blind)

Here the selection of the target cell in the alternative RAT is based on radio conditions only, which are measured by the signal to interference ratio of the pilot channels. No load information of the potential target cells is used in this algorithm. In other words the handover is tried "blindly" with respect to the load situations in the cells. Due to the missing load information it might happen that a cell is selected, which is not able to accept the handover, because it is already fully loaded. Hence there is some probability that a handover to the other RAT may not be accepted by the target and is therefore rejected. In these cases the call cannot be established or has to be finally dropped.

Due to the missing signalling support for inter system load information exchange in Release 99 load information cannot be employed in an algorithm. This strategy therefore represents what could be already implemented within the currently deployed 3GPP Release 99 systems. It is therefore a reference to which more sophisticated MxRRM strategies are compared.

4.4.1.3 Intersystem Handover (MxRRM)

This algorithm makes use of load information as it could be exchanged between different RATs using Release 5 signalling mechanisms. When a call cannot be supported by its current RAT then a target cell in the alternative RAT is selected, which has the best radio conditions out of the cells that could accept the call according to their load situation. This avoids the deficiency of the algorithm "Intersystem Handover Blind" where it might occur that a cell with best radio conditions is selected as target, even though this cell may not be able to accept the call due to its load situation whilst in parallel there are other target cells which also would have sufficient radio conditions and would be able to accept the call.

Due to this improvement it should happen more rarely that the intersystem handover or directed retry cannot be performed successfully.

4.4.1.4 Load Balancing

With this strategy, in addition to the "Intersystem Handover (MxRRM)" algorithm a load balancing is applied at call setup. This means that even if a call could be accepted by a cell in the default RAT the current load of this cell is evaluated and, if it is above a threshold value, it is compared to the load of the potential target cell in the alternative RAT. Finally the call is assigned to the cell being less loaded.

4.4.2 MxRRM for PS Services

The MxRRM strategies presented in this section are suitable for packet switched traffic. Each time a new packet call arrives in the system, the MxRRM algorithm is triggered and decides on, to which RAT the packet call should be assigned. If the MxRRM algorithms evaluates to the result that an alternative RAT should be assigned to the packet call instead of its current RAT then the call would be directed to the alternative RAT.

Each time a newly arrived packet call is assigned to an alternative RAT, instead of its current RAT, is counted as one intersystem cell reselection (IS-CRS).

Based on the scenario configuration for a packet session (see section 5.2.2 for details of the traffic model) a default RAT is defined randomly. For the first packet call of a session the current RAT is set to the default RAT. For further packet calls of the same session the current RAT is the RAT the previous packet call was assigned to.

For this investigation we considered a case with two co-existing RATs, i.e. one layer of GSM / GPRS and one layer of UMTS, so there is one alternative RAT to be evaluated. For the algorithms presented in the following subsections, there are parameters whose values could be chosen arbitrarily. In preparatory studies, by scanning the parameter space, their values have been determined to achieve the best mean bit rate per user for the given scenario.

4.4.2.1 Separate Systems

This strategy does not allow for IS-CRS, i.e. the systems are separated completely. As a result a packet session is served in its default RAT only. This strategy represents a reference case allowing the comparison of gains achievable by different grades of RAT integration by different MxRRM strategies.

4.4.2.2 Cell Load based

If the cell of the current RAT, to which the packet call would be assigned, has a load exceeding a pre-defined threshold then the potential target cell in the alternative RAT and its load is determined. If the load of the alternative cell is below the threshold value then the call is assigned to the alternative cell else it stays in the current cell.

A preparatory study determined the optimal threshold value for the investigated scenario as 75 %.

The idea of this strategy is to balance the load between the RATs to achieve a better utilization of resources as well as a higher mean data rate per user.

Since the algorithm does not compare to the load of the alternative RAT but only to the predefined threshold value, no actual exchange of load information would be required. Hence, this approach would be feasible with the signalling capabilities provided by UMTS Release 99.

4.4.2.3 Bit Rate based

This strategy achieves load balancing by striving for similar mean data rates per user in all RATs. At packet call setup the mean data rate per user of the target cell in the current RAT is compared to the mean data rate per user of the potential target cell in the alternative RAT. If the difference of the data rates exceeds a certain amount in favour of the alternative RAT then an IS-CRS is performed.

A preparatory study determined the optimal threshold value, for the investigated scenario as 225 %, which means that for example if in the current RAT the mean data rate was 10 kb/s then in the alternative RAT it would have to be at least 32.5 kb/s to trigger an IS-CRS.

The aim of this approach is by achieving a “fair” distribution of users to increase the mean data rate per user.

This approach would be feasible with the signalling capabilities provided by UMTS Release 5.

4.4.2.4 Minimum Service based

A minimum quality of service has been defined for this investigation as being achieved if a user has a current data rate of at least 50 % of the maximum data rate that his terminal could support. Those users, who experience “minimum service” by receiving a data rate above this threshold, are assumed to be satisfied users. The percentage of satisfied users in a cell is used as a measure of cell load, where less users being satisfied means a higher cell load.

The actual percentage of satisfied users is mapped to one of four values (low load, medium load, high load, overload) describing the cell load according to the “Non-Real-Time Load Information Value” field of the load info message defined in [8] for UMTS Release 5.

At a packet call setup, the NRT load information value of the cell in the current RAT is then compared with the one of the potential target cell in the alternative RAT. An IS-CRS is performed if the value of alternative cell is less than the value of the current cell.

A preparatory study determined the optimal threshold values, which then were set to 40 %, 70 % and 100 %, meaning that less than 40 % of a cell’s users being satisfied indicates “overload”, less than 70 % indicates “high load”, less than 100 % indicates “medium load” and having all users being satisfied indicates “low load”.

The aim of this approach is not only achieving an increase of the mean data rate per user but also to increase minimum data rates that a majority of the users has to experience.

This approach would be feasible with the signalling capabilities provided by UMTS Release 5.

4.5 Analytical Estimation

For an estimation of the potential trunking gain generated by MxRRM we assume a system configuration providing the equivalent of $n_{Channel}$ channels per GSM cell as well as per UMTS cell. This assumption neglects the fact that UMTS is a CDMA based system that does not have a fixed capacity per cell but has a soft capacity that depends on parameters like interference, distances between users and base station, load, etc. Since for the analytical estimation we consider one particular scenario having similar characteristics for all the cases investigated the mean capacity per UMTS cell is supposed to be approximately constant, thus justifying this assumption and allowing for a sufficiently accurate estimation.

Circuit switched service like telephony is considered with one call using one channel.

To calculate the blocking probability based on Erlang B, as given by Equation (19), the amount of resources available to potentially serve a request has to be determined. Here the size of the resource pool is depending on the number of channels per cell as well as on the number of cells $n_{Cell\_available}$ that could potentially serve the call setup request. So the number of servers N as required by Erlang B is given by Equation (20).

$$p_B = \frac{\frac{A^N}{N!}}{\sum_{i=0}^{N} \frac{A^i}{i!}} \tag{19}$$

With p_B *being the blocking probability,* A *being the traffic offered in Erlang, and* N *being the number of servers (i.e. channels, trunks, …)*

$$N = n_{Cell\_available} \cdot n_{Channel} \tag{20}$$

With $n_{Cell\_available}$ *being the number of cells that potentially could serve the call setup request and* $n_{Channel}$ *being the number of channels per cell.*

Depending on the position of the mobile terminal, the cell layout and the radio conditions in a cellular system, typically more than one cell can be received by the mobile terminal [25]. Due to the cell layout shown in Figure 5.4, three cells are assumed to be “visible” typically. The results presented in the following are given for a number of visible cells varying between two and four. Additionally the possibility for inter system handover increases the number of available cells, because then cells from the alternative RATs add to this number. Basically here we have to distinguish three types of RAT coupling:

- “Separate Systems”: Only cells of the default RAT are “visible” to the mobile terminal.

Cells "visible" at call setup	Separate Systems	IS HO (Blind)	IS HO (MxRRM)
2	17.30	18.19	18.70
3	18.19	18.70	19.30
4	18.70	19.05	19.63

Table 4.1: Maximum traffic offer (in Erlang) per cell for 5 % blocking rate (20 channels per cell)

Cells "visible" at call setup	IS HO (blind) vs. Separate Systems	IS HO (MxRRM) vs. Separate Systems	IS HO (MxRRM) vs. IS HO (Blind)
2	5.1 %	8.1 %	2.8 %
3	2.8 %	6.1 %	3.2 %
4	1.8 %	5.0 %	3.1 %

Table 4.2: Relative gain generated by different inter system handover capabilities

- "Intersystem Handover (blind)": For inter system handover only one cell of the alternative RAT (the cell with the best pilot signal) is considered, so the number of "visible" cells is increased by one.
- "Intersystem Handover (MxRRM)": For inter system handover all "visible" cells of the alternative RAT are considered, so the number of "visible" cells is doubled in case of two RATs.

The maximum traffic that can be offered per cell to achieve 5 % blocking rate is given in Table 4.1.

The relative gain generated by intersystem handover capabilities is given in Table 4.2.

To sum up we can say that by enabling intersystem handovers the blocking rate is reduced. Consequently, to achieve a given maximum blocking rate a higher load per cell can be offered if inter system handovers are possible.

A trunking gain is generated even with “blind” handovers, i.e. if only the cell providing the best signal quality is considered as target cell. By evaluating load information the MxRRM approach further increases the gain.

When comparing these analytical results to the results obtained by simulation then we find a good compliance. In Table 6.1 a performance gain of 7.6 % is given for “Intersystem Handover (MxRRM)” against “Separate Systems” (101.4 / 94.2), whereas in Table 4.2 a gain of 5.0 to 8.1 % is given. For “Intersystem Handover (MxRRM)” against “Intersystem Handover (blind)” in Table 6.1 a gain of 1.4 % is given whereas in Table 4.2 a gain of 2.8 to 3.2 % is given. Adding load balancing to the “Intersystem Handover (MxRRM)” algorithm results in 3.4 % performance gain given in Table 6.1 against “Intersystem Handover (blind)”, which is close to the calculated values.

Thus, the analytical estimation validates the simulation results.

5 MxRRM Simulation Environment

"Basically, simulation is the process by which understanding of the behaviour of an already existing (or to be constructed) physical system is obtained by observing the behaviour of a model representing the system." [52]

There are several approaches to realize a computer simulation of communication systems. Here the discrete event type was chosen. Section 5.1 gives an overview of the tools and software environment that was used for implementing and executing the simulations.

From the definition above it can be concluded that the development of an appropriate system model is one of the first and most important steps, when performing an investigation by means of simulation. The characteristics of the model determine the type and quality of information that can be obtained from the simulation. However, the system model is not an end in itself, but rather it is a mere tool for achieving knowledge about the system to be simulated. Therefore, the system model should reproduce only those parts and behaviours of the real system, which are meaningful in respect to the focus of investigation, in order to limit complexity, implementation effort and computing time. Consequently, there is a challenge in keeping a system model efficient while providing results with sufficient accuracy and level of detail.

How this challenge was solved for the research work presented in this thesis is described in section 5.2 by introducing the models for the individual parts of the system. In section 5.3 the scenario used for the investigation is presented.

5.1 Tools

The simulation was created using Opnet Modeler, a commercially available simulation package [53]. Opnet Modeler already includes a high number of models, which could be used instantly for simulation, but since for the investigation presented in this thesis these models were not appropriate, new models had to be developed. These were implemented in C++ following the object oriented paradigm [54], [55], [56]. By carefully adhering ANSI compliance and extensively using the Standard Template Library

(STL) [57], [58], full portability has been achieved, i.e. the software can be run under Windows as well as under Solaris.

The software development including test and debugging was performed under Windows using the Microsoft Developer Studio 2003, whereas the simulations for the actual investigation have been performed under Solaris using the Sun Forte 5.4 C++ compiler. In both cases, the Opnet Modeler 10.0 has been used.

Further details regarding the discrete event simulation paradigm can be found in the appendix at page 127 and further details regarding the software architecture of the simulator can be found in the appendix at page 133.

An important issue to be considered when using stochastic simulations is the statistical confidence of the results. An overview of this topic is provided in the appendix, starting at page 129.

5.2 Models

Capacity estimations of single systems have been done by several authors. For CDMA typically the focus was on the up link (reverse link), because from pure system point of view this one was supposed to be the potential bottle neck (for example [59]). Other work also investigated down link (forward link), because the data applications may create significantly more traffic in that direction thus shifting the potential bottle neck to the down link (for example [60]). This thesis complies with the second view.

For TDMA systems it is more straightforward to estimate the capacity simply based on the amount of available channels, i.e. time slots, provided that radio resource management uses simple and linear mechanisms and provided that interference can be assumed either to be constant or to be negligible. For example the blocking probability can be acquired by applying the classical Erlang analysis.

Here we do not strive for a detailed performance investigation of the GSM and UMTS system but rather concentrate on comparing the different results regarding capacity measures – i.e. number of users, mean data rate, blocking rate or dropping rate – when applying different MxRRM strategies (see Figure 5.1).

When creating the simulation environment the intention was to investigate the impact of multi standard radio resource management on system performance, i.e. to compare the results of a given

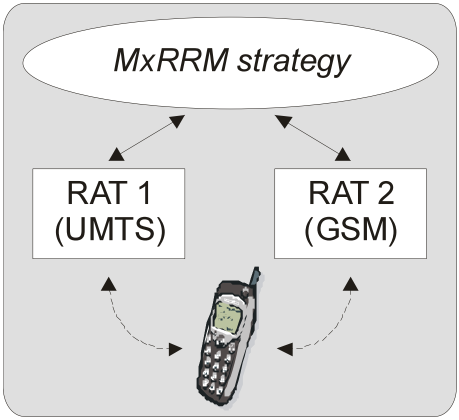


Figure 5.1: All scenarios investigated have identical systems and configuration but different MxRRM algorithms

system that uses none or one out of several MxRRM algorithms. The idea was clearly not to investigate nor optimize radio resource mechanisms that are implicit to a particular radio access technology, like e.g. power control or channel assignment.

Consequently the modelling of the individual radio access technologies was kept to a level of detail that provides sufficiently accurate results on capacity, i.e. number of active users and throughput, without replicating each single air interface feature. For the same reasons also the fixed network part of the communication systems was neglected.

Our simulation is limited to the down link according to the reasons discussed above.

The simulator was designed as event driven simulation. It has been built in context of the commercially available simulation package Opnet. The actual models have been implemented in C++ using an object-oriented approach.

5.2.1 Mobility Model

The mobility model used in our simulation environment is based on the models described in [31]. Basically there are three different types of mobility: stationary, pedestrian and vehicular, but the stationary type typically is used for test and validation purposes only.

The pedestrian mobility type models the behaviour of a slow moving user who potentially changes the direction often and without preference. The speed of a pedestrian user is set to 3 km/h constantly.

The vehicular mobility type models the behaviour of a fast moving user, who may make small changes to the current direction but keeps to an overall main direction. The speed of a vehicular user is set to 120 km/h constantly.

The movement itself is not modelled as a continuous motion for reducing complexity and computation time. When moving from one position to the next position the mobile terminal stays on the first position until the distance to the next position would be covered. Then the terminal "jumps" to the next position and performs a position update. So the movement of a mobile terminal is modelled as a sequence of large steps with each step covering the same distance, which is called the step length.

With each position update also the direction for the next step may be changed randomly. The probability for a direction change to happen as well as the possible amount of change is defined by the actual type of mobility of the particular mobile terminal.

This approach is feasible because small changes of position cause only a very small change in propagation loss and also in shadowing (slow fading). The distance that two positions have to be away from each other to cause a significant change of total path loss is called the decorrelation length [61]. Hence, it is considered as sufficient to update the path loss value for a particular link only if the mobile terminal has been moved at least by the decorrelation length since the most recent path loss update has been performed.

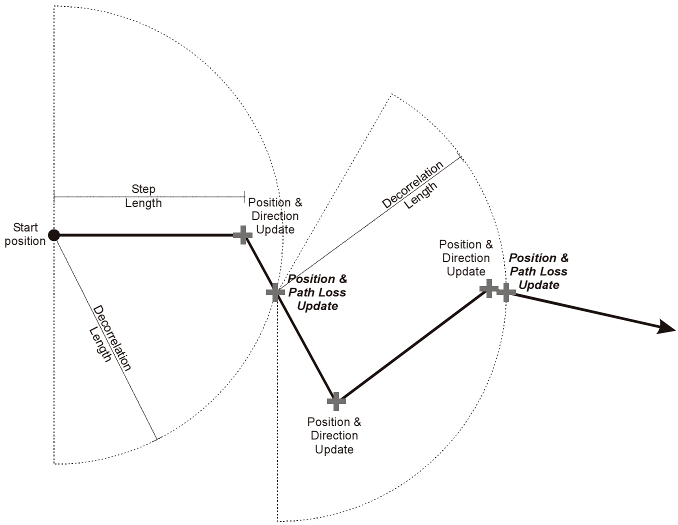


Figure 5.2: Relation of decorrelation length and step length (exemplified)

In [31] the step length for movement has been chosen to equal the decorrelation length of UMTS, which is given there as 20 m. This simplifies the implementation of the mobility model as well as it improves computation time, because as a result each path loss update has to be performed at the same time instant (i.e. at the same simulation event) as each position update and each direction update.

Since our simulator includes more than one radio access technology there is more than one decorrelation length to be considered. If there would be this strict coupling of step length and decorrelation length this would mean that the step length would depend on the current radio access technology being used by the mobile terminal. As a result the movement pattern of a mobile terminal would de different for different radio access technologies, which is not desirable. Hence, we decoupled these two items thus creating a mobility model with two series of events; the updates of position and direction happen based on the step length of 20 m while the update of path loss is performed each time the decorrelation length valid for the current radio access technology is exceeded, so there may be several, one or even none path loss update during one step (see Figure 5.2).

5.2.2 Traffic Model

For each traffic type that should be included in a simulation an individual traffic generator has to be configured. The random arrival of a new call or session is based on an exponential distribution independently for each generator.

For circuit switched services the application considered is voice telephony assuming an adaptive multi rate coder working constantly at high quality, thus requiring a constant user data rate of typically 12.2 kb/s (configurable). For each call the duration is chosen randomly based on exponential distribution. The activity factor is set to 1 so the model would also apply to real time data services provided appropriate data rates are configured.

Web browsing is the application considered for packet switched services. The model assumes a user to have a packet session consisting of several packet calls that represent the subsequent downloads of web pages. Between each packet call there is an idle phase called reading time, which starts after the transmission of the preceding packet call has been completed and is randomly determined based on a geometrical distribution. The duration of a packet call results from the user data rate that is assigned to a data connection and by the amount of data contained in the packet call, i.e. its size. A packet call for downloading a web page actually would contain several packets, since a web page may consist of several files (e.g. pictures, text, etc.). The sizes of these packets are randomly determined based on a Pareto distribution and their number is randomly determined based on a geometrical distribution. The sizes of the packets are summed to get the size of the packet call. Then only the size of the packet call is considered. Hence, the model for PS traffic effectively generates only complete packet calls but still incorporates the fact that there are actually several packets contained in a packet call (see Figure 5.3).

The traffic models are based on the proposals made by [31].

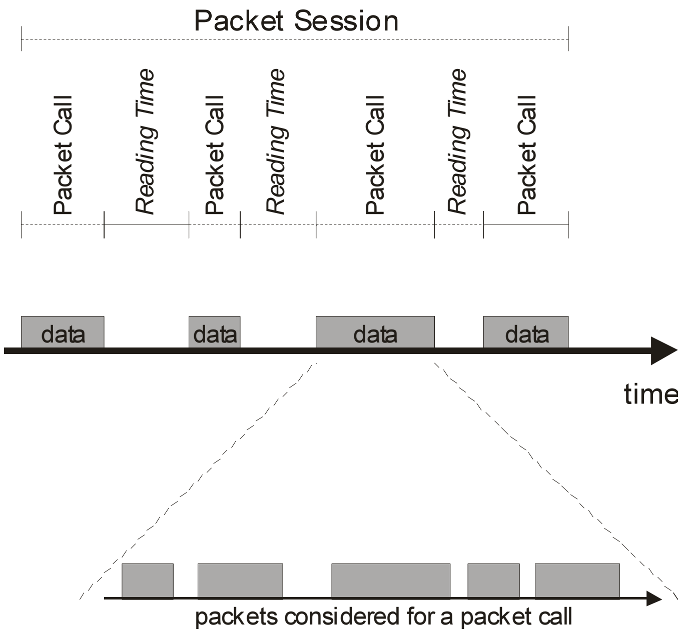


Figure 5.3: Traffic model for web browsing

5.2.3 UMTS Air Interface Model

Being CDMA based a characteristic of the UMTS air interface is that its capacity as well as the coverage is not fixed but depends on several factors like for example distance between base station and mobile terminal, number of active users or individual user data rates [32], [62], [63]. Hence, the terms "soft capacity"' and "cell breathing" have been established. For the purpose of validating the simulator also some investigations of the latter have been accomplished (see section 3.2.1).

Due to the use of CDMA the resource being shared among the users is the transmission power. Thus an adequate determination of each communication link's transmission power has to be performed any time the radio conditions of a link have been changed. For the simulator an approach was developed that allows for the closed calculation of transmission powers of all links being in one cell [Hil_4]. The total transmission power required for a cell is calculated according to Equation (21), and then the power required for each link is determined based on Equation (22).

$$P_k = \frac{P_{k,common} + \sum_{i=1}^{n_k} \left(\frac{\frac{1}{L_{k,i}} \cdot \left(I_{i,Extra} + N \right)}{\frac{1}{SIR_{i,required}} + \alpha} \right)}{1 - \alpha \cdot \sum_{i=1}^{n_k} \left(\frac{1}{\frac{1}{SIR_{i,required}} + \alpha} \right)} \tag{21}$$

With P_k *being the total transmission power of cell* k*,* $P_{k,common}$ *being the transmission power of the common (broadcast) channels of cell* k*,* α *being the non-orthogonality factor,* $L_{k,i}$ *being the path loss on link* i *of cell* k*,* $I_{i,Extra}$ *being the extra cell interference experienced on link* i*,* N *being the thermal noise and* $SIR_{i,required}$ *being the signal to interference ratio required to achieve the desired QoS for link* i *(target SIR).*

$$P_{k,i} = \frac{\alpha \cdot P_k + \frac{1}{L_{k,i}} \cdot \left(I_{i,Extra} + N \right)}{\frac{1}{SIR_{i,required}} + \alpha} \tag{22}$$

With $P_{k,i}$ *being the transmission power of cell* k *for link* i*,* α *being the non-orthogonality factor,* P_k *being the total transmission power of cell* k*,* $L_{k,i}$ *being the path loss on link* i *of cell* k*,* $I_{i,Extra}$ *being the extra cell interference experienced on link* i*,* N *being the thermal noise and* $SIR_{i,required}$ *being the signal to interference ratio required to achieve the desired QoS for link* i *(target SIR).*

In the Equations (21) and (22) it can be seen, that the transmission power required would depend only on extra cell interference and noise if a perfect orthogonality could be achieved for the intra cell signals ($\alpha = 0$). Also, a higher propagation loss (i.e. a smaller value for *L*) between base station and mobile terminal will increase the impact of interference and noise, thus causing distant users to require a higher transmission power.

In our simulator the non-orthogonality factor is set to $\alpha = 0.4$ based on [32]. This value was confirmed by the results presented in [64].

Each time an event occurs that has impact on the radio conditions of a cell the re-calculation of all links supported by this cell is performed and the transmission power of the base station is adjusted. Such an event may be a change of

- path loss due to movement of a mobile station,
- cell load due to call arrival (call setup, handover) or call leaving (call termination, dropping, handover),
- extra cell interference.

As a result all the links of the updating cell achieve again a perfect signal to interference ratio (SIR) but mobile stations having links to adjacent cells may experience a change of extra cell interference. Hence, all mobile stations belonging to other cells but being in the range of this cell have to check their current SIR. If due to the power update such a mobile station detected that its current SIR differs from the target SIR by a value greater then a configurable threshold, then it triggers a power update of its own cell. Typically, the threshold is set to 0.5 dB. Because the UMTS closed loop power control applies 1.0 dB steps per power update [32], an inaccuracy of half of this value is considered by the authors to be acceptable.

In fact a power update of one cell may involve subsequent power updates of adjacent cells, which may cause a further update of the original cell, thus causing the sequence to start over. The simulation time does not continue until the power update process is completed for all cells involved. Although the calculation of transmission power for a cell is done in one deterministic step, consequently a complete power update is rather an iterative process. Due to the fact that the changes causing a particular update typically are of minor size and due to the introduction of the above mentioned threshold value this algorithm converges very fast, though.

This approach models an immediate and perfect power control, contrary to the real system, where some power control cycles may be needed to adapt the transmission power accordingly and where the optimum transmission power may never be achieved exactly due to the continuously changing environment. As a result, the cell capacity obtained by the simulator is supposed to be optimistic. Since the investigation presented in this paper does not aim on UMTS physical layer, power control or cell capacity as such, but on the comparison of different MxRRM algorithms applied to one given system, this modelling approach is considered suitable.

Circuit switched calls put a strict requirement on QoS, i.e. the requested QoS has to be satisfied all the time during an ongoing call. Here QoS is expressed by the data rate. Based on results from link level simulations [65], the simulator applies a mapping from data rate to SIR and vice versa also considering the velocity of the mobile station. Thus, for a CS call requesting a particular data rate the SIR required to provide this data rate is determined and set as target SIR. The relation between power density per bit and SIR is given in Equation (23).

Packet switched calls are treated in a best effort manner. The amount of resources (in UMTS: transmission power) available for a particular PS call is determined and the resulting SIR is calculated. Based on this, the current data rate of this connection is derived.

$$\left(\frac{E_b}{I_0}\right)_i = \eta_i \cdot SIR_i \tag{23}$$

With E_b/I_0 *being the power density per bit over interference and noise power density,* η *being the processing gain due to spreading and coding,* SIR *being the signal to interference ratio of the actual radio signal and* i *being the index of the particular link.*

The propagation loss model is chosen according to the "vehicular" model described in [31]:

$$L = 15{,}3 + 37{,}6 \cdot \lg(r) \tag{24}$$

With L *being the propagation loss in dB, and* r *being the distance between BS and MS in m.*

A directional antenna was assumed for the base stations with an antenna pattern typical for sectored sites according to [31].

5.2.4 GSM Air Interface Model

For GSM no sophisticated power control is required, hence only the SIR has to be monitored to determine whether a mobile terminal is still within the range of a base station or which data rate currently could be achieved. The GSM path loss L is calculated according to [66]:

$$L = 18{,}8 + 38 \cdot \lg(r) \tag{25}$$

With L *being the propagation loss in dB, and* r *being the distance between BS and MS in m.*

The same directional antenna was assumed for the base stations as for UMTS.

Since interference has a minor impact in GSM than in UMTS a simplified model for extra cell interference was applied that assumes a constant value of -105 dBm, which is a valid approximation according to [67], [68].

For PS services EDGE capabilities were assumed and the user data rates were determined based on SIR, as given in [67].

5.3 Scenario

This investigation compares different systems to evaluate the possible impact of MxRRM and to identify promising approaches on how to design appropriate algorithms. Hence, a “system” is supposed to consist of several independent radio access technologies (here: UMTS and GSM / GPRS) that are coupled by a particular MxRRM mechanism. For the individual subsystems, i.e. each RAT, a configuration is defined that is sufficiently close to a real system, but that can be implemented and understood relatively easily. Since it is out of focus to optimize the radio resource management of the actual RATs this configuration is kept constant during the investigation and only the mechanisms relevant for MxRRM are modified.

To avoid peculiar effects that are simply based on different RAT capacity, in preparatory simulation studies configuration parameters of the different systems have been tuned such that the UMTS system and the GSM system exhibit nearly identical behaviour with regard to blocking and dropping – i.e. for CS services – at the chosen operating point, which is defined by an offered load of

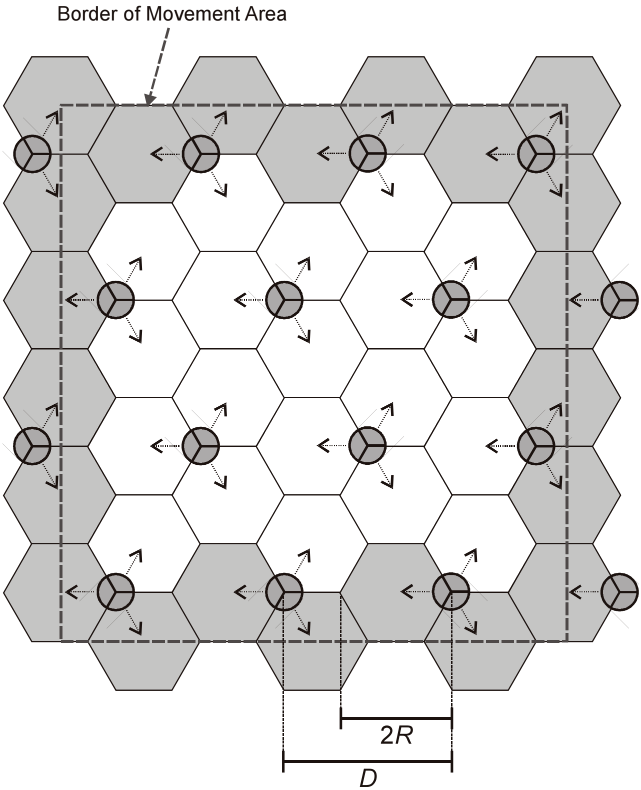


Figure 5.4: Network topology: 42 cells with the base stations being placed at the cell corner

18.2 Erlang per cell (i.e. 36.4 Erlang per cell area) and a blocking rate of approx. 5 %. Table 5.1 summarizes the determined parameters. This parameter set is used persistently for this investigation.

As a consequence, in the simulated scenario the PS capacity of the GSM / GPRS system is higher than the one of the UMTS system, because for PS the EDGE features are enabled for GSM / GPRS, whereas UMTS is basically using the same modulation and coding techniques for CS and PS, respectively. By using different configurations or by even adding HSDPA features to UMTS, the PS capacity could be matched, but here the priority was given to achieving similar CS capacities for both systems.

A "call setup margin" of x % means that when processing the arrival of a new call only x % of the resources of the cell are

	UMTS	GSM / GPRS
Max TX power	42.2 dBm	30 dBm
Pilot TX power	30 dBm	30 dBm
Max power per link	33 dBm	-
Time slots	-	21
Call setup margin	70 %	95 %
	CS traffic	*PS traffic*
Application	Voice call	Web browsing
Data rate	12.2 kb/s (fixed)	128 kb/s (max)
Mean duration / size	120 s	58.6 kB/session
	Pedestrian mobile terminals	*Vehicular mobile terminals*
Fraction	87.5 %	12.5 %

Table 5.1: Scenario configuration parameters

considered. If this fraction of the resources is already completely utilized then the call request will be rejected. In other words, (100 – x) % of the resources is reserved for already ongoing calls, thus giving them higher priority over newly arriving calls.

The actual study comparing the performance of the different algorithms was made using three different sub scenarios:

- Symmetrical Load in which the total load offered in a cell area was equally distributed between UMTS and GSM.
- Asymmetrical Load (UMTS 80 %) in which the total load offered in a cell area was distributed in a way that 80 % of all new calls arrived in UMTS.
- Asymmetrical Load (GSM 80 %) in which the total load offered in a cell area was distributed in a way that 80 % of all new calls arrived in GSM.

The simulated time was six hours for the CS investigations and three hours for the PS investigations, from which each first hour did not contribute to the results to avoid influence of transient system states.

The simulated network consists of 42 cells placed in the usual hexagonal layout, each having a cell radius of 800 m. The base stations are placed at the cell corner (see Figure 5.4), thus having a distance of three times the cell radius (i.e. 2400 m). The border cells (grey coloured) have a special configuration and do not contribute to the simulation results to avoid possible peculiar border effects.

6 MxRRM Simulation Results

In this chapter the results of the simulations are presented. Based on the types of service being considered three different investigations can be distinguished. The results for MxRRM applied to circuit switched services only are presented in section 6.1, and the results for MxRRM applied to packet switched services only are presented in section 6.2, respectively. In section 6.3 the results for MxRRM applied to mixed services are given. For details of the scenario and the configuration being used refer to section 5.3.

6.1 MxRRM for CS Services

The total load being offered was adjusted to achieve 5 % blocking for each scenario (symmetric / asymmetric load) and each algorithm. The simulated time was six hours, from which the first hour did not contribute to the results to avoid influence of transient system states.

Three scenarios (see section 5.3) and four algorithms (see section 4.4.1) result in a total number of twelve simulations to be evaluated. Figure 6.1 to Figure 6.3 visualize the results for symmetrical load and Table 6.1 to Table 6.3 give detailed results as a comparison to the reference algorithm “Inter System Handover Blind” by normalizing the results for this reference to 100 % and showing the other results with respect to this basis. All results are obtained at a blocking rate of 5 %. Detailed results with absolute values are provided by Table 6.4 to Table 6.6.

In Figure 6.2 the offered and the carried loads are displayed for each of the algorithms. With increasing MxRRM capabilities the difference between offered and carried load becomes smaller while the load increases.

The blocking and dropping rates are depicted in Figure 6.1. The blocking is kept constantly at 5 % because this was the working point specified. With increasing MxRRM capabilities the dropping is reduced significantly.

	Separate Systems	IS-HO Blind	IS-HO MxRRM	IS-HO Load Balancing
Blocking Rate	98.9 %	*100.0 %*	100.3 %	100.2 %
Dropping Rate	323.5 %	*100.0 %*	18.1 %	19.7 %
Offered Load	94.2 %	*100.0 %*	101.4 %	103.4 %
Carried Load	89.6 %	*100.0 %*	102.6 %	104.7 %
IS-HO Requests	-	*100.0 %*	147.8 %	200.9 %
IS-HO Failure Rate	-	*100.0 %*	12.3 %	10.1 %
Directed Retries	-	*100.0 %*	166.4 %	576.4 %

Table 6.1: Comparison, symmetrical load (UMTS/GSM 50:50)

	Separate Systems	IS-HO Blind	IS-HO MxRRM	IS-HO Load Balancing
Blocking Rate	101.6 %	*100.0 %*	100.9 %	100.2 %
Dropping Rate	795.6 %	*100.0 %*	41.3 %	41.3 %
Offered Load	63.0 %	*100.0 %*	106.7 %	106.9 %
Carried Load	61.0 %	*100.0 %*	106.0 %	106.4 %
IS-HO Requests	-	*100.0 %*	195.7 %	205.4 %
IS-HO Failure Rate	-	*100.0 %*	22.4 %	21.5 %
Directed Retries	-	*100.0 %*	143.4 %	231.6 %

Table 6.2: Comparison, asymmetrical load (UMTS/GSM 80:20)

In Figure 6.3 the number of directed retries and intersystem handover requests is presented. With increasing MxRRM capabilities these numbers increase as well. Particularly with the “Load Balancing” strategy there is a much higher number of directed retries to be found.

	Separate Systems	IS-HO Blind	IS-HO MxRRM	IS-HO Load Balancing
Blocking Rate	98.8 %	*100.0 %*	99.2 %	99.9 %
Dropping Rate	230.6 %	*100.0 %*	19.5 %	14.0 %
Offered Load	60.1 %	*100.0 %*	99.1 %	102.0 %
Carried Load	58.0 %	*100.0 %*	100.4 %	103.8 %
IS-HO Requests	-	*100.0 %*	129.5 %	198.9 %
IS-HO Failure Rate	-	*100.0 %*	14.9 %	7.0 %
Directed Retries	-	*100.0 %*	104.1 %	205.2 %

Table 6.3: Comparison, asymmetrical load (UMTS/GSM 20:80)

A significant capacity increase of up to ca. 70 % and a strong reduction of dropping by up to 95 % could be found with MxRRM compared to separate systems. When comparing MxRRM to a mere “blind” inter system handover, a capacity increase of up to 6.9 % and a still significant reduction of dropping by up to 86 % could be observed.

In general the simulation results show each time an improvement when switching the MxRRM strategy from “Separate Systems” to “Inter System Handover (Blind)”, then to “Inter System Handover (MxRRM)” and finally to “Load Balancing”, with the differences becoming smaller with each step. At the same time the number of directed retries and inter system handovers is increasing. Particularly with the “Load Balancing” strategy there is a much higher number of directed retries to be found.

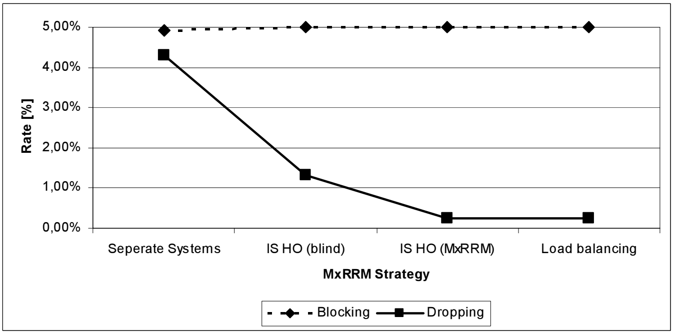


Figure 6.1: Blocking and dropping (UMTS/GSM 50:50)

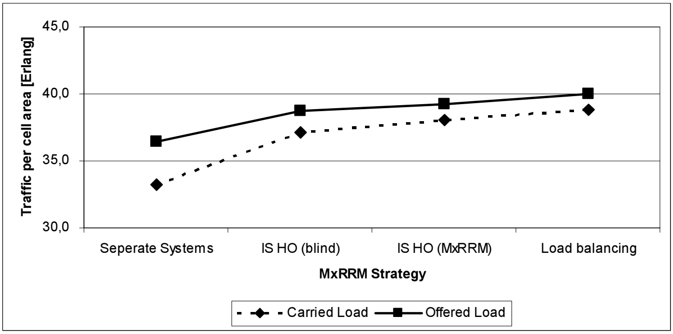


Figure 6.2: Traffic (UMTS/GSM 50:50)

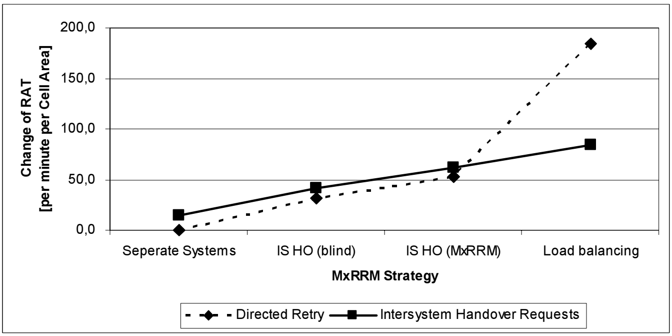


Figure 6.3: Directed retries and inter system handovers (UMTS/GSM 50:50)

	Separate Systems	IS-HO Blind	IS-HO MxRRM	IS-HO Load Balancing
Blocking Rate [%]	4.93 (± 7.5 %)	4.99 (± 9.1 %)	5.00 (± 9.9 %)	5.00 (± 8.3 %)
Dropping Rate [%]	4.29 (± 4.0 %)	1.33 (± 10.4 %)	0.24 (± 15.1 %)	0.26 (± 17.0 %)
Offered Load [Erlang/Cell Area]	36.4 (± 0.0 %)	38.7 (± 0.0 %)	39.2 (± 0.0 %)	40.0 (± 0.0 %)
Carried Load [Erlang/Cell Area]	33.2 (± 0.5 %)	37.1 (± 0.8 %)	38.0 (± 0.6 %)	38.8 (± 0.6 %)
IS-HO Requests [$min^{-1} \cdot$ Cell $Area^{-1}$]	-	42.0 (± 4.9 %)	62.0 (± 4.8 %)	84.3 (± 4.9 %)
IS-HO Failure Rate [%]	-	11.7 (± 9.1 %)	1.44 (± 15.0 %)	1.18 (± 16.3 %)
Directed Retries [$min^{-1} \cdot$ Cell $Area^{-1}$]	-	32.0 (± 5.9 %)	53.2 (± 5.3 %)	184.3 (± 1.4 %)

Table 6.4: Results, symmetrical load (UMTS/GSM 50:50) with relative confidence interval for 99 % confidence

	Separate Systems	IS-HO Blind	IS-HO MxRRM	IS-HO Load Balancing
Blocking Rate [%]	5.00 (± 7.4 %)	4.93 (± 7.8 %)	4.97 (± 11.3 %)	4.94 (± 7.2 %)
Dropping Rate [%]	3.57 (± 3.7 %)	0.45 (± 15.0 %)	0.19 (± 19.1 %)	0.19 (± 19.2 %)
Offered Load [Erlang/Cell Area]	23.6 (± 0.0 %)	37.5 (± 0.0 %)	40.0 (± 0.0 %)	40.1 (± 0.0 %)
Carried Load [Erlang/Cell Area]	22.2 (± 0.7 %)	36.4 (± 1.0 %)	38.6 (± 0.7 %)	38.7 (± 0.7 %)
IS-HO Requests [$min^{-1} \cdot$ Cell $Area^{-1}$]	-	34.7 (± 7.2 %)	68.0 (± 7.2 %)	71.3 (± 4.7 %)
IS-HO Failure Rate [%]	-	4.59 (± 12.4 %)	1.03 (± 14.8 %)	0.98 (± 18.0 %)
Directed Retries [$min^{-1} \cdot$ Cell $Area^{-1}$]	-	77.7 (± 2.7 %)	111.5 (± 2.7 %)	180.0 (± 1.2 %)

Table 6.5: Results, asymmetrical load (UMTS/GSM 80:20) with relative confidence interval for 99 % confidence

	Separate Systems	IS-HO Blind	IS-HO MxRRM	IS-HO Load Balancing
Blocking Rate [%]	4.94 (± 8.7 %)	5.00 (± 8.0 %)	4.96 (± 13.1 %)	5.00 (± 12.5 %)
Dropping Rate [%]	4.56 (± 5.5 %)	1.98 (± 5.7 %)	0.39 (± 13.6 %)	0.28 (± 32.5 %)
Offered Load [Erlang/Cell Area]	23.6 (± 0.0 %)	39.3 (± 0.0 %)	39.0 (± 0.0 %)	40.1 (± 0.0 %)
Carried Load [Erlang/Cell Area]	21.6 (± 0.9 %)	37.3 (± 0.5 %)	37.4 (± 0.7 %)	38.7 (± 0.6 %)
IS-HO Requests [$min^{-1} \cdot$ Cell $Area^{-1}$]	-	44.8 (± 4.2 %)	58.1 (± 4.5 %)	89.2 (± 5.0 %)
IS-HO Failure Rate [%]	-	16.59 (± 5.1 %)	2.48 (± 13.7 %)	1.16 (± 30.7 %)
Directed Retries [$min^{-1} \cdot$ Cell $Area^{-1}$]	-	93.1 (± 2.2 %)	97.0 (± 2.8 %)	191.1 (± 1.05 %)

Table 6.6: Results, asymmetrical load (UMTS/GSM 20:80) with relative confidence interval for 99 % confidence

6.2 MxRRM for PS Services

The 99 % confidence intervals for the simulation results on mean user data rates presented in this section are typically smaller than ± 1.5 %. Since confidence intervals of this order of magnitude do not have further impact on the conclusions that have been drawn from the results, the detailed indication of confidence intervals has been omitted here. In the appendix, starting at page 129, the determining of the confidence intervals is described.

For this investigation the load being offered (ca. 960 kb/s per cell area) was relatively high, i.e. it was close to the maximum that could be handled by the system (for details of scenarios and con-

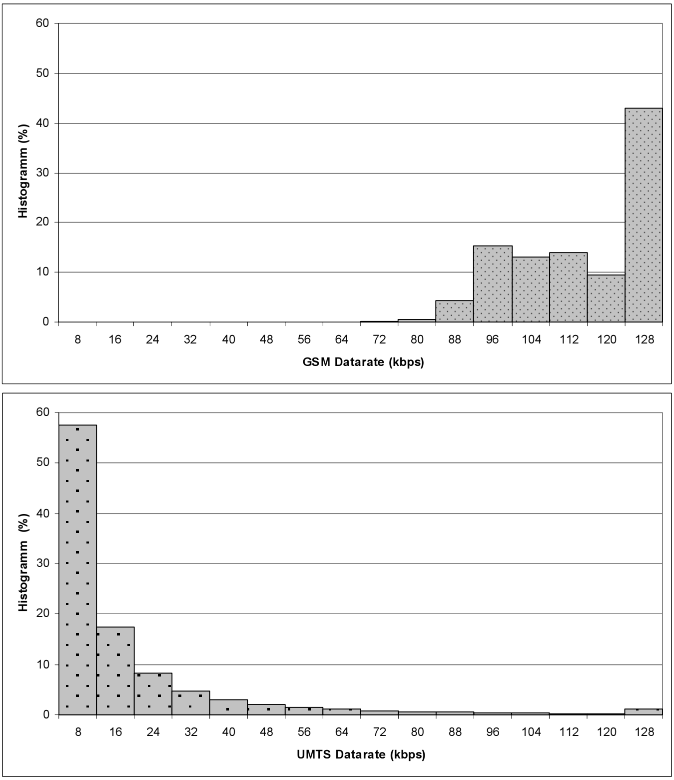


Figure 6.4: Histograms of GSM (top) and UMTS (bottom) user data rates with 80 % of the calls having UMTS as default RAT, using no MxRRM (Separate Systems).

figuration refer to section 5.3). As a result, in the scenarios using the "Separate Systems" algorithm the system experiences a kind of overload situation, where most users achieve only very low data rates, resulting in a much longer duration of packet calls and thus in less packet calls being completed during the simulation, which can be seen clearly e.g. in Figure 6.8.

In worst case, in a cell the number of packet calls being completed could be less than the number of packet calls arriving at the same time, which could irrecoverably start a feedback loop, where low data rates increase the number of concurrent users and thus further decrease the data rates. So for a system, to reliably provide

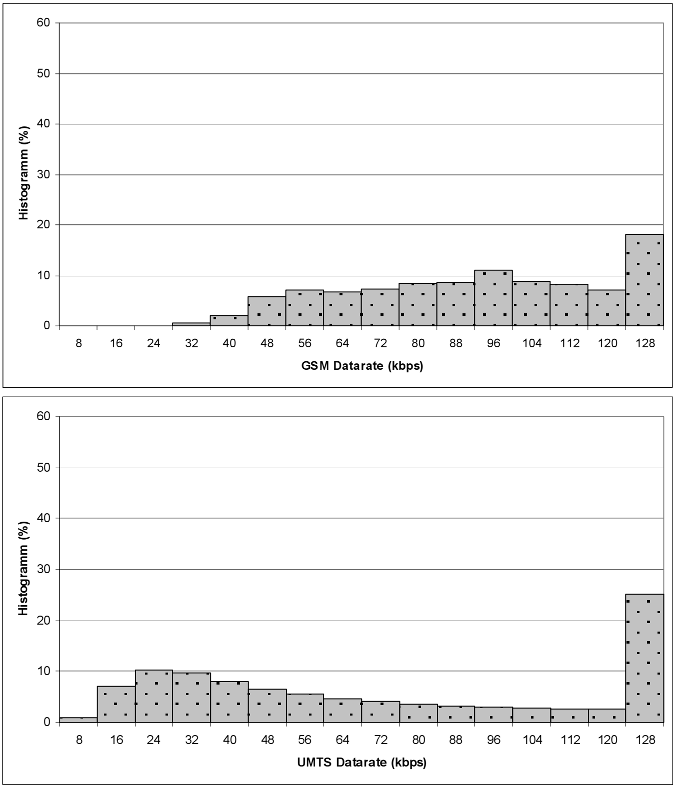


Figure 6.5: Histograms of GSM (top) and UMTS (bottom) user data rates with 80 % of the calls having UMTS as default RAT, using MxRRM "Bit Rate based" algorithm.

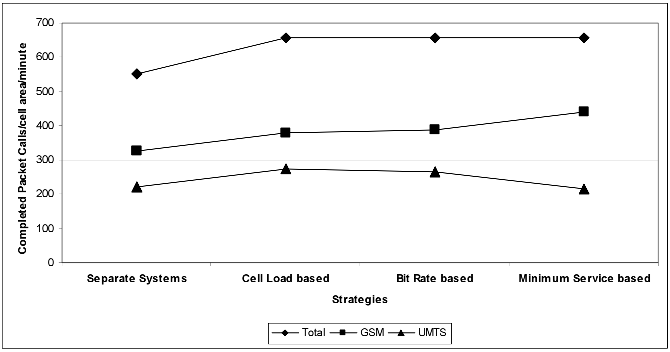


Figure 6.6: Completed PS calls per cell per minute for symmetric traffic load

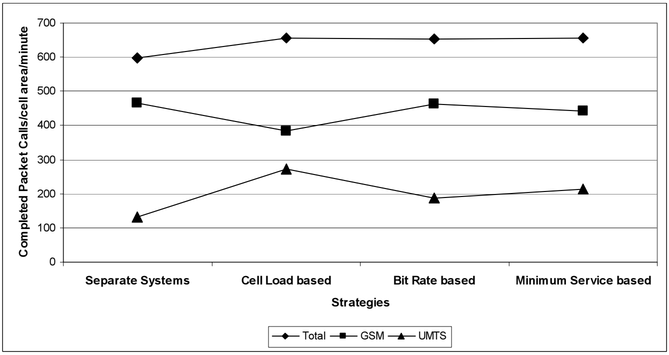


Figure 6.7: Completed PS calls/cell/minute for asymmetric load GSM 80%

sufficient quality of service, it is crucial not to enter such a state of being overloaded.

When applying an MxRRM strategy to the system, the users are directed to the system that currently would offer better performance. As a result, the mean data rates per user are increasing and the total number of packet calls being completed is increased as well. So with MxRRM the overload situation mentioned above could be avoided, thus MxRRM results in higher mean data rates per packet call as well as in increased capacity.

The impact on user data rate is shown in detail in Figure 6.4 and Figure 6.5. Figure 6.4 shows the mean data rate per packet call for separated systems. In this example, UMTS is overloaded since almost no user is achieving an acceptable data rate whereas in GSM almost all users savour excellent data rates. After application of

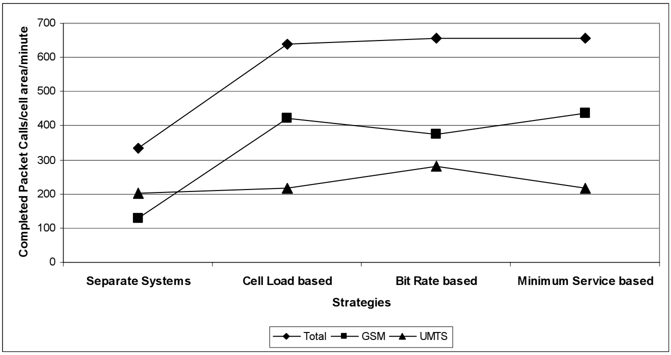


Figure 6.8: Completed PS calls/cell/minute for asymmetric load UMTS 80%

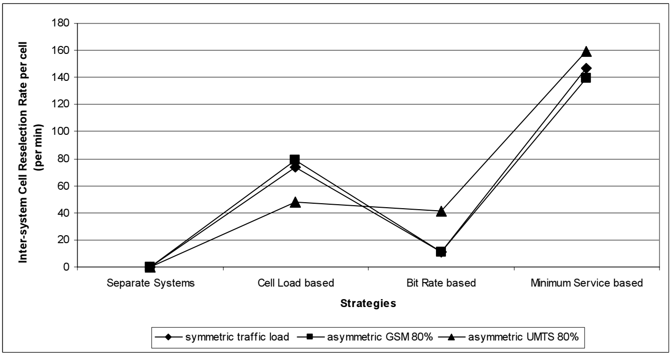


Figure 6.9: Average inter-system cell reselection rate

MxRRM (in Figure 6.5: Bit Rate based algorithm) the distribution of data rates is more balanced, which is shown in Figure 6.5, and the overall performance is improved, as indicated by Figure 6.10

Figure 6.10 shows the increase in mean data rate per user. The results show a different performance for different loads, but MxRRM always improves the data rate significantly compared to separate systems.

Figure 6.6Figure 6.7 and Figure 6.8, respectively, the number of completed packet calls is given in total and per RAT. It can be seen that with MxRRM the number is increased and that the users are distributed among the two RATs according to their capabilities, i.e. since in this configuration the GSM / GPRS systems provides a higher capacity, less users are assigned to UMTS.

Strategy	Symmetric Load	Asymmetric (GSM 80 %)	Asymmetric (GSM 80 %)
Cell Load based	36 %	255 %	25 %
Bit Rate based	30 %	162 %	53 %
Minimum Service based	19 %	205 %	49 %

Table 6.7: Improvement of mean user data rates achieved by MxRRM compared to "Separate Systems"

It can be observed that the maximum value for the total number of completed packet calls for the three algorithms allowing for IS-CRS is approaching the same maximum value, whereas for Separate Systems the value is significantly lower. This behaviour is caused by the fact that with MxRRM disabled for the given traffic load the system is not able to serve the PS calls sufficiently, which leads to more users arriving than leaving the system, i.e. the system is completely overloaded here. With MxRRM enabled, however, for the given traffic load the PS calls are served sufficiently resulting in a stable system state where as many users leave the system as arrive. As indicated by Table 6.7, the individual MxRRM algorithms still differ in their actual performance results, though.

The relative performance gains given in Table 6.7 should only be compared directly within a column, because the reference case

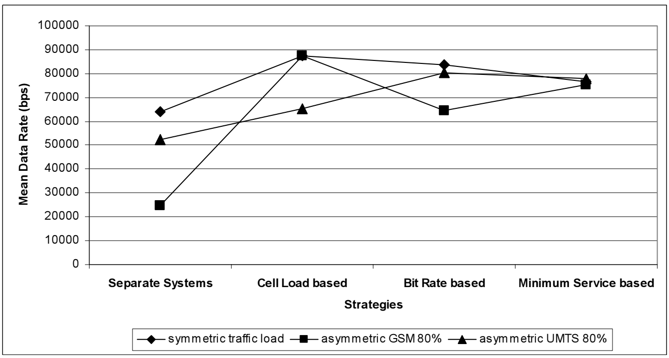


Figure 6.10: Average data rate per completed PS call

"Separated Systems" yields different absolute performance results for each sub scenario. Furthermore, these values tend to be pessimistic, because only completed packet calls are considered, i.e. packet calls that could not be completed during the simulation due to their very low data rates do not contribute to this statistic.

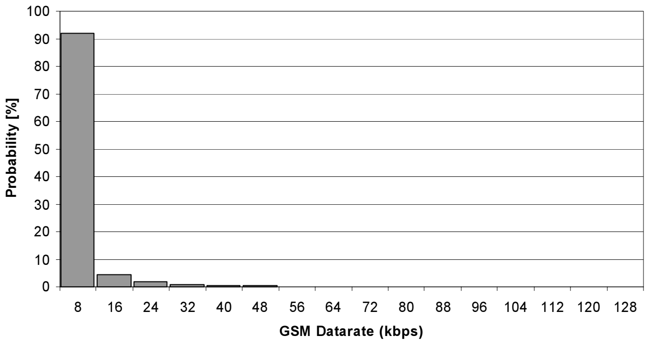
The costs of these performance improvements are displayed in Figure 6.9. With the possibility for IS-CRS the amount of signalling will increase. As a metric the number of IS-CRS is given. With more IS-CRS happening, the signalling load is assumed to increase.

There has no "best" algorithm been identified yet, since an algorithm that achieves a higher mean data rate in one situation may be outperformed in another situation. The Minimum Service based algorithm seems to be less sensitive to the symmetry or asymmetry of the offered load (see Figure 6.6, Figure 6.7 and Figure 6.8), however with the cost of very numerous IS-CRS (see Figure 6.9).

6.3 MxRRM for Mixed Services

The confidence intervals for the simulation results on mean user data rates presented in this section are typically smaller than ± 7 %. In the appendix, starting at page 129, the determining of the confidence intervals is described.

One particular MxRRM algorithm for CS was chosen to be used throughout the whole investigation whereas for PS four different algorithms were used. A medium CS traffic load of 30 Erlang per cell area was configured so that virtually no blocking or dropping occurred (Remark: CS has priority to PS). For PS a mean load of 291 kb/s per cell area was configured. For further details of the scenario and the configuration please refer to section 5.3.



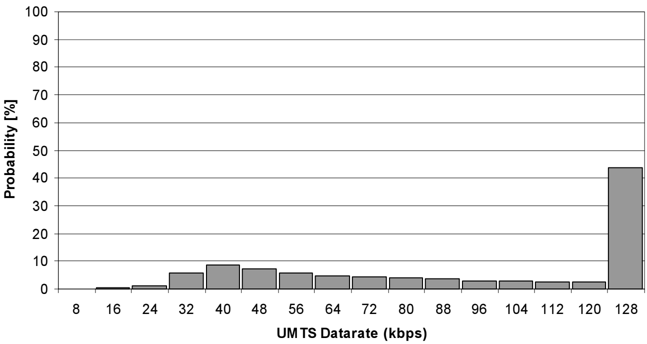


Figure 6.11: Histograms of GSM (top) and UMTS (bottom) user data rates with 80 % of the calls having GSM as default RAT, using no MxRRM ("Separate Systems").

For each of the four MxRRM algorithms three scenarios concerning the traffic arrivals in GSM and UMTS were simulated:

- Symmetrical load: 50 % of the CS calls and 50 % of the PS sessions arrived in UMTS.
- Asymmetrical load UMTS: 50 % of the CS calls and 80 % of the PS sessions arrived in UMTS.
- Asymmetrical load GSM: 50 % of the CS calls and 20 % of the PS sessions arrived in UMTS.

So there were a total number of twelve scenarios to be simulated for this investigation.

The simulation results could be evaluated considering several measures, e.g. cell throughput, number of inter system cell reselections or number of completed packet calls per RAT, but for this paper we focus on the mean data rate per packet call.

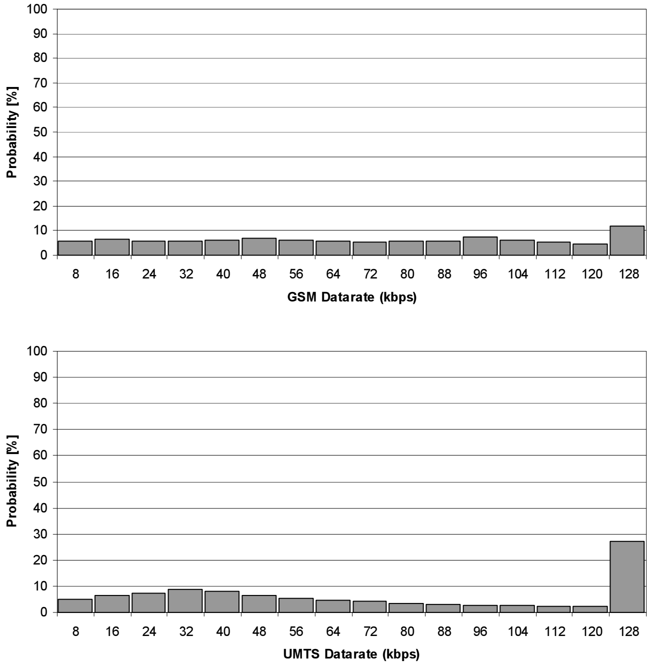


Figure 6.12: Histograms of GSM (top) and UMTS (bottom) user data rates with 80 % of the calls having GSM as default RAT, using MxRRM "Bit Rate based" algorithm.

MxRRM algorithm	Mean data rate per packet call (symmetrical load)	Mean data rate per packet call (asymm. load GSM)	Mean data rate per packet call (asymm. load UMTS)
Separate Systems	64.0 kb/s	21.3 kb/s	27.5 kb/s
Cell Load based	72.1 kb/s	74.2 kb/s	52.9 kb/s
Bit Rate based	67.1 kb/s	68.2 kb/s	66.4 kb/s
Minimum Service based	70.6 kb/s	70.9 kb/s	65.4 kb/s

Table 6.8: Simulation results for mixed services

Figure 6.11 and Figure 6.12 compare the histograms of the mean data rate per packet call of GSM / GPRS and UMTS with and without application of MxRRM. Here, the algorithm "Bit Rate based" and the asymmetrical load GSM were used as an example. It is clearly visible that without MxRRM the GSM system is overloaded, i.e. is providing insufficient data rates, while at the same time UMTS offers good performance. When enabling MxRRM the situation for GSM greatly improves while UMTS still maintains high user data rates.

In Table 6.8 the simulation results are summarized. The greatest gain can be observed in case of asymmetrical load GSM where the algorithm "Cell Load based" achieves an approximately 3.5 times higher data rate than "Separate Systems". Since the absolute values heavily depend on the particular scenario and configuration, the more general conclusion should be emphasized, that also in case of symmetric load the application of MxRRM can lead to significant performance improvements for PS users while maintaining the original service availability for CS users, and that in case of asymmetric load these performance gains can even be further increased.

7 Summary

This doctoral thesis presented approaches for optimizing the network access in heterogeneous wireless communication networks. Three aspects of wireless network access were discussed and performance results have been provided. Most of the results have been published in several articles and patent applications (see Preface on page 9).

The first aspect was indoor wireless network access based on diffuse infrared links. Infrared as transmission medium offers several potential advantages compared to radio waves, e.g. the robustness against electro-magnetic interference or the huge quantity of available spectrum. Infrared transmissions are more or less confined to a room, which enables very efficient spectrum re-use as well as it complicates eavesdropping. In the past high data rates were achieved using directed links, which are inconvenient for the user. For a convenient use, diffuse links are required to enable the user to move or position his device arbitrarily. The realization of an indoor wireless communication system based on diffuse infrared links is complicated by technical and physical constraints, though. Up to now, no high speed system is available to the mass market and even specialized niche products are limited to a maximum data rate of 4 Mb/s.

A new infrared detecting semiconductor element, the Photonic Mixer Device (PMD), was invented at the University of Siegen for the use in Infrared Laser Radar applications. In this thesis, the use of PMD for realizing a wireless communication system was considered. PMD allows for a reliable detection of the transmitted signal even in difficult scenarios with high path loss and strong interference. Hence, it seems well suited for receiving diffuse infrared signals. An evaluation of wireless network access based on diffuse infrared using PMD was performed leading to the conclusion that this approach is very promising.

For future work aiming at creating such a system, several issues still have to be solved. The PMD has to show that it is really capable of continuously achieving high user bit rates using diffuse links. An appropriate physical layer has to be developed as well as a hardware module containing the PMD, the corresponding optics and the circuits for controlling the sensor and for connecting the module to the actual user device, e.g. a notebook computer.

The second aspect discussed in this thesis was the extension of a cellular network by means of ad-hoc relaying. The basic idea is that a user, the mobile client (MC), who is experiencing a high path loss to the base station of the cellular system, does not have a direct connection to the base station but instead uses the terminal of another user as a mobile relay (MR). The connection between MC and MR uses an air interface with ad-hoc capabilities like e.g. WLAN. The resulting advantages are increased coverage as well as increased capacity of the cellular system. The coverage is increased, because even a MC having no physical radio link to the base station could communicate by utilizing a MR. Since the communication link between MC and base station is split up into a first link between base station and MR using the cellular air interface and a second link between MC and MR using a non-cellular air interface, the cellular link physically ends at the mobile relay and the distance to be covered by the cellular air interface is significantly reduced, depending on the range of the non-cellular air interface. Hence, the transmission power required for the cellular link is reduced, which leads to a reduction of interference. Particularly the capacity of CDMA-based cellular systems like UMTS greatly benefits from reduced interference.

Existing work typically focuses on fixed relays or on the use of the same air interface for both links. These approaches offer different characteristics. With fixed relays, the relaying capabilities can be better controlled, but a deliberate deployment of relay stations has to be done. When using the cellular air interface for both links the resources can be better controlled (because systems like WLAN use unlicensed spectrum, which cannot be reserved for exclusive use), but the capacity gain is expected to be less.

In this thesis, the potential capacity gain was estimated considering UMTS as the cellular system and WLAN as the ad-hoc air interface. Based on simulation results, it was shown that significant capacity gains of more than 100 % could be achieved by using ad-hoc relays for extending the cellular system. The actual capacity gain depends on the range of the ad-hoc air interface, the user data rate and the size of the UMTS cells.

For this approach to become operable, an adequate number of users, i.e. mobile terminals that could serve as mobile relays, has to be available. In this thesis, an analytical estimation of the user density required to achieve a particular service availability was performed. The results show that user densities as they can be found in city centres or even in urban areas are sufficient to enable the use of mobile relays.

The existing cellular system would have to be modified to enable the extension by means of ad-hoc relaying. Particularly the signalling procedures would have to be enhanced to provide functionality for discovering potential mobile relays and for establishing the relayed connection. In this thesis, a generalized approach for such signalling procedures was introduced.

In total, the investigation of an extension of a cellular system based on ad-hoc relays presented here showed promising results regarding possible performance gains as well as regarding the feasibility of this approach. For further research work, the signalling procedures could be adopted according to the requirements of a concrete cellular system like UMTS. Additionally algorithms, that determine whether a mobile relay should be used and which (including potential handovers between base station and mobile relay or between two mobile relays), could be investigated. Based on that, a more detailed simulation could be performed to learn more details on possible performance gains.

The main aspect discussed in this thesis focused on improved resource management when having a heterogeneous communication infrastructure, the so called Multi Standard Radio Resource Management (MxRRM). With MxRRM, previously independent radio access technologies are coupled to enable the evaluation of current load information exchanged mutually. Based on this evaluation and the capabilities of each radio access technology, calls can be assigned to the most appropriate system, which leads to a better utilization of scarce air interface resources. Currently communication systems that are potentially available in the same area exist independently from each other. A kind of “emergency” or “blind” handover between the systems may be possible, but a sophisticated overall radio resource management does not exist. As a result, it could happen that e.g. a GSM cell is overloaded, offering poor service to the users or even rejecting users, while at the same time e.g. a co-located UMTS cell is idle. With UMTS Release 5 an interface was introduced for the exchange of cell load information, but no mechanisms have been defined for actually utilizing this information.

Related research work was mainly focused on architectural issues, only recently the EU funded IST project EVEREST has been started, which will include also the investigation of algorithms.

In this thesis, several algorithms for Multi Standard Radio Resource Management were defined. Since the signalling capabilities of the real system, i.e. UMTS Release 5, have been considered, the practical application of the algorithms would be possible. Their performance was investigated by extensive simulations considering GSM/GPRS/EDGE and UMTS as two concurrently available radio access technologies. For this purpose, a powerful simulator has been developed. The simulation results show that MxRRM could significantly increase the capacity and the service availability of a heterogeneous mobile communication system.

For circuit switched services, four algorithms were defined. If blocking or dropping would occur, the algorithms try to perform an intersystem handover for the call affected. If this attempt fails, then the call is blocked or dropped effectively.

The reference case “Separate Systems” does not allow intersystem handovers, i.e. the radio access technologies (RATs) are completely independent. With “Intersystem Handover (blind)” intersystem handovers are possible, but not the exchange of load information. Hence, the target cell selected in the alternative RAT may not be able to accept an additional call, so there is a high probability of intersystem handover failure.

The third algorithm, “Intersystem Handover (MxRRM)”, introduces MxRRM functionality. Here load information is exchanged between the GSM and UMTS system, so the probability of intersystem handover failure is reduced.

The algorithm “Load Balancing” is identical to the previous algorithm, with the addition that during call setup a load balancing is applied to equalize the load between GSM and UMTS.

For maintaining the same blocking rate of 5 % the simulation results show an increase of capacity (offered load in Erlang) of up to 6.9 % while the dropping rate is reduced by up to 86 %, when comparing MxRRM algorithms to “Intersystem Handover (blind)”. When comparing to “Separate Systems” a capacity increase of up to 70 % and a reduction of dropping by up to 95 % is obtained.

For packet switched services, again four algorithms were defined. When a new packet call (i.e. a download of a web page) arrives to the system, then the radio access technology potentially offering the best performance to the user is determined and the packet call is assigned accordingly to a cell. This means that a change of the radio access technology, i.e. an intersystem cell reselection (IS-CRS), can only happen at the beginning of a packet call. The MxRRM algorithms for packet switched services were distinguished by the parameter being used as a measure for determining the best radio access technology.

The reference case was again "Separate Systems", which does not allow intersystem cell reselections.

The "Cell load based" algorithm compares the current cell load of the potential target cells in both radio access technologies and selects the least loaded one.

The "Bit Rate based" algorithm compares the current cell mean user data rates of the potential target cells in both radio access technologies and selects the one providing the highest data rate.

For the "Minimum Service based" algorithm a user is supposed to be "satisfied" if he is currently achieving at least half of the maximum data rate supported by his terminal. This algorithm compares the current percentage of satisfied users of the potential target cells in both radio access technologies and selects the one having the highest proportion of satisfied users.

The simulation results show an increase of the mean user data rates of up to 255 % when using an MxRRM algorithm compared to "Separate Systems". At load levels where "Separate Systems" resulted in an overloaded system, with MxRRM algorithms a stable operation was still possible, so not only the mean user data rates, but also the capacity was improved by MxRRM.

In mixed service scenarios dealing with circuit switched as well as packet switched traffic at the same time, the increase of mean user data rates for packet switched traffic is even higher while for circuit switched traffic identical service availability was maintained, when comparing the MxRRM algorithms to "Separate Systems".

The algorithms for packet switched traffic yield different gains with different scenarios, so there is no "best" algorithm to be clearly identified yet.

From this simulation results it can be concluded that significant performance gains (in terms of increased service availability, capacity and mean user data rates) for the heterogeneous mobile communication system can be achieved by applying MxRRM. Since these gains can be realized without building additional base stations, maybe even without replacing any equipment provided the functionality could be added by software updates, this approach seems very promising.

For future research, it could be further investigated how the models used influence the simulation results, which could lead to a more accurate simulation. For example, the modelling of the UMTS power control is expected to be optimistic.

Since the MxRRM algorithms investigated here offer many degrees of freedom regarding their actual configuration, further studies could be performed to better understand the interaction of algorithms and systems as well as to better understand the impact of individual parameters. A next step could be the development and investigation of new algorithms.

Interesting results could be obtained by including additional characteristics of real systems like for example, constraints created by signalling capabilities, like delay, possible update intervals, possible range of values, etc. Also variations of the scenarios including the network topology or the addition of other service types and applications, like e.g. video streaming, could lead to new insights.

An important enhancement to the simulator would be the addition of other radio access technologies, like for example WLAN or WiMAX. The impact would be twofold. In the first place, with more than two radio access technologies available the probability of choosing correctly the best radio access technology by accident decreases, so the advantage of MxRRM based approaches compared to others should increase. Secondly, the MxRRM algorithms face a greater challenge when trying to find the best radio access technology out of more than two possibilities. Additionally, the other radio access technologies may have other characteristics, which may have impact on the algorithms. Hence, there would be more room for developing and optimizing algorithms.

Thus MxRRM will continue to be a promising and challenging topic of research within the field of mobile communications.

Appendix: Discrete Event Simulation

Since continuous time simulations are difficult if not impossible to apply to the simulation of whole communication systems, in principle, there are only two different possibilities of modelling "time".

In the discrete time step model (in [69] also named fixed-increment time advance approach), during simulation the state of the system models is evaluated only at specific regularly spaced time instants (see Figure A.1). Changes in system state happen only at these particular points in time. If an event – like e.g. the arrival of a new call or the movement of a mobile terminal – happens between two evaluation points, then effectively the event is not evaluated until the next evaluation point after this event (in Figure A.1 this can be observed for events 2, 3, 4, 5 and 6).

This behaviour can be seen as a kind of sampling error, as indicated for event 4 in Figure A.1. By increasing the sampling rate, i.e. by decreasing the time intervals between two evaluation points, this error can be reduced, but as long as the events can happen at any time a sampling error will always persist. So, when using the discrete time paradigm, it has to be estimated, which time resolution would be required to achieve the desired quality for the results. A further drawback is that the evaluation of system state is performed periodically in the pre-defined intervals even if there was no change in system state, i.e. even if no event happened, which may reduce the simulation performance significantly.

An advantage of using discrete time is the possibility to represent the system state as vectors or matrices, which can be processed efficiently by specialized software and hardware and even allow for the processing being performed in parallel on several CPUs or computers. [25]

In the event driven model (in [69] also named next-event time advance approach) the evaluation of system state happens each time an event occurs (see Figure A.2). All events are placed into an event queue and are processed one event at a time. If more than one event is to happen at exactly the same time then typically the events are processed in the order they were scheduled. With this paradigm the time an event takes place can be accurately modelled

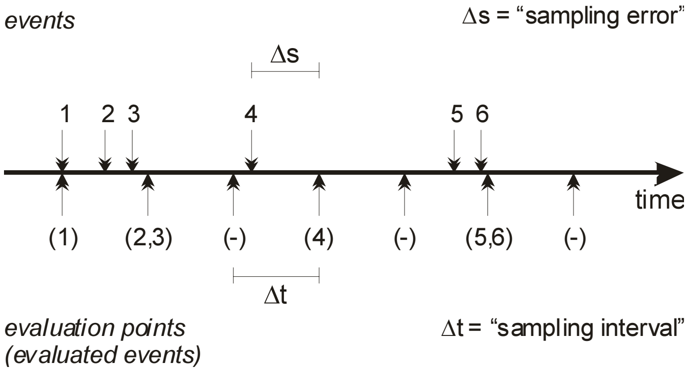


Figure A.1: Discrete time model

without a sampling error. Additionally, during periods with no events to happen no computation is required. [25]

So for both approaches, there are good reasons to use one or the other, depending on the actual purpose and also on personal preferences. For simulations of economical systems, the input data are often available based on fixed periods, e.g. per year, so a fixed-increment time advance approach could be appropriate. For simulations of communications systems, the discrete event simulation is widely adopted and many of the major simulation software packages are based on this, like e.g. OPNET [53], ns2 [70], OMNet [71] and others. [69]

For the research work presented in this thesis the discrete event paradigm was selected.

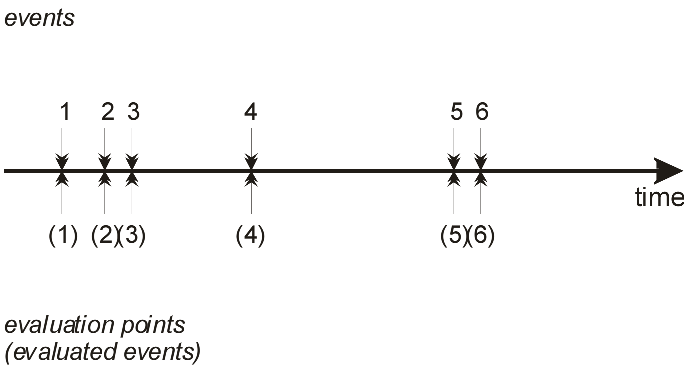


Figure A.2: Discrete event model

Appendix: Statistical Confidence

In general, the collection of elements under investigation is named the population and the selected subset is known as a sample. For estimating the characteristics of the entire population based on the data collected by a sample, methods of statistical inference are used. This estimation becomes more accurate with increasing number of independent samples and reaches perfect consistence with the true value when the number of samples includes all elements of the population.

A simulation project is composed of experiments. Experiments are differentiated by the use of alternatives in a model's logic and/or data. Each experiment consists of one or more replications (trials). A replication is a simulation run that uses the experiment's model logic and data but a different set of random numbers, and so produces different statistical results that can then be analyzed across a set of replications.

For stochastic simulations, the population can be considered to consist of all possible replications (simulation runs). The set of actual simulation runs constitutes the samples, from which the characteristic of the population can be estimated. [29], [69]

Since the actual results of a particular simulation run depend on the sequence of random numbers used to control the course of events during the simulation, each different sequence of random numbers will lead to a different outcome. Consequently, the number of possible simulations runs (population) is virtually infinite, so the number of simulation runs being actually performed (samples) has to be limited to a reasonable value.

A second possibility to get independent samples is to divide one simulation run into sections of the same duration and to evaluate each section as if it was an independent simulation run. This approach is called the "method of batch means" [29], [69]. Since this approach is based on a single long simulation run, it has to go through the transient period at the beginning of each simulation run only once, thus increasing the simulation efficiency. A necessary prerequisite for this is the sections (batches) being statistically independent from each other or in other words, the sections have to be of adequate length. If the duration of the batches is long enough, it can be shown that the results obtained for each batch are approximately uncorrelated. Consequently, the batch means constitute the actual samples, from which the characteristic of the population can be estimated. For example, the mean value can be obtained as given by Equation (26) and the standard deviation can be obtained as given by Equation (27).

$$\bar{x} = \frac{1}{n}\sum_{i=1}^{n} x_i \tag{26}$$

With $\bar{x}$ *being the sample mean,* n *being the number of samples, and* x_i *being the* i-th *sample.*

$$s = \sqrt{\frac{1}{n-1}\sum_{i=1}^{n}\left(x_i - \bar{x}\right)^2} \tag{27}$$

With s *being the estimate of the population's standard deviation based on the sample standard deviation,* n *being the number of samples,* x_i *being the* i-th *sample, and* $\bar{x}$ *being the sample mean.*

Since for obtaining a good estimate of the "true" value a sufficient number of samples has to be taken - either by performing several simulation runs or by using the method of batch means – it has to be determined how many samples actually constitute a sufficient number.

The answer is provided by the concept of confidence intervals. The confidence interval is a range of values that has a high probability of containing the true value of the parameter being estimated. If our estimation, i.e. the sample mean, satisfies the condition given in Equation (28), then we say that the interval $\left(\bar{x} - \varepsilon,\ \bar{x} + \varepsilon\right)$ is a *(100 α) %* confidence interval. For example, the 99 % confidence interval is constructed in such a way that 99 % of such intervals will contain the true value of the parameter [29], [69].

$$P\left(\bar{x} - \varepsilon < \mu < \bar{x} + \varepsilon\right) = \alpha \tag{28}$$

With P *being the probability that the true mean value is within the given interval,* $\bar{x}$ *being the sample mean,* 2ε *being the size of the interval,* μ *being the true value of the parameter under investigation, and* α *being the value of* P.

The size of a confidence interval can be calculated according to Equation (29). This approach is valid only if the sample values x_i (i.e. the mean values of the individual batches) are normal distributed. With increasing number of samples, the distribution of an average is asymptotically Normal, even if the distribution from which the average is computed is decidedly non-Normal (Central Limit Theorem). Hence, for the investigation presented in this thesis, the Equation (29) can be applied.

$$\varepsilon = t_{n-1,\alpha/2} \frac{s}{\sqrt{n}} \tag{29}$$

With 2ε *being the size of the confidence interval,* $t_{n-1,\alpha/2}$ *being defined based on the Student t distribution such that the area under the t probability density function to its right is equal to* $\alpha/2$ *(actual values of* t *for given* n *and* α *can be found in tables),* s *being the estimate of the population's standard deviation based on the sample standard deviation, and* n *being the number of samples.*

As can be seen from Equation (29), for a given confidence and a given standard deviation, the size of the confidence interval becomes smaller with increasing number of samples. Furthermore, it can be seen that the impact of the standard deviation is more significant than the impact of the number of samples.

One important prerequisite for calculating a confidence interval

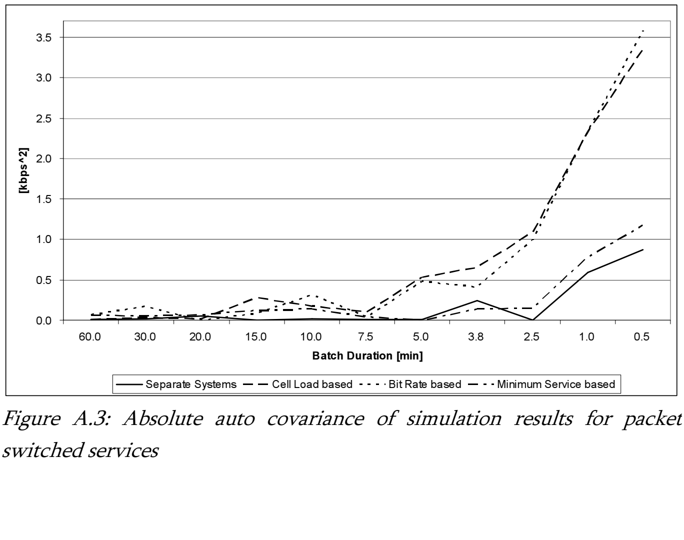


Figure A.3: Absolute auto covariance of simulation results for packet switched services in this way is the statistical independence of the samples, as mentioned above. If there are correlations between the samples then the confidence interval obtained is too optimistic, i.e. too small. As a measure the auto covariance is used according to Equation (30). If the auto covariance equals zero then there is no linear dependence between the samples (Caveat: non-linear dependencies will not be indicated by covariance!). For evaluating simulation results using the method of batch means this means that the batch duration has to be set in order to achieve a small absolute value for the resulting auto covariance [29], [69], [72].

$$C = \frac{1}{n-1-\tau} \sum_{i=1}^{n-\tau} \left(x_i - \overline{x}\right)\left(x_{i+1} - \overline{x}\right) \tag{30}$$

With C *being the auto covariance with lag* τ*,* n *being the number of samples,* x_i *being the* i-th *sample, and* $\overline{x}$ *being the sample mean.*

In Figure A.3 as an example the absolute auto covariances (with lag 1) of four different simulations are shown. It can be seen that they never achieve zero, but that the values are becoming smaller with increasing batch duration. To achieve a small value, here the batch size should be set to five minutes at least. The batch size used for the investigation presented in this thesis was 20 minutes (e.g. resulting in typical confidence intervals between ± 0.5 % and ± 1.5 % for the results presented in section 6.2).

As a consequence, the confidence intervals given in this thesis are supposed to be optimistic. Since here even much larger confidence intervals would not change the conclusions that have been derived from the simulation results, this approach is considered to be adequate.

Appendix: Overview of the Software Architecture of the Simulator

The commercially available simulation package Opnet [53] provides a graphical user interface for the implementation of models. At source code level, state machines have to be defined by implementing corresponding chunks of source code that will be executed when state transitions occur.

Albeit this approach is intuitive, e.g. for modelling of communication protocols, it is not suitable for implementing a large software package using an object-oriented approach. Hence, the major part of the simulator has been implemented as a class hierarchy, which is visualized in Figure A.4. The implementation has been performed using C++ conforming to the ANSI standard ([54], [73]) und using the Standard Library [57], particularly the Standard Template Library (STL) [58]. The STL provides well tested and efficient source code for container classes, which contributes to the creation of fast and reliable software.

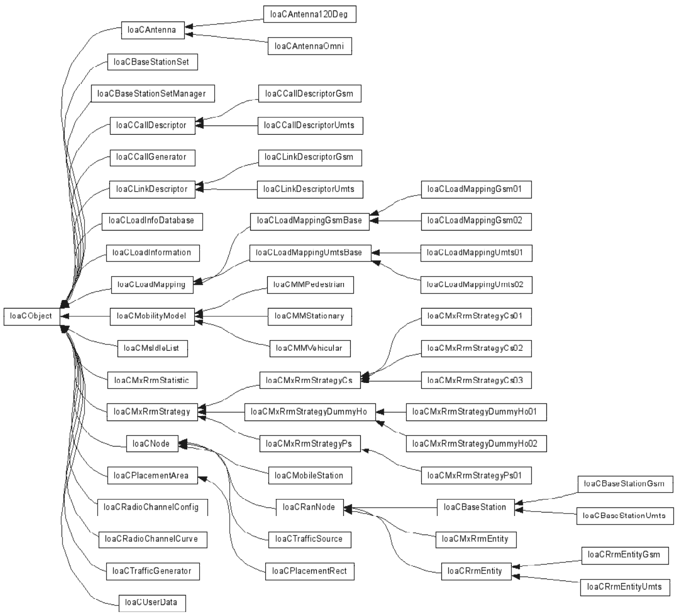


Figure A.4: Overview of the class hierarchy (inheritance diagram)

Opnet allows regular including of these classes as “external sources”. Due to the ANSI compliance of the source code developed during this work, the simulator can be run on all platforms supported by Opnet.

The entities existing in a simulation can be structured according to Figure A.5. The entities belonging to a particular radio access technology are grouped into one subnet (subnets are a logical structure within Opnet and must not be confused with subnets of the Internet Protocol) and also the mobile stations are grouped into a subnet of their own. This structure has no actual impact on the software architecture but was introduced for the convenience of the user, since otherwise the handling of several thousand entities

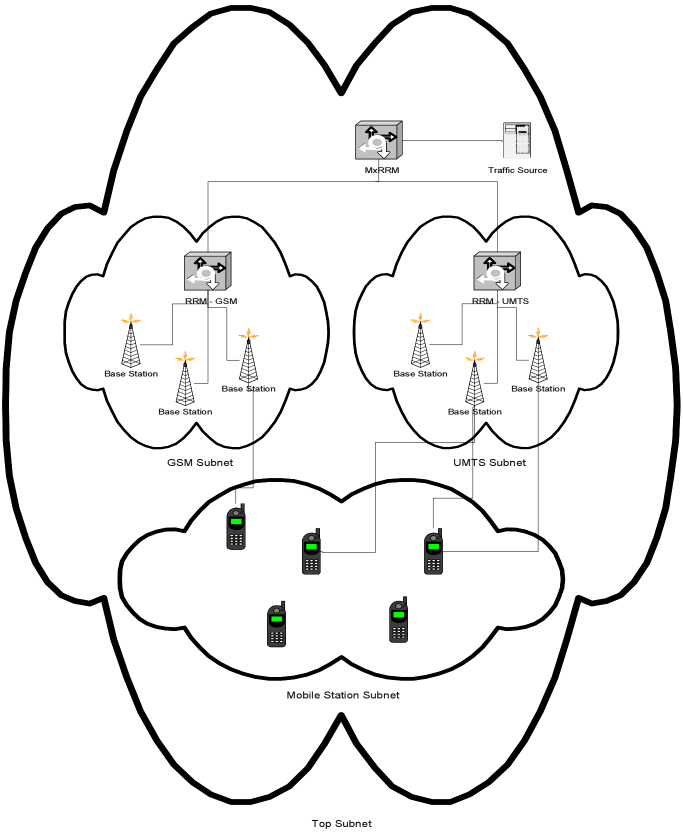


Figure A.5: Entities of a simulation scenario would be very uncomfortable.

The object-oriented approach allows for a general definition of base classes, which define the class interface and may even provide fundamental functionality. In derived classes only specialized functions have to be added.

For example, the asynchronous communication between node objects is done by message passing. The processing of incoming messages is implemented in the base class *IoaCNode* (see Figure A.6) including a default message handler, which creates a warning message that there is no appropriate message handler. In derived classes, only a message handler function for each message type, that has to be handled by the particular class, has to be implemented. If e.g. a base station object receives a message, than this message is processed using the functionality of the base class. If the base station object can handle the particular message type, then the message is passed to the object's appropriate message handler function. If the message type is unknown to the base station object then the default handler of the base class will create a warning message, thus informing the developer that there was an unexpected behaviour.

Since the object-oriented paradigm is a well known approach for software development, an extensive discussion is not provided here.

For classes, of which objects are frequently dynamically created and deleted, particular operators *new* and *delete* have been implemented to improve the memory management. These operators use a "pooled memory" approach supported by Opnet to reduce the performance costs of again and again allocating and de-allocating memory for numerous objects of the same type.

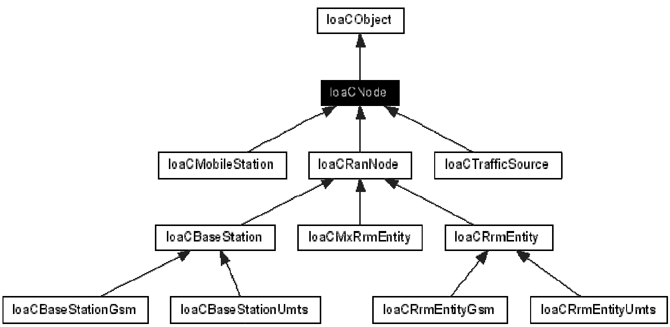


Figure A.6: Overview of classes derived from IoaCNode

A profiler has been used to identify the parts of the software that consume the most processing time. As to be expected, the functions related to UMTS power control have been identified. Consequently, for further significant performance improvements these functions as well as the actual modelling of the UMTS power control would have to be reconsidered. Nevertheless, the simulator yields very good performance; even with the large scenarios used for this investigation, the computation time is between 0.5 and 3.0 times the simulated time, depending on the configuration and the workstation being used.

A more detailed description of the simulator software would exceed the scope of this thesis significantly, so please refer to [74] for further information.

Acronyms

3GPP	3rd Generation Partnership Project
AAA	Authentication, Authorization and Accounting
ANSI	American National Standards Institute
AP	Access Point
ASIC	Application Specific Integrated Circuit
BSC	Base Station Controller
CCD	Charge Coupled Device
CDMA	Code Division Multiple Access
CMOS	Complementary Metal Oxide Semiconductor
CN	Core Network
CPLD	Complex Programmable Logic Device
CPU	Central Processing Unit
CRRM	Common Radio Resource Management
CS	Circuit Switched
DS-CDMA	Direct Sequence Code Division Multiple Access
EDGE	Enhanced Data rates for Global Evolution
FDD	Frequency Division Duplex
FPGA	Field-Programmable Gate Array
GAL	Generic Array Logic
GPRS	General Packet Radio Service
GSM	Global System for Mobile communication
HLR	Home Location Register
HSDPA	High Speed Downlink Packet Access
IE	Information Element
IEEE	The Institute of Electrical and Electronics Engineers
IR	Infra Red
IS-CRS	Inter System Cell Reselection
IS-HO	Inter System Hand Over
JRRM	Joint Radio Resource Management
LED	Light Emitting Diode
lpp	Link Performance Parameters
MC	Mobile Client
MR	Mobile Relay
MxRRM	Multi Standard Radio Resource Management
NRT	Non Real Time
PDA	Personal Digital Assistant
PLL	Phase Locked Loop

PMD	Photonic Mixer Device
PS	Packet Switched
QoS	Quality of Service
RAN	Radio Access Network
RANAP	Radio Access Network Application Part
RAT	Radio Access Technology
RF	Radio Frequency
RNC	Radio Network Controller
RNSAP	Radio Network Subsystem Application Part
RT	Real Time
SDR	Software Defined Radio
SGSN	Serving GPRS Support Node
UMTS	Universal Mobile Telecommunication System
UTRAN	UMTS Terrestrial Radio Access Network
WLAN	Wireless Local Area Network

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